

The Canonical Csound Reference Manual

Version 5.10

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The Canonical Csound Reference Manual: Version 5.10

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Preface

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Preface to the Csound Manual

Barry Vercoe, MIT Media Lab

Realizing music by digital computer involves synthesizing audio signals with discrete points or samples representative of continuous waveforms. There are many ways to do this, each affording a different manner of control. Direct synthesis generates waveforms by sampling a stored function representing a single cycle; additive synthesis generates the many partials of a complex tone, each with its own loudness envelope; subtractive synthesis begins with a complex tone and filters it. Non-linear synthesis uses frequency modulation and waveshaping to give simple signals complex characteristics, while sampling and storage of a natural sound allows it to be used at will.

Since comprehensive moment-by-moment specification of sound can be tedious, control is gained in two ways: 1) from the instruments in an orchestra, and 2) from the events within a score. An orchestra is really a computer program that can produce sound, while a score is a body of data which that program can react to. Whether a rise-time characteristic is a fixed constant in an instrument, or a variable of each note in the score, depends on how the user wants to control it.

The instruments in a Csound orchestra (see *Syntax of the Orchestra*) are defined in a simple syntax that invokes complex audio processing routines. A score (see *The Standard Numeric Score*) passed to this orchestra contains numerically coded pitch and control information, in standard numeric score format. Although many users are content with this format, higher level score processing languages are often convenient.

The programs making up the Csound system have a long history of development, beginning with the Music 4 program written at Bell Telephone Laboratories in the early 1960's by Max Mathews. That initiated the stored table concept and much of the terminology that has since enabled computer music researchers to communicate. Valuable additions were made at Princeton by the late Godfrey Winham in Music 4B; my own Music 360 (1968) was very indebted to his work. With Music 11 (1973) I took a different tack: the two distinct networks of control and audio signal processing stemmed from my intensive involvement in the preceding years in hardware synthesizer concepts and design. This division has been retained in Csound.

Because it is written entirely in C, Csound is easily installed on any machine running Unix or C. At MIT it runs on VAX/DECstations under Ultrix 4.2, on SUNs under OS 4.1, SGI's under 5.0, on IBM PC's under DOS 6.2 and Windows 3.1, and on the Apple Macintosh under ThinkC 5.0. With this single language for defining the audio signal processing, and portable audio formats like AIFF and WAV, users can move easily from machine to machine.

The 1991 version added phase vocoder, FOF, and spectral data types. 1992 saw MIDI converter and control units, enabling Csound to be run from MIDI score-files and external keyboards. In 1994 the sound analysis programs (lpc, pvoc) were integrated into the main load module, enabling all Csound processing to be run from a single executable, and Cscore could pass scores directly to the orchestra for iterative performance. The 1995 release introduced an expanded MIDI set with MIDI-based linseg, butterworth filters, granular synthesis, and an improved spectral-based pitch tracker. Of special importance was the addition of run-time event generating tools (Cscore and MIDI) allowing run-time sensing and response setups that enable interactive composition and experiment. It appeared

that real-time software synthesis was now showing some real promise.

History of the Canonical Csound Reference Manual

This initial version of this manual for early versions of Csound was started at MIT by Barry L. Vercoe and maintained there during the 1980's and start of the 1990's. Some of the manual comes from documents for programs like *Music11* from the 1970's. This original manual was improved and worked on by Richard Boulanger, John ffitich, Jean Piché and Rasmus Ekman.

This manual led to the Official Csound Reference Manual, still located at: <http://www.lakewoodsound.com/csound> [http://www.lakewoodsound.com/csound/hypertext/manual.htm], for Csound version 4.16, November, 1999, which was maintained by David M. Boothe.

A parallel version of the manual called the Alternative Csound Reference Manual, was developed by Kevin Conder using *DocBook/SGML* [<http://www.docbook.org/>]. This version later became the Canonical version.

When Csound was licenced as LGPL by MIT in 2003, the manual was licenced GFDL and placed on Sourceforge along with the sources of Csound.

In the winter of 2004, the Canonical Manual was converted to DocBook/XML by Steven Yi to allow for more people to be able to compile and maintain the manual.

The manual is currently maintained by Andrés Cabrera with continuous contributions from the Csound Community.

The manual continues to be a community run project that depends on the contributions of developers and users to help refine the coverage and accuracy of its contents. All contributions are welcome and appreciated.

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This list is by no means complete. More information can be gathered from the Changelog file in the manual's sources repository.

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Developed by *Barry L. Vercoe* at the Experimental Music Studio, Media Laboratory, M.I.T., Cambridge, Massachusetts, with partial support from the System Development Foundation and from National Science Foundation Grant # IRI-8704665.

Manual

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This Csound language documentation in this manual is derived from Kevin Conder's *Alternative Csound Reference Manual*, which in turn is derived from the *Public Csound Reference Manual*.

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This legal notice is from the *Public Csound Reference Manual*: "The original Hypertext Edition of the MIT Csound Manual was prepared for the World Wide Web by *Peter J. Nix* of the Department of Music at the University of Leeds and *Jean Piché* of the Faculté de musique de l'Université de

Montréal. A Print Edition, in Adobe Acrobat format, was then maintained by *David M. Boothe*. The editors fully acknowledge the rights of the authors of the original documentation and programs, as set out above, and further request that this notice appear wherever this material is held.”

The Public Csound Reference Manual's last known network location was <http://www.lakewoodsound.com/csound/hypertext/manual.htm>.

The Alternative Csound Reference Manual's network location, for both the Transparent and Opaque copies, is <http://kevindumpscore.com/download.html#csound-manual>.

The Csound and CsoundAC Manual's network location is <http://sourceforge.net/projects/csound>.

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Virtual Synthesis Technology

Virtual Synthesis Technology (VST) PlugIn interface technology by Steinberg Soft- und Hardware GmbH.

Getting Started with Csound

Downloading

In case you don't already have Csound (or have an older version) download the appropriate Csound version for your platform from the *Sourceforge Csound5 Download Page* [http://sourceforge.net/project/showfiles.php?group_id=81968&package_id=120482]. Installers for Windows have '.exe' extension and for Mac '.dmg' or '.tar.gz'. If the installer's filename ends in '-d' it means the installer has been built with *double* precision (64-bit) which provides higher quality output than the ordinary *float* precision (32-bit), which provides quicker output. You can also download the sources and build them, but this requires more expertise (See the section *Building Csound*).

It's also useful to download the most recent version of this manual, which you will also find there.

Running

Csound can be run in different ways. Since Csound is a command line program (DOS in Windows terms), just clicking on the csound executable will have no effect. Csound must be called either from a terminal (or DOS prompt), or from a front-end. To use Csound from the command line, you must open a *Terminal* (Command Prompt or DOS Prompt on Windows). Using Csound from the command line can be difficult if you've never used the terminal, so you may want to try to use one of the front-ends included with your distribution. A *front-end* is a graphical program that assists running Csound and can usually help edit csound files.

Both in the case of front-ends as well as execution from the command line, Csound needs two

things:

- A Csound file ('.csd' or possibly an '.orc' and a '.sco' file)
- A list of command line flags (or configuration options) that configure execution. They determine things like output filename and format, whether real-time audio and MIDI are enabled, which soundcard to use, the buffer size, the amount of messages printed, etc. These options can be included in the '.csd' file itself, so for the examples included in this manual *you shouldn't need to worry about them*. The complete and very long list of available command flags can be found *here*, but you might want to have a look there later...

See the section *Configuring* if Csound is giving you trouble.

This documentation includes many '.csd' files which you can try out, and which should work directly from the command line or from any frontend. A simple example is *oscil.csd* [examples/oscil.csd] that can be found in the *examples* folder of this documentation. Your front-end should allow you to choose the file, and it should have a 'play' or 'render' button.



Note for MacCsound users

You might need to remove all the lines from the command options slot in order for the manual examples to work.

You can also try the manual examples from the command line by navigating to the examples directory of the manual using something like this on Windows (assuming the manual is located at c:\Program Files\Csound>manual\):

```
cd "c:\Program Files\Csound>manual\examples"
```

or something like:

```
cd /manualdirectory/manual/examples
```

for the Mac or linux terminals and then typing:

```
csound oscil.csd
```

The example files are configured to run in realtime by default, so you should have heard a 2 second sine wave.

Writing your own .csd files

A .csd file looks like this (this file is *oscils.csd* [examples/oscils.csd]):

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o oscils.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a fast sine oscillator.
instr 1
  iamp = 10000
  icps = 440
  iphs = 0
```

```
al oscils iamp, icps, iphs
out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsSoundSynthesizer>
```

Csound's .csd files have 3 main sections between the `<CsSynthesizer>` and `</CsSynthesizer>` tags:

- *CsOptions* - Includes the *Command Line flags* specific to this particular file. These options can also be set using the .csoundrc file or directly in the *command line*. Some frontends also provide ways to specify global or local options.
- *CsInstruments* - Contains the instruments or processes available in the file. Instruments are defined using the *instr* and *endin* opcodes. The *CsInstruments* section also contains the *Orchestra Header* which defines things like *sample rate*, the *number of samples in a control period* and the *number of output channels*.
- *CsScore* - Contains the 'notes' to be played optionally the definition of f-tables. Notes are created using the *i statement*, and f-tables are created using the *f statement*. There are several other *score statements* available.

Note that anything after a semicolon (;) until the end of the line is a comment, and is ignored by csound.

You can write csd files in any plain text editor like notepad or textedit. Just be sure to save the file as plain text (not rich text). Many *frontends* include advanced editing capabilities with syntax highlighting and completion.

You can find an in depth tutorial on getting started with Csound written by Michael Gogins here [<http://csound.sourceforge.net/tutorial.pdf>].

What's new in Csound 5.10

New in Version 5.10 (December 2008)

- New functionality
 - New option to listen to all MIDI devices using the portmidi realtime module. To enable listening to all devices use "-+rtmidi=portmidi -Ma".
 - Dither on output implemented; rectangular and triangular dither available in some cases
 - *GEN20* type 6 now has option to set variance
- Bug fixes and improvements:
 - Locale set to C numeric to avoid , versus . problems.
 - *diskin* fixed
 - *outo* was broken regarding channel 6
 - *pitchamdf* fixed

- *zfilter2* initialization fixed
- *s32b14* fixed
- Fixed other bugs fixed that have not been reported publicly.
- Internal Changes:
 - The major version of the Csound API is increased to 2; affected *csound.so* as well. This means that Csound 5.10 is incompatible with applications ("front ends", "clients", or "hosts") that were built for Csound 5.08 and earlier and that use API version 1.x. These applications will need to be rebuilt to work with the current and future versions of Csound. Csound front ends written in interpreted languages such as Python or Java may continue to work without modification. It may also be possible to keep both an earlier version of the Csound library and an API 2.0 version on the same machine together so that new and old Csound-based applications can run side-by-side. These changes do not in any way affect the compatibility of Csound orchestras and scores: all old documents should continue to work as before.
 - Time now counted internally in samples, overcoming a longstanding bug with rounding of time to k-rate.
 - Many internal changes related to branch prediction. Some opcodes are substantially quicker.
 -

New in Version 5.09 (October 2008)

- New opcodes:
 - New *vosim* opcode by Rasmus Ekman which recreates the historic VOSIM (VOcal SIMulator) technique.
 - New *dcblock2* opcode by Victor Lazzarini.
 - New Chua's oscillator model: *chuap* by Michael Gogins.
 - New *Linear Algebra* opcodes by Michael Gogins. Standard Linear algebra over real and complex vectors and matrices: elementwise arithmetic, norms, transpose and conjugate, inner products, matrix inverse, LU decomposition, QR decomposition, and QR-based eigenvalue decomposition. Includes copying vectors to and from a-rate signals, function tables, and f-signals.
 - New ambisonic opcodes: *bformdec1* and *bformenc1*. These opcodes deprecate the older *bformdec* and *bformenc*.
 - New Score control opcodes by Victor Lazzarini: *rewindscore* and *setscorepos*.
- New functionality:
 - The *vbap* family of opcodes (*vbap4*, *vbap8*, *vbap16* and *vbapz*) now accept k-rate variables for all their input arguments.
 - New pulseaudio I/O module on Linux.
 - New optional *ienv* parameter to generate envelopes for the soundfont opcodes: *sfplay*, *sfplay3*, *sfplaym* and *sfplay3m*.
 - Added 'skip normalisation argument' to "tanh" named GEN routine. (See *Named GEN Routines*)

- Added scheduler priority option on alsa.
- Bug fixes and improvements:
 - Allow scientific notation (as was in csound4!) in *GEN23*.
 - Fixed bug in FLTK initialization. Should make FLTK usage more stable.
 - Error on */* */* comments in orchestra fixed.
 - *poscil* no longer overwrites frequency if variable is shared.
 - *printk* and *printks* check that opcode is initialised.
 - Deprecate *soundout* and *soundouts* in favour of *fout*.
 - Fixed *space* opcode to accept non-pow-2 (deferred) tables.
 - Fixed *pvsmorph* bug.
- Internal Changes:
 - New parser has *#include* and argumentless macros.
 - Less casting between floats and doubles in float version.
 - Includes experimental multicore support.
 - *buzz* opcode rewritten.
 - Many other internal changes and small bug fixes.

New in Version 5.08 (February 2008)

- New opcodes:
 - *imagecreate*, *imagesize*, *imagegetpixel*, *imagesetpixel*, *imagesave*, *imageload* and *imagefree*: New image file processing opcodes by Cesare Marilungo to read/write png images from Csound.
 - *pvsbandp* and *pvsbandr* by John ffitich, which perform band-pass and band-reject filtering in the spectral domain on a pvs signal.
 - New HRTF opcodes by Brian Carty: *hrtfmove*, *hrtfmove2* and *hrtfstat*.
 - New waveshaping opcodes: *powershape*, *polynomial*, *chebyshevpoly*, *pdclip*, *pdhalf*, *pdhalfy*, and *syncphasor*
 - New jack transport control opcode: *jacktransport*
- New functionality
 - Added *--csd-line-nums=* command line option to select mode for error line reporting.
 - New "no-carry" operator (!) for score language that prevents implicit carrying of p-fields in i-statements.
 - Added *--syntax-check-only* commandline flag (exclusive with *--i-only*)
 - *<CsLicence>* tag for CSDs. *<CsLicense>* is accepted as an alternative to *<CsLicence>*.

- Bug fixes and improvements:
 - Changed order of outputs for *hilbert*. This change breaks compatibility with previous versions, but fixes the opcode and now works as documented.
 - Messages about loading opcode plugins modified so can be suppressed with message level flag.
 - Major changes to score error reporting; now accurately reports the line numbers for the chain of inputs for most errors.
 - Corrected *pan2* so it agrees with documentation.
 - *<CsVersion>* tag works again according to the manual.
 - Fixed the { and } score looping statements. Added missing documentation for them and ~, &, |, and # operators in score expressions.
 - *hilbert* had its outputs reversed, now correct. Manual example updated.
- Internal Changes:
 - Change to gettext localisation; French and Columbian-Spanish translations available.
 - Internal changes to *partikkel*, interpolation of waveform read and windowing, allowing more precise pitch synchronous granular synthesis. Updated examples for *partikkel*.
 - *pvscale*: Improved algorithm for SDFT case so no amplitude variation.

New in Version 5.07 (October 2007)

- New opcodes:
 - *pan2*: a stereo panning opcode
 - *cpsmidinn*, *pchmidinn*, *octmidinn*: converters for MIDI note numbers
 - *fluidSetInterpMethod*: interpolation in fluid sound fonts
 - *sflooper*: a soundfont version of *flooper2*
 - *pvsbuffer* and *pvsbufread*: buffering/reading of fsigs for delays/timescale changes.
- New functionality
 - SDFT - the Sliding Discrete Fourier Transform -- added seamlessly to *pvsanal*, etc opcodes if the overlap is less than the ksmps or less than 10. Some pvsXXX opcodes extended to take a-rate parameters when sliding.
 - New feature (*-O null / --logfile=null*) that disables all messages and printing to the console.
- Bug fixes and improvements:
 - *partikkel* -- particle synthesis had an inadvertent bug, now fixed.
 - Closing of MIDI input on Windows(MM) failed; now fixed
 - *fluidEngine* opcode now takes optional number of channels (range 16-256, default to 256) and polyphony (range 16-4096, default to 4096) to use.

- *atsa* utility safer when given silence.
- *ATSaddnz*: improved checking.
- Ambisonics (*bformdec*, *bformenc*) has more options for controlled opposites.
- Bug in *turnoff2* fixed.
- *het_export*: invalid check caused export to fail.
- Internal Changes:
 - Improved Windows installer.
 - CsoundVST replaced by CsoundAC, that does not depend on the VST SDK headers.
 - Less messages in Windows(MM) startup.
 - P argument type added (k-rate defaults to 1) for opcode in and out types.

New in Version 5.06 (June 2007)

- New granular opcodes: *partikkel*, *partikkelsync* and *diskgrain*.
- New opcode for event dispatch: *scoreline*.
- Many new opcodes from Gabriel Maldonado's CsoundAV: *hvs1*, *hvs2*, *hvs3*, *vphaseseg*, *inrg*, *outrg*, *lposcila*, *lposcilsa*, *lposcilsa2*, *tabmorph*, *tabmorpha*, *tabmorphi*, *tabmorphak*, *trandom*, *vt-able1k*, *slider8table*, *slider16table*, *slider32table*, *slider64table*, *slider8tablef*, *slider16tablef*, *slider32tablef*, *slider64tablef*, *sliderKawai* and the a-rate version of *ctrl7*.
- Also from CsoundAV, many new FLTK widget opcodes: *FLkeyIn*, *FLslidBnk2*, *FLvslidBnk*, *FLvslidBnk2*, *FLmouse*, *FLxyin*, *FLhvsBox*, *FLslidBnkSet*, *FLslidBnkSetk*, *FLslidBnk2Set*, *FLslidBnk2Setk*, *FLslidBnkGetHandle*,
- New pvs opcodes: *pvsdiskin*, *pvsmorph*,
- *eqfil*
- New command line options (*--m-warnings*) to control messages
- *csladspa*: a CSD to LADSPA plugin kit.
- And many bug fixes including (but not limited to): fixed k-rate version of *system*; fixed scaling problems of *vrandh* and *vrandi*; fixed occasional failure of *turnoff*; fixed OS X bug; fixed *ATScross* and fixed *mod*.

Csound5GUI now works properly on all platforms and *csoundapi~* (pd object) has been updated.

Part I. Overview

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Introduction

Csound is a unit generator-based, user-programmable computer music system. It was originally written by Barry Vercoe at the Massachusetts Institute of Technology in 1984 as the first C language version of this type of software. Since then Csound has received numerous contributions from researchers, programmers, and musicians from around the world.

Around 1991, John ffitch ported Csound to Microsoft DOS. Csound currently runs on many varieties of UNIX and Linux, Microsoft DOS and Windows, all versions of the Macintosh operating system including Mac OS X, and others.

There are newer computer music systems that have graphical patch editors (e.g. Max/MSP, PD, jMax, or Open Sound World), or that use more advanced techniques of software engineering (e.g. Nyquist or SuperCollider). Yet Csound still has the largest and most varied set of unit generators, is the best documented, runs on the most platforms, and is the easiest to extend. It is possible to compile Csound using double-precision arithmetic throughout for superior sound quality. In short, Csound must be considered one of the most powerful musical instruments ever created.

In addition to this "canonical" version of Csound and CsoundAC, there are other versions of Csound and other front ends for Csound, many of which can be found at <http://csounds.com>.

Recent Developments

In the time since Barry Vercoe wrote the original Preface to this manual, printed above, many further contributions have been made to Csound. CsoundAC is an extended version of Csound 5.

Features of Csound 5

Csound 5 begins a new major version of Csound that includes the following new features:

- Now licensed under the GNU Lesser General Public License, an open source license.
- A new, easier to manage build system using SCons.
- The use of widely--accepted open source libraries:
 - libsndfile for soundfile input and output.
 - PortAudio with ASIO drivers for low-latency, real-time audio input and output.
 - FLTK for graphical widgets that can be programmed in orchestra code.
 - PortMidi for real-time MIDI input and output.

In addition, Istvan Varga has contributed native MIDI and audio drivers for Windows and Linux.

- Simplified audio buffering system.
- Status returns from all internal functions, including opcode functions.
- MIDI interop opcodes, that enable the same instrument definitions to be used interchangeably for either live MIDI performance or off-line, score-driven performance.
- Plugin opcodes are working and becoming more widely accepted. Many opcodes have been moved to plugins. Most new opcodes are plugins, including:
 - The FluidSynth-based SoundFont opcodes.
 - Python opcodes allowing Python code to execute in the orchestra header or in instrument code, at `i-rate` or `k-rate`.
 - Loris opcodes for time/frequency analysis and resynthesis.
 - Control bus opcodes.
 - Audio mixer opcodes.
 - String conversion opcodes.
 - Improved Open Sound Control (OSC) opcodes.
 - Vectorial opcodes.
 - The pvs opcodes for real-time spectral processing, a port of Mark Dolson's phase vocoder code.
 - The ATS opcodes for spectral Analysis, Transformation, and Synthesis of sound based on a sinusoidal plus critical-band noise model. A sound in ATS is a symbolic object representing a spectral model that can be sculpted using a variety of transformation functions. These opcodes can read, transform and resynthesize ATS analysis files. Please note that you need the ATS application to produce analysis files.
 - The STK opcodes, consisting of Perry Cook's original Synthesis Toolkit in C++ instruments, in

C++, adapted as opcodes.

- DSSI and LADSPA adapter opcodes for hosting DSSI and LADSPA plugins in Csound.
- vst4cs VST adapter opcodes for hosting VST plugins in Csound. (Distributed in source form only due to the VST SDK licence restrictions.)
- The `OpcodeBase.hpp` header file for writing plugin opcodes in C++. This is based on the technique of static polymorphism via template inheritance.
- Istvan Varga's `csound5gui` frontend for Csound, simplifying the editing of Csound, the use of Csound especially for live performance, and the monitoring of performances.
- Victor Lazzarini's Tcl/Tk frontends for Csound, `csctlsh` and `cswish`.
- The Csound API is becoming more standardized and more widely used. There are interfaces or wrappers to the API in the following languages:
 - C (include `csound.h`).
 - C++ (include `csound.hpp`). This API includes Csound score and orchestra file container functions.
 - Python (`import csnd`).
 - Java (`import csnd.*`).
 - Lua (`require "csnd"`).
 - Lisp (use the CFFI file `csound5.lisp`).
- Csound is now truly re-entrant, meaning that multiple instances of Csound can run at the same time, in the same process.

John ffitich plans to replace the handwritten parser with one written using a parser generator, which should make it more bug-free and perhaps more efficient.

Features of CsoundAC

CsoundAC is a Python extension module for writing music by programming in Python. CsoundAC is based on Michael Gogins' concept of music graphs, in which a score is represented as hierarchical tree of nodes, which can contain notes, score generators, score transforms, and other nodes.

CsoundAC also provides a Python interface to the Csound API. This makes it very easy to use Csound to render CsoundAC compositions. Using Python's triple quotes, it is even possible to embed the formatted Csound orchestra code for a piece directly into the Python code for that piece, so that all programming for a composition can be maintained in a single file.

The coordinate system in CsoundAC is based on a Euclidean music space with dimensions {time, duration, event type, instrument number, pitch as MIDI key, loudness as MIDI velocity, phase, spatial X coordinate, spatial Y coordinate, spatial Z coordinate, pitch-class set, 1}. A point in music space can be a note, an inflection of a note, or even a grain of sound.

A music graph is a directed acyclical graph, or tree, of nodes in music space. These nodes are associated with local transformations of coordinate system. There are nodes for containing scores or fragments of scores, for generating scores, and for transforming scores. In addition, any node may contain child nodes that inherit the parent's coordinate system.

Thus, it is possible to compose a musical score by containing or generating notes in lower level nodes, assembling them into a score using higher level nodes, and finally rendering the score with Csound. The process is strictly analogous to the construction of a 3-dimensional scene in computer

graphics by generating primitive objects such as spheres, cones, and cubes and moving them around in space to assemble a scene.

Some of the node classes included in CsoundAC are:

- **ScoreNode**: Simply contains a sequence of notes or other points in music space, perhaps imported from a MIDI file.
- **Rescale**: Rescales child points to fit a desired range in time, duration, pitch, and/or other dimensions.
- **Cell**: Repeats child points in a sequence at regular intervals; the interval can be shorter or longer than the actual duration of the child points.
- **Hocket**: Hockets points produced by child nodes.
- **Lindenmayer**: Generates scores using O-L Lindenmayer systems.
- **StrangeAttractor**: Generates scores from a variety of tunable chaotic dynamical systems.
- **MCRM**: Generates scores using the Multiple Copy Reducing Machine algorithm.
- **ImageToScore**: Generates scores by translating image files into points in music space.
- **Random**: Randomizes child points on any dimension or dimensions of music space, using a variety of random variables.
- **VoiceleadingNode**: Generates chord progressions and voice-leading for child notes, using operations based on the mathematical music theory of Dmitri Tymoczko.

Finally, it is possible to derive a new Node class in Python from any existing Node, in order to create new score generators and transforms as part of the composing process.

The Csound Command

Csound is a command to generate a sound output from an *orchestra* file and a *score* file (or a unified *csd* file). It is designed to be called from a terminal or DOS window, but can be called from an easier-to-use *front-end*. The score file can be in one of many different formats, according to user preference. Translation, sorting, and formatting into orchestra-readable numeric text is handled by various preprocessors; all or part of the score is then sent on to the orchestra. Orchestra performance is influenced by *command flags*, which set the level of displays and console reports, specify I/O filenames and sample formats, and declare the nature of real-time sensing and control.

Order of Precedence

There are five places where options for Csound performance may be set. They are processed in the following order:

1. Csound's own defaults
2. File defined by the CSOUNDRC *environment variable*, or .csoundrc file in the HOME directory
3. A .csoundrc file in the current directory
4. <CsOptions> tag in a .csd file
5. Passed on the Csound *command* line

The later options in the list will override any earlier ones. As of version 5.01 of Csound, sample and control rate override flags (-r and -k) specified anywhere override sr, kr, and ksmps defined in the orchestra header.

Description of the command syntax

The csound command is followed by a set of *Command Line Flags* and the name of the orchestra (.orc) and score (.sco) files or the *Unified csd file* (containing both orchestra and score) to process. *Command Line Flags* to control input and output configuration may appear anywhere in the command line, either separately or bundled together. A flag taking a Name or Number will find it in that argument, or in the immediately subsequent one. The following are thus equivalent commands:

```
csound -nm3 orchname -Sxxfilename scorename
csound -n -m 3 orchname -x xfilename -S scorename
```

All flags and names are optional. The default values are:

```
csound -s -otest -b1024 -B1024 -m7 -P128 orchname scorename
```

where *orchname* is a file containing Csound orchestra code, and *scorename* is a file of score data in standard numeric score format, optionally presorted and time-warped. If *scorename* is omitted, there are two default options:

1. if real-time input is expected (e.g. -L, -M, -iadc or -F), a dummy score file is substituted consisting of the single statement 'f 0 3600' (i.e. listen for RT input for one hour)
2. else Csound uses the previously processed *score.srt* in the current directory.

Csound reports on the various stages of score and orchestra processing as it executes, performing various syntax and error checks along the way. Once the actual performance has begun, any error messages will derive from either the instrument loader or the unit generators themselves. A CSound command may include any rational combination of flag arguments.

Running the examples in this manual from the command line

Most of the manual's examples come ready to run without the need of adding any command line flags since they specify options within the csd file's <CsOptions> tag, so you only need to type something like:

```
csound oscil.csd
```

within the examples folder, and realtime audio output should be generated.

Csound command line

csound — Csound command.

Description

The *csound* command executes Csound.

Syntax

```
csound [flags] [orchname] [scorename]
```

```
csound [flags] [csdfilename]
```

Csound command line flags

Listed below are the command line flags available in Csound5 in alphabetical order. Various platform implementations may not react the same way to different flags! You can view the command line flags organized by category in *Command-line Flags (by Category)*.

The command line arguments are of 2 types: *flags* arguments (beginning with a “-”, “--” or “+”), and *name* arguments (such as filenames). Certain flag arguments take a following name or numeric argument. Flags that start with “--” and “+” usually take an argument themselves using “=”.

Command-line Flags

-@ FILE	Provide an extended command-line in file “FILE”
-3, --format=24bit	Use 24-bit audio samples.
-8, --format=uchar	Use 8-bit unsigned character audio samples.
--format=type	Set the audio file output format to one of the formats available in libsndfile. At present the list is aiff, au, avr, caf, flac, htk, ircam, mat4, mat5, nis, paf, pvf, raw, sd2, sds, svx, voc, w64, wav, wavex and xi. Can also be used as -format=type:format or --format=format:type to set both the file type (wav, aiff, etc.) and sample format (short, long, float, etc.) at the same time.
-A, --aiff, --format=aiff	Write an AIFF format soundfile. Use with the -c, -s, -l, or -f flags.
-a, --format=alaw	Use a-law audio samples.
-B NUM, -hardwarebufsamps=NUM	Number of audio sample-frames held in the DAC <i>hardware</i> buffer. This is a threshold on which <i>software</i> audio I/O (above) will wait before returning. A small number reduces audio I/O delay; but the value is often hardware limited, and small values will risk data lates. In the case of portaudio output (the default real-time output), the -B parameter (more precisely, -B / sr) is passed as the “suggested latency” value. Other than that, Csound has no control over how PortAudio interprets the parameter. The default is 1024 on Linux, 4096 on Mac OS X and 16384 on Windows.

<code>-b NUM, --iobufsamps=NUM</code>	<p>Number of audio sample-frames per sound i/o <i>software</i> buffer. Large is efficient, but small will reduce audio I/O delay and improve the accuracy of the timing of real time events. The default is 256 on Linux, 1024 on MacOS X, and 4096 on Windows. In real-time performance, Csound waits on audio I/O on <i>NUM</i> boundaries. It also processes audio (and polls for other input like MIDI) on orchestra <i>ksmps</i> boundaries. The two can be made synchronous. For convenience, if <i>NUM</i> is negative, the effective value is <i>ksmps</i> * <i>-NUM</i> (audio synchronous with <i>k</i>-period boundaries). With <i>NUM</i> small (e.g. 1) polling is then frequent and also locked to fixed DAC sample boundaries.</p> <p>Note: if both <code>-iadc</code> and <code>-odac</code> are used at the same time (full duplex real time audio), the <code>-b</code> option should be set to an integer multiple of <i>ksmps</i>.</p>
<code>-C, --cscore</code>	Use Cscore processing of the scorefile.
<code>-c, --format=schar</code>	Use 8-bit signed character audio samples.
<code>--csd-line-nums=NUM</code>	<p>Determines how line numbers are counted and displayed for error messages when processing a Csound Unified Document file (.csd). This flag has no effect if separate orchestra and score files are used. (Csound 5.08 and later).</p> <ul style="list-style-type: none"> • 0 = line numbers are relative to the beginning of the orchestra or score sections of the CSD • 1 = line numbers are relative to the beginning of the CSD file. This is the default as of Csound 5.08.
<code>-D, --defer-gen1</code>	Defer GEN01 soundfile loads until performance time.
<code>-d, --nodisplays</code>	Suppress all displays.
<code>--displays</code>	Enables displays, reverting the effect of any previous <code>-d</code> flag.
<code>--default-paths</code>	Reenables adding of directory of CSD/ORC/SCO to search paths, if it has been disabled by a previous <code>--no-default-paths</code> (e.g. in <i>.csoundrc</i>).
<code>--env:NAME=VALUE</code>	Set environment variable <i>NAME</i> to <i>VALUE</i> . Note: not all environment variables can be set this way, because some are read before parsing the command line. <i>INCDIR</i> , <i>SADIR</i> , <i>SFDIR</i> , and <i>SSDIR</i> are known to work.
<code>--env:NAME+=VALUE</code>	Append <i>VALUE</i> to ';' separated list of search paths in environment variable <i>NAME</i> (should be <i>INCDIR</i> , <i>SADIR</i> , <i>SFDIR</i> , or <i>SSDIR</i>). If a file is found in multiple directories, the last will be used.
<code>--expression-opt</code>	<p>Since Csound 5. Turns on some optimizations in expressions:</p> <ul style="list-style-type: none"> • Redundant assignment operations are eliminated whenever possible. This means that for example this line <i>a1 = a2 + a3</i> will compile as <i>a1 Add a2, a3</i> instead of <i>#a0 Add a2, a3 a1 = #a0</i> saving a temporary variable and an opcode call. Less opcode calls result in reduced CPU usage (an average orchestra may compile about 10% faster with <code>-expression-opt</code>, but it depends largely on how many expressions are used, what the control rate is (see also below), etc.; thus, the difference may be less, but also much

more).

- number of a- and k-rate temporary variables is significantly reduced. This expression

```
(a1 + a2 + a3 + a4)
```

will compile as

```
#a0 Add a1, a2
#a0 Add #a0, a3
#a0 Add #a0, a4      ; (the result is in #a0)
```

instead of

```
#a0 Add a1, a2
#a1 Add #a0, a3
#a2 Add #a1, a4      ; (the result is in #a2)
```

The advantages of less temporary variables are:

- less cache memory is used, which may improve performance of orchestras with many a-rate expressions and a low control rate (e.g. ksmps = 100)
- large orchestras may load faster due to less different identifier names
- index overflow errors (i.e. when messages like this Case2: indx=-56004 (ffff253c); (short)indx = 9532 (253c) are printed and odd behavior or a Csound crash occurs) may be fixed, because such errors are triggered by too many different (especially a-rate) variable names in a single instrument.

Note that this optimization (due to technical reasons) is not performed on i-rate temporary variables.



Warning

When `--expression-opt` is turned on, it is not allowed to use the `i()` function with an expression argument, and relying on the value of k-rate expressions at i-time is unsafe.

`-F FILE, --midifile=FILE`

Read MIDI events from MIDI file *FILE*. The file should have only one track in Csound versions 4.xx and earlier; this limitation is removed in Csound 5.00.

`-f, --format=float`

Use single-format float audio samples (not playable on some systems, but can be read by *-i, soundin* and *GEN01*

`-G, --postscriptdisplay`

Suppress graphics, use PostScript displays instead.

`-g, --asciideisplay`

Suppress graphics, use ASCII displays instead.

`-H#, --heartbeat=NUM`

Print a heartbeat after each soundfile buffer write:

- no NUM, a rotating bar.

	<ul style="list-style-type: none"> • NUM = 1, a rotating bar. • NUM = 2, a dot (.) • NUM = 3, filesize in seconds. • NUM = 4, sound a bell.
-h, --noheader	No header on output soundfile. Don't write a file header, just binary samples.
--help	Display on-line help message.
-I, --i-only	<i>i-time only</i> . Allocate and initialize all instruments as per the score, but skip all p-time processing (no k-signals or a-signals, and thus no amplitudes and no sound). Provides a fast validity check of the score pfields and orchestra i-variables. This option is exclusive of the --syntax-check-only flag.
-i FILE, --input=FILE	Input soundfile name. If not a full pathname, the file will be sought first in the current directory, then in that given by the environment variable SSDIR (if defined), then by SFDIR. The name <i>stdin</i> will cause audio to be read from standard input. The name <i>devaudio</i> or <i>adc</i> will request sound from the host audio input device. It is possible to select a device number by appending an integer value in the range 0 to 1023, or a device name separated by a : character (e.g. -iadc3, -iadc:hw:1,1). It depends on the host audio interface whether a device number or a name should be used. In the first case, an out of range number usually results in an error and listing the valid device numbers.
--id_artist=string	(max. length = 200 characters) Artist tag in output soundfile (no spaces)
--id_comment=string	(max. length = 200 characters) Comment tag in output soundfile (no spaces)
--id_copyright=string	(max. length = 200 characters) Copyright tag in output soundfile (no spaces)
--id_date=string	(max. length = 200 characters) Date tag in output soundfile (no spaces)
--id_software=string	(max. length = 200 characters) Software tag in output soundfile (no spaces)
--id_title=string	(max. length = 200 characters) Title tag in output soundfile (no spaces)
--ignore_csopts=integer	If set to 1, Csound will ignore all options specified in the csd file's CsOptions section. See <i>Unified File Format for Orchestras and Scores</i> .
--input_stream=string	Pulseaudio input stream name.
-J, --ircam, --format=ircam	Write an IRCAM format soundfile.
--jack_client=[client_name]	The client name used by Csound, defaults to 'csound5'. If multiple instances of Csound connect to the JACK server, different client names need to be used to avoid name conflicts.

(Linux and Mac OS X only)

--jack_inportname=[input port
name prefix], -
+jack_outportname=[output port
name prefix]

Name prefix of Csound JACK input/output ports; the default is 'input' and 'output'. The actual port name is the channel number appended to the name prefix. (Linux and Mac OS X only)

Example: with the above default settings, a stereo orchestra will create these ports in full duplex operation:

```
csound5:input1      (record left)
csound5:input2      (record right)
csound5:output1     (playback left)
csound5:output2     (playback right)
```

-K, --nopeaks

Do not generate any PEAK chunks.

-k NUM, --control-rate=NUM

Override the control rate (*KR*) supplied by the orchestra.

-L DEVICE, --score-in=DEVICE

Read line-oriented real-time score events from device *DEVICE*. The name *stdin* will permit score events to be typed at your terminal, or piped from another process. Each line-event is terminated by a carriage-return. Events are coded just like those in a *standard numeric score*, except that an event with *p2=0* will be performed immediately, and an event with *p2=T* will be performed *T* seconds after arrival. Events can arrive at any time, and in any order. The score *carry* feature is legal here, as are held notes (*p3* negative) and string arguments, but ramps and *pp* or *np* references are not.

-l, --format=long

Use long integer audio samples.

-M DEVICE, -
-midi-device=DEVICE

Read MIDI events from device *DEVICE*. If using ALSA MIDI (--rtmidi=alsa), devices are selected by name and not number. So, you need to use an option like -M hw:CARD,DEVICE where *CARD* and *DEVICE* are the card and device numbers (e.g. -M hw:1,0). In the case of PortMidi and MME, *DEVICE* should be a number, and if it is out of range, an error occurs and the valid device numbers are printed.

-m NUM, --messagelevel=NUM

Message level for standard (terminal) output. Takes the *sum* of any of the following values:

- 1 = note amplitude messages
 - 2 = samples out of range message
 - 4 = warning messages
 - 128 = print benchmark information
- And exactly one of these to select note amplitude format:
- 0 = raw amplitudes, no colours
 - 32 = dB, no colors
 - 64 = dB, out of range highlighted with red
 - 96 = dB, all colors
 - 256 = raw, out of range highlighted with red
 - 512 = raw, all colours

The default is 135 (128+4+2+1), which means all messages, raw amplitude values, and printing elapsed time at the end of performance. The coloring of raw amplitudes was introduced in version 5.04.

<code>--m-amps=NUM</code>	<p>Message level for amplitudes on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no note amplitude messages • 1 = note amplitude messages
<code>--m-range=NUM</code>	<p>Message level for out of range messages on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no samples out of range message • 1 = samples out of range message
<code>--m-warnings=NUM</code>	<p>Message level for warnings on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no warning messages • 1 = warning messages
<code>--m-dB=NUM</code>	<p>Message level for amplitude format on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = absolute amplitude messages • 1 = dB amplitude messages
<code>--m-colours=NUM</code>	<p>Message level for amplitude format on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no colouring of amplitude messages • 1 = colouring of amplitude messages
<code>--m-benchmarks=NUM</code>	<p>Message level for benchmark information on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no benchnark numbers • 1 = print benchnark numbers
<code>++max_str_len=integer</code>	<p>(min: 10, max: 10000) Maximum length of string variables + 1; defaults to 256 allowing a length of 255 characters. The length of string constants is not limited by this parameter.</p>
<code>--midi-key=N</code>	<p>Route MIDI note on message key number to pfield N as MIDI value [0-127].</p>
<code>--midi-key-cps=N</code>	<p>Route MIDI note on message key number to pfield N as cycles per second.</p>
<code>--midi-key-oct=N</code>	<p>Route MIDI note on message key number to pfield N as linear octave.</p>
<code>--midi-key-pch=N</code>	<p>Route MIDI note on message key number to pfield N as oct.pch (pitch class).</p>
<code>--midi-velocity=N</code>	<p>Route MIDI note on message velocity number to pfield N as MIDI value [0-127].</p>

<code>--midi-velocity-amp=N</code>	Route MIDI note on message velocity number to pfield N as amplitude [0-0dbFS].
<code>--midioutfile=FILENAME</code>	Save MIDI output to a file (Csound 5.00 and later only).
<code>--msg_color=boolean</code>	Enable message attributes (colors etc.); might need to be disabled on some terminals which print strange characters instead of modifying text attributes. default: true.
<code>--mute_tracks=string</code>	(max. length = 255 characters) Ignore events (other than tempo changes) in MIDI file tracks defined by pattern (for example, <code>--mute_tracks=00101</code> will mute the third and fifth tracks).
<code>-N, --notify</code>	Notify (ring the bell) when score or MIDI track is done.
<code>-n, --nosound</code>	No sound. Do all processing, but bypass writing of sound to disk. This flag does not change the execution in any other way.
<code>--no-default-paths</code>	Disables adding of directory of CSD/ORC/SCO to search paths.
<code>--no-expression-opt</code>	Disables expression optimization.
<code>-O FILE, --logfile=FILE</code>	Log output to file <i>FILE</i> . If <i>FILE</i> is null (i.e. <code>-O null</code> or <code>--logfile=null</code>) all printing of messages to the console is disabled.
<code>-o FILE, --output=FILE</code>	Output soundfile name. If not a full pathname, the soundfile will be placed in the directory given by the environment variable <i>SFDIR</i> (if defined), else in the current directory. The name <i>stdout</i> will cause audio to be written to standard output, while <i>null</i> results in no sound output similarly to the <code>-n</code> flag. If no name is given, the default name will be <i>test</i> . The name <i>devaudio</i> or <i>dac</i> (you can use <code>-odac</code> or <code>-o dac</code>) will request writing sound to the host audio output device. It is possible to select a device number by appending an integer value in the range 0 to 1023, or a device name separated by a <code>:</code> character (e.g. <code>-odac3</code> , <code>-odac:hw:1,1</code>). It depends on the host audio interface whether a device number or a name should be used. In the first case, an out of range number usually results in an error and listing the valid device numbers.
<code>--omacro:XXX=YYY</code>	Set orchestra macro XXX to value YYY
<code>--output_stream=string</code>	Pulseaudio output stream name.
<code>-Q DEVICE</code>	Enables MIDI OUT operations to device id <i>DEVICE</i> . This flag allows parallel MIDI OUT and DAC performance. Unfortunately the real-time timing implemented in Csound is completely managed by DAC buffer sample flow. So MIDI OUT operations can present some time irregularities. These irregularities can be reduced by using a lower value for the <code>-b</code> flag. If using ALSA MIDI (<code>--rtmidi=alsa</code>), devices are selected by name and not number. So, you need to use an option like <code>-Q hw:CARD,DEVICE</code> where CARD and DEVICE are the card and device numbers (e.g. <code>-Q hw:1,0</code>). In the case of PortMidi and MME, DEVICE should be a number, and if it is out of range, an error occurs and the valid device numbers are prin-

	ted.
-R, --rewrite	Continually rewrite the header while writing the soundfile (WAV/AIFF).
-r NUM, --sample-rate=NUM	Override the sampling rate (<i>SR</i>) supplied by the orchestra.
--raw_controller_mode=boolean	Disable special handling of MIDI controllers like sustain pedal, all notes off etc., allowing the use of all the 128 controllers for any purpose. This will also set the initial value of all controllers to zero. Default: no.
--rtaudio=string	(max. length = 20 characters) Real time audio module name. The default is PortAudio. Also available, depending on platform and build options: Linux: alsa, jack; Windows: mme; Mac OS X: CoreAudio. In addition, null can be used on all platforms, to disable the use of any real time audio plugin.
--rtmidi=string	(max. length = 20 characters) Real time MIDI module name. Defaults to PortMidi, other options (depending on build options): Linux: alsa; Windows: mme, winmm. In addition, null can be used on all platforms, to disable the use of any real time MIDI plugin. ALSA MIDI devices are selected by name and not number. So, you need to use an option like -M hw:CARD,DEVICE where CARD and DEVICE are the card and device numbers (e.g. -M hw:1,0).
-s, --format=short	Use short integer audio samples.
--sched	<i>Linux only.</i> Use real-time scheduling and lock memory. (Also requires -d and either -o dac or -o devaudio). See also -sched=N below.
--sched=N	<i>Linux only.</i> Same as --sched, but allows specifying a priority value: if N is positive (in the range 1 to 99) the scheduling policy SCHED_RR will be used with a priority of N; otherwise, SCHED_OTHER is used with the nice level set to N. Can also be used in the format --sched=N,MAXCPU,TIME to enable the use of a "watchdog" thread that terminates Csound if the average CPU usage exceeds MAXCPU percents over a period of TIME seconds (new in Csound 5.00).
--server=string	Pulseaudio server name.
--skip_seconds=float	(min: 0) Start playback at the specified time (in seconds), skipping earlier events in the score and MIDI file.
--smacro:XXX=YYY	Set score macro XXX to value YYY
--strset	<i>Csound 5.</i> The --strset option allows setting strset string values from the command line, in the format '--strsetN=VALUE'. It is useful for passing parameters to the orchestra (e.g. file names).
--syntax-check-only	Causes Csound to exit immediately after the orchestra and score parsers finish checking the syntax of the input files and before the orchestra performs the score. This option is exclusive of the --i-only flag. (Csound 5.08 and later).
-T, --terminate-on-midi	Terminate the performance when the end of MIDI file is

	reached.
-t0, --keep-sorted-score	Prevents Csound from deleting the sorted score file, <i>score.srt</i> , upon exit.
-t NUM, --tempo=NUM	Use the uninterpreted beats of <i>score.srt</i> for this performance, and set the initial tempo at <i>NUM</i> beats per minute. When this flag is set, the tempo of score performance is also controllable from within the orchestra. WARNING: this mode of operation is experimental and may be unreliable.
-U UTILITY, --utility=UTILITY	Invoke the utility program <i>UTILITY</i> . Use any invalid name to list the available utilities.
-u, --format=ulaw	Use u-law audio samples.
-v, --verbose	Verbose translate and run. Prints details of orch translation and performance, enabling errors to be more clearly located.
-W, --wave, --format=wave	Write a WAV format soundfile.
-x FILE, --extract-score=FILE	Extract a portion of the sorted score, <i>score.srt</i> , using the extract file <i>FILE</i> (see <i>Extract</i>).
-Z, --dither	Switch on dithering of audio conversion from internal floating point to 32, 16 and 8-bit formats. The default form of the dither is triangular.
-Z, --dither--triangular, - -dither--uniform	Switch on dithering of audio conversion from internal floating point to 32, 16 and 8-bit formats. In the case of -Z the next digit should be a 1 (for triangular) or a 2 (for uniform). The exact interpretation depends on the output system.
-z NUM, --list-opcodesNUM	List opcodes in this version: <ul style="list-style-type: none"> • no NUM, just show names • NUM = 0, just show names • NUM = 1, show arguments to each opcode using the format <opname> <outargs> <inargs>

Command-line Flags (by Category)

Listed below are the command line available in Csound5 organized by categories. Various platform implementations may not react the same way to different flags!

You can view the command line flags organized alphabetically in *Command-line Flags (Alphabetically)*.

The format of a command is either:

```
csound [flags] [orchname] [scorename]
or
```

```
csound [flags] [csdfilename]
```

where the arguments are of 2 types: *flags* arguments (beginning with a “-”, “--” or “-+”), and *name* arguments (such as filenames). Certain flag arguments take a following name or numeric argument. Flags that start with “--” and “-+” usually take an argument themselves using “=”.

Audio File Output

-3, --format=24bit	Use 24-bit audio samples.
-8, --format=uchar	Use 8-bit unsigned character audio samples.
-A, --aiff, --format=aiff	Write an AIFF format soundfile. Use with the <i>-c</i> , <i>-s</i> , <i>-l</i> , or <i>-f</i> flags.
-a, --format=alaw	Use a-law audio samples.
-c, --format=schar	Use 8-bit signed character audio samples.
-f, --format=float	Use single-format float audio samples (not playable on some systems, but can be read by <i>-i</i> , <i>soundin</i> and <i>GEN01</i>
--format=type	Set the audio file output format to one of the formats available in libsndfile. At present the list is aiff, au, avr, caf, flac, htk, ircam, mat4, mat5, nis, paf, pvf, raw, sd2, sds, svx, voc, w64, wav, wavex and xi. Can also be used as <i>-format=type:format</i> or <i>--format=format:type</i> to set both the file type (wav, aiff, etc.) and sample format (short, long, float, etc.) at the same time.
-h, --noheader	No header on output soundfile. Don't write a file header, just binary samples.
-i FILE, --input=FILE	Input soundfile name. If not a full pathname, the file will be sought first in the current directory, then in that given by the environment variable SSDIR (if defined), then by SFDIR. The name <i>stdin</i> will cause audio to be read from standard input. The name <i>devaudio</i> or <i>adc</i> will request sound from the host audio input device. It is possible to select a device number by appending an integer value in the range 0 to 1023, or a device name separated by a : character. It depends on the host audio interface whether a device number or a name should be used. In the first case, an out of range number usually results in an error and listing the valid device numbers.
-J, --ircam, --format=ircam	Write an IRCAM format soundfile.
-K, --nopeaks	Do not generate any PEAK chunks.
-l, --format=long	Use long integer audio samples.
-n, --nosound	No sound. Do all processing, but bypass writing of sound to disk. This flag does not change the execution in any other way.
-o FILE, --output=FILE	Output soundfile name. If not a full pathname, the soundfile will be placed in the directory given by the environment variable SFDIR (if defined), else in the current directory. The name <i>stdout</i> will cause audio to be written to standard output, while <i>null</i> results in no sound output similarly to the <i>-n</i> flag. If no name is given, the default name will be <i>test</i> . The name <i>dac</i> or <i>devaudio</i> (you can use <i>-odac</i> or <i>-o dac</i>) will request writing sound to the host audio output device. It is possible to select a device number by appending an integer

	value in the range 0 to 1023, or a device name separated by a : character. It depends on the host audio interface whether a device number or a name should be used. In the first case, an out of range number usually results in an error and listing the valid device numbers.
-R, --rewrite	Continually rewrite the header while writing the soundfile (WAV/AIFF).
-s, --format=short	Use short integer audio samples.
-u, --format=ulaw	Use u-law audio samples.
-W, --wave, --format=wave	Write a WAV format soundfile.
-Z, --dither	Switch on dithering of audio conversion from internal floating point to 32, 16 and 8-bit formats. The default form of the dither is triangular.
-Z, --dither--triangular, -dither--uniform	Switch on dithering of audio conversion from internal floating point to 32, 16 and 8-bit formats. In the case of -Z the next digit should be a 1 (for triangular) or a 2 (for uniform). The exact interpretation depends on the output system.

Output File Id tags

--id_artist=string	(max. length = 200 characters) Artist tag in output soundfile (no spaces)
--id_comment=string	(max. length = 200 characters) Comment tag in output soundfile (no spaces)
--id_copyright=string	(max. length = 200 characters) Copyright tag in output soundfile (no spaces)
--id_date=string	(max. length = 200 characters) Date tag in output soundfile (no spaces)
--id_software=string	(max. length = 200 characters) Software tag in output soundfile (no spaces)
--id_title=string	(max. length = 200 characters) Title tag in output soundfile (no spaces)

Realtime Audio Input/Output

-i adc[DEVICE], -input=adc[DEVICE]	The name <i>devaudio</i> or <i>adc</i> will request sound from the host audio input device. It is possible to select a device number by appending an integer value in the range 0 to 1023, or a device name separated by a : character (e.g. -iadc3, -iadc:hw:1,1). It depends on the host audio interface whether a device number or a name should be used. In the first case, an out of range number usually results in an error and listing the valid device numbers.
-o dac[DEVICE], -output=dac[DEVICE]	The name <i>dac</i> or <i>devaudio</i> (you can use <i>-odac</i> or <i>-o dac</i>) will request writing sound to the host audio output device. It is possible to select a device number by appending an integer

	value in the range 0 to 1023, or a device name separated by a : character (e.g. -odac3, -odac:hw:1,1). It depends on the host audio interface whether a device number or a name should be used. In the first case, an out of range number usually results in an error and listing the valid device numbers.
-+rtaudio=string	(max. length = 20 characters) Real time audio module name. The default is PortAudio (all platforms). Also available, depending on platform and build options: Linux: alsa, jack; Windows: mme; Mac OS X: CoreAudio. In addition, null can be used on all platforms, to disable the use of any real time audio plugin.
-+server=string	Pulseaudio server name.
-+output_stream=string	Pulseaudio output stream name.
-+input_stream=string	Pulseaudio input stream name.
-+jack_client=[client_name]	The client name used by Csound, defaults to 'csound5'. If multiple instances of Csound connect to the JACK server, different client names need to be used to avoid name conflicts. (Linux and Mac OS X only)
-+jack_inportname=[input port name prefix], - +jack_outportname=[output port name prefix]	Name prefix of Csound JACK input/output ports; the default is 'input' and 'output'. The actual port name is the channel number appended to the name prefix. (Linux and Mac OS X only)

Example: with the above default settings, a stereo orchestra will create these ports in full duplex operation:

```
csound5:input1      (record left)
csound5:input2      (record right)
csound5:output1     (playback left)
csound5:output2     (playback right)
```

MIDI File Input/Output

-F FILE, --midifile=FILE	Read MIDI events from MIDI file <i>FILE</i> . The file should have only one track in Csound versions 4.xx and earlier; this limitation is removed in Csound 5.00.
--midioutfile=FILENAME	Save MIDI output to a file (Csound 5.00 and later only).
-+mute_tracks=string	(max. length = 255 characters) Ignore events (other than tempo changes) in MIDI file tracks defined by pattern (for example, -+mute_tracks=00101 will mute the third and fifth tracks).
-+raw_controller_mode=boolean	Disable special handling of MIDI controllers like sustain pedal, all notes off etc., allowing the use of all the 128 controllers for any purpose. This will also set the initial value of all controllers to zero. Default: no.
-+skip_seconds=float	(min: 0) Start playback at the specified time (in seconds), skipping earlier events in the score and MIDI file.
-T, --terminate-on-midi	Terminate the performance when the end of MIDI file is reached.

MIDI Realtime Input/Ouput

<code>-M DEVICE, - -midi-device=DEVICE</code>	Read MIDI events from device <i>DEVICE</i> . If using ALSA MIDI (<code>-+rtmidi=alsa</code>), devices are selected by name and not number. So, you need to use an option like <code>-M hw:CARD,DEVICE</code> where <i>CARD</i> and <i>DEVICE</i> are the card and device numbers (e.g. <code>-M hw:1,0</code>). In the case of PortMidi and MME, <i>DEVICE</i> should be a number, and if it is out of range, an error occurs and the valid device numbers are printed.
<code>--midi-key=N</code>	Route MIDI note on message key number to pfield <i>N</i> as MIDI value [0-127].
<code>--midi-key-cps=N</code>	Route MIDI note on message key number to pfield <i>N</i> as cycles per second.
<code>--midi-key-oct=N</code>	Route MIDI note on message key number to pfield <i>N</i> as linear octave.
<code>--midi-key-pch=N</code>	Route MIDI note on message key number to pfield <i>N</i> as oct.pch (pitch class).
<code>--midi-velocity=N</code>	Route MIDI note on message velocity number to pfield <i>N</i> as MIDI value [0-127].
<code>--midi-velocity-amp=N</code>	Route MIDI note on message velocity number to pfield <i>N</i> as amplitude [0-0dbFS].
<code>--midioutfile=FILENAME</code>	Save MIDI output to a file (Csound 5.00 and later only).
<code>-+rtmidi=string</code>	(max. length = 20 characters) Real time MIDI module name. Defaults to PortMidi, other options (depending on build options): Linux: alsa; Windows: mme, winmm. In addition, null can be used on all platforms, to disable the use of any real time MIDI plugin. ALSA MIDI devices are selected by name and not number. So, you need to use an option like <code>-M hw:CARD,DEVICE</code> where <i>CARD</i> and <i>DEVICE</i> are the card and device numbers (e.g. <code>-M hw:1,0</code>).
<code>-Q DEVICE</code>	Enables MIDI OUT operations to device id <i>DEVICE</i> . This flag allows parallel MIDI OUT and DAC performance. Unfortunately the real-time timing implemented in Csound is completely managed by DAC buffer sample flow. So MIDI OUT operations can present some time irregularities. These irregularities can be reduced by using a lower value for the <i>-b</i> flag. If using ALSA MIDI (<code>-+rtmidi=alsa</code>), devices are selected by name and not number. So, you need to use an option like <code>-Q hw:CARD,DEVICE</code> where <i>CARD</i> and <i>DEVICE</i> are the card and device numbers (e.g. <code>-Q hw:1,0</code>). In the case of PortMidi and MME, <i>DEVICE</i> should be a number, and if it is out of range, an error occurs and the valid device numbers are printed.

Display

<code>--csd-line-nums=NUM</code>	<p>Determines how line numbers are counted and displayed for error messages when processing a Csound Unified Document file (.csd). This flag has no effect if separate orchestra and score files are used. (Csound 5.08 and later).</p> <ul style="list-style-type: none"> • 0 = line numbers are relative to the beginning of the orchestra or score sections of the CSD • 1 = line numbers are relative to the beginning of the CSD file. This is the default as of Csound 5.08.
<code>-d, --nodisplays</code>	Suppress all displays.
<code>--displays</code>	Enables displays, reverting the effect of any previous <code>-d</code> flag.
<code>-G, --postscriptdisplay</code>	Suppress graphics, use PostScript displays instead.
<code>-g, --asciidisplay</code>	Suppress graphics, use ASCII displays instead.
<code>-H#, --heartbeat=NUM</code>	<p>Print a heartbeat after each soundfile buffer write:</p> <ul style="list-style-type: none"> • no NUM, a rotating bar. • NUM = 1, a rotating bar. • NUM = 2, a dot (.) • NUM = 3, filesize in seconds. • NUM = 4, sound a bell.
<code>-m NUM, --messagelevel=NUM</code>	<p>Message level for standard (terminal) output. Takes the <i>sum</i> of any of the following values:</p> <ul style="list-style-type: none"> • 1 = note amplitude messages • 2 = samples out of range message • 4 = warning messages • 128 = print benchmark information <p>And exactly one of these to select note amplitude format:</p> <ul style="list-style-type: none"> • 0 = raw amplitudes, no colours • 32 = dB, no colors • 64 = dB, out of range highlighted with red • 96 = dB, all colors • 256 = raw, out of range highlighted with red • 512 = raw, all colours <p>The default is 135 (128+4+2+1), which means all messages, raw amplitude values, and printing elapsed time at the end of performance. The coloring of raw amplitudes was introduced in version 5.04</p>
<code>--m-amps=NUM</code>	<p>Message level for amplitudes on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no note amplitude messages • 1 = note amplitude messages

<code>--m-range=NUM</code>	<p>Message level for out of range messages on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no samples out of range message • 1 = samples out of range message
<code>--m-warnings=NUM</code>	<p>Message level for warnings on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no warning messages • 1 = warning messages
<code>--m-dB=NUM</code>	<p>Message level for amplitude format on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = absolute amplitude messages • 1 = dB amplitude messages
<code>--m-colours=NUM</code>	<p>Message level for amplitude format on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no colouring of amplitude messages • 1 = colouring of amplitude messages
<code>--m-benchmarks=NUM</code>	<p>Message level for benchmark information on standard (terminal) output.</p> <ul style="list-style-type: none"> • 0 = no benchnark numbers • 1 = print benchnark numbers
<code>--msg_color=boolean</code>	<p>Enable message attributes (colors etc.); might need to be disabled on some terminals which print strange characters instead of modifying text attributes. default: true.</p>
<code>-v, --verbose</code>	<p>Verbose translate and run. Prints details of orch translation and performance, enabling errors to be more clearly located.</p>
<code>-z NUM, --list-opcodesNUM</code>	<p>List opcodes in this version:</p> <ul style="list-style-type: none"> • no NUM, just show names • NUM = 0, just show names • NUM = 1, show arguments to each opcode using the format <opname> <outargs> <inargs>

Performance Configuration and Control

<code>-B NUM, -hardwarebufsamps=NUM</code>	<p>Number of audio sample-frames held in the DAC <i>hardware</i> buffer. This is a threshold on which <i>software</i> audio I/O (above) will wait before returning. A small number reduces audio I/O delay; but the value is often hardware limited, and small values will risk data lates. In the case of portaudio output (the default real-time output), the <code>-B</code> parameter (more precisely, <code>-B / sr</code>) is passed as the "suggested latency" value. Other than that, Csound has no control over how PortAudio</p>
--	--

	interprets the parameter. The default is 1024 on Linux, 4096 on Mac OS X and 16384 on Windows.
-b NUM, --iobufsamps=NUM	Number of audio sample-frames per sound i/o <i>software</i> buffer. Large is efficient, but small will reduce audio I/O delay and improve the accuracy of the timing of real time events. The default is 256 on Linux, 1024 on MacOS X, and 4096 on Windows. In real-time performance, Csound waits on audio I/O on <i>NUM</i> boundaries. It also processes audio (and polls for other input like MIDI) on orchestra <i>ksmps</i> boundaries. The two can be made synchronous. For convenience, if NUM is negative, the effective value is <i>ksmps</i> * -NUM (audio synchronous with k-period boundaries). With NUM small (e.g. 1) polling is then frequent and also locked to fixed DAC sample boundaries.
	Note: if both -iadc and -odac are used at the same time (full duplex real time audio), the -b option should be set to an integer multiple of <i>ksmps</i> .
-k NUM, --control-rate=NUM	Override the control rate (<i>KR</i>) supplied by the orchestra.
-L DEVICE, --score-in=DEVICE	Read line-oriented real-time score events from device <i>DEVICE</i> . The name <i>stdin</i> will permit score events to be typed at your terminal, or piped from another process. Each line-event is terminated by a carriage-return. Events are coded just like those in a <i>standard numeric score</i> , except that an event with p2=0 will be performed immediately, and an event with p2=T will be performed T seconds after arrival. Events can arrive at any time, and in any order. The score <i>carry</i> feature is legal here, as are held notes (p3 negative) and string arguments, but ramps and <i>pp</i> or <i>np</i> references are not.
--omacro:XXX=YYY	Set orchestra macro XXX to value YYY
-r NUM, --sample-rate=NUM	Override the sampling rate (<i>SR</i>) supplied by the orchestra.
--sched	<i>Linux only.</i> Use real-time scheduling and lock memory. (Also requires -d and either -o <i>dac</i> or -o <i>devaudio</i>). See also -sched=N below.
--sched=N	<i>Linux only.</i> Same as --sched, but allows specifying a priority value: if N is positive (in the range 1 to 99) the scheduling policy SCHED_RR will be used with a priority of N; otherwise, SCHED_OTHER is used with the nice level set to N. Can also be used in the format --sched=N,MAXCPU,TIME to enable the use of a "watchdog" thread that terminates Csound if the average CPU usage exceeds MAXCPU percents over a period of TIME seconds (new in Csound 5.00).
--smacro:XXX=YYY	Set score macro XXX to value YYY
--strset	<i>Csound 5.</i> The --strset option allows setting strset string values from the command line, in the format '--strsetN=VALUE'. It is useful for passing parameters to the orchestra (e.g. file names).
++skip_seconds=float	(min: 0) Start playback at the specified time (in seconds), skipping earlier events in the score and MIDI file.
-t NUM, --tempo=NUM	Use the uninterpreted beats of <i>score.srt</i> for this performance, and set the initial tempo at <i>NUM</i> beats per minute. When this flag is set, the tempo of score performance is also controllable

from within the orchestra. WARNING: this mode of operation is experimental and may be unreliable.

Miscellaneous

<code>-@ FILE</code>	Provide an extended command-line in file “FILE”
<code>-C, --cscore</code>	Use Cscore processing of the scorefile.
<code>--default-paths</code>	Reenables adding of directory of CSD/ORC/SCO to search paths, if it has been disabled by a previous <code>--no-default-paths</code> (e.g. in <code>.csoundrc</code>).
<code>-D, --defer-genl</code>	Defer GEN01 soundfile loads until performance time.
<code>--env:NAME=VALUE</code>	Set environment variable NAME to VALUE. Note: not all environment variables can be set this way, because some are read before parsing the command line. INCDIR, SADIR, SFDIR, and SSDIR are known to work.
<code>--env:NAME+=VALUE</code>	Append VALUE to ';' separated list of search paths in environment variable NAME (should be INCDIR, SADIR, SFDIR, or SSDIR). If a file is found in multiple directories, the last will be used.
<code>--expression-opt</code>	<p><i>Since Csound 5.</i> Turns on some optimizations in expressions:</p> <ul style="list-style-type: none"> • Redundant assignment operations are eliminated whenever possible. This means that for example this line <code>a1 = a2 + a3</code> will compile as <code>a1 Add a2, a3</code> instead of <code>#a0 Add a2, a3 a1 = #a0</code> saving a temporary variable and an opcode call. Less opcode calls result in reduced CPU usage (an average orchestra may compile about 10% faster with <code>--expression-opt</code>, but it depends largely on how many expressions are used, what the control rate is (see also below), etc.; thus, the difference may be less, but also much more). • number of a- and k-rate temporary variables is significantly reduced. This expression <pre>(a1 + a2 + a3 + a4)</pre> <p>will compile as</p> <pre>#a0 Add a1, a2 #a0 Add #a0, a3 #a0 Add #a0, a4 ; (the result is in #a0)</pre> <p>instead of</p> <pre>#a0 Add a1, a2 #a1 Add #a0, a3 #a2 Add #a1, a4 ; (the result is in #a2)</pre> <p>The advantages of less temporary variables are:</p> <ul style="list-style-type: none"> • less cache memory is used, which may improve performance of orchestras with many a-rate expressions and a

low control rate (e.g. ksmpr = 100)

- large orchestras may load faster due to less different identifier names
- index overflow errors (i.e. when messages like this Case2: indx=-56004 (ffff253c); (short)indx = 9532 (253c) are printed and odd behavior or a Csound crash occurs) may be fixed, because such errors are triggered by too many different (especially a-rate) variable names in a single instrument.

Note that this optimization (due to technical reasons) is not performed on i-rate temporary variables.



Warning

When `--expression-opt` is turned on, it is not allowed to use the `i()` function with an expression argument, and relying on the value of k-rate expressions at i-time is unsafe.

<code>--help</code>	Display on-line help message.
<code>-I, --i-only</code>	<i>i-time only</i> . Allocate and initialize all instruments as per the score, but skip all p-time processing (no k-signals or a-signals, and thus no amplitudes and no sound). Provides a fast validity check of the score pfields and orchestra i-variables. This option is exclusive of the <code>--syntax-check-only</code> flag.
<code>--ignore_csopts=integer</code>	If set to 1, Csound will ignore all options specified in the csd file's CsOptions section. See <i>Unified File Format for Orchestras and Scores</i> .
<code>--max_str_len=integer</code>	(min: 10, max: 10000) Maximum length of string variables + 1; defaults to 256 allowing a length of 255 characters. The length of string constants is not limited by this parameter.
<code>-N, --notify</code>	Notify (ring the bell) when score or MIDI track is done.
<code>--no-default-paths</code>	Disables adding of directory of CSD/ORC/SCO to search paths.
<code>--no-expression-opt</code>	Disables expression optimization.
<code>-O FILE, --logfile=FILE</code>	Log output to file <i>FILE</i> . If <i>FILE</i> is null (i.e. <code>-O null</code> or <code>--logfile=null</code>) all printing of messages to the console is disabled.
<code>--syntax-check-only</code>	Causes Csound to exit immediately after the orchestra and score parsers finish checking the syntax of the input files and before the orchestra performs the score. This option is exclusive of the <code>--i-only</code> flag. (Csound 5.08 and later).
<code>-t0, --keep-sorted-score</code>	Prevents Csound from deleting the sorted score file, <code>score.srt</code> , upon exit.
<code>-U UTILITY, --utility=UTILITY</code>	Invoke the utility program <i>UTILITY</i> . Use any invalid name to list the available utilities.

-x FILE, --extract-score=FILE Extract a portion of the sorted score, score.srt, using the extract file *FILE* (see *Extract*).

Csound Environment Variables

The following environment variables can be used by Csound:

- SFDIR: Default directory for sound files. Used if no full path is given for sound files.
- SSDIR: Default directory for input (source) audio and MIDI files. Used if no full path is given for sound files. May be used in conjunction with SFDIR to set separate input and output directories. Please note that MIDI files as well as audio files are also sought inside SSDIR.
- SADIR: Default directory for analysis files. Used if no full path is given for analysis files.
- SFOUTYP: Sets the default output file type. Currently only 'WAV', 'AIFF' and 'IRCAM' are valid. This flag is checked by the csound executable and the utilities and is used if no file output type is specified.
- INCDIR: Include directory. Specifies the location of files used by *#include* statements.
- OPCODEDIR: Defines the location of csound opcode plugins for the single precision float (32-bit) version.
- OPCODEDIR64: Defines the location of csound opcode plugins for the double precision float (64-bit) version.
- SNAPDIR: Is used by the FLTK widget opcodes when loading and saving snapshots.
- CSOUNDR: Defines the csound resource (or configuration) file. A full path and filename containing csound flags must be specified. This variable defaults to .csoundrc if not present.
- CSSTRNGS: In Csound 5.00 and later versions, the localisation of messages is controlled by two environment variables CSSTRNGS and CS_LANG, both of which are optional. CSSTRNGS points to a directory containing .xmg files.
- CS_LANG: Selects a language for csound messages.
- RAWWAVE_PATH: Is used by the STK opcodes to find the raw wave files. Only relevant if you are using STK wrapper opcodes like STKBowed or STKBrass.
- CSNOSTOP: If this environment variable is set to "yes", then any graph displays are closed automatically at the end of performance (meaning that you possibly will not see much of them in the case of a short non-realtime render). Otherwise, you need to click "Quit" in the FLTK display window to exit, allowing for viewing the graphs even after the end of score is reached.
- MFDIR: Default directory for MIDI files. Used if no full path is given for MIDI files. Please note that MIDI files are sought in SSDIR and SFDIR as well.

For more information about SFDIR, SSDIR, SADIR, MFDIR and INCDIR see *Directories and files*.

The only mandatory environment variables are OPCODEDIR and OPCODEDIR64. It is very important to set them correctly, otherwise most of the opcodes will not be available. Make sure you set the path correctly depending on the precision of your binary. if you run csound on a command line without any arguments you should see some text like : Csound version 5.01.0 beta (float samples) Mar 23 2006. This text refers to the single precision version.

CSSTRNGS and CS_LANG currently have very limited use since Csound has not yet been completely translated into other languages.

Other environment variables which are not exclusive to Csound but which might be of importance are:

- **PATH:** The directory containing csound executables should be listed in this variable.
- **PYTHONPATH:** If you intend to use CsoundVST and python, the directory containing the `_CsoundVST` shared library and the `CsoundVST.py` file must be in your **PYTHONPATH** environment variable (or the default path python searches in), so that the Python runtime knows how to load these files.
- **LADSPA_PATH and DSSI_PATH:** These environment variables are required if you are using the *dssi4cs* (LADSPA and DSSI host) plug-in opcodes.
- **CSDOCDIR:** Specifies the directory where the html help files are located. Though not used by Csound directly, this environment variable can help front-ends and editors (which implement it) to find the csound manual.

Setting environment variables

On the command line

You can set environment variables on the command line or the configuration file `.csoundrc` by using the command line flag `--env:NAME=VALUE` or `--env:NAME+=VALUE`, where **NAME** is the environment variable name, and **VALUE** is its value. See *Command-line Flags*



Note

Please note that this method of setting environment variables will not work for variables which are parsed before the command line arguments. **SADIR**, **SSDIR**, **SFDIR**, **INCDIR**, **SNAPDIR**, **RAWWAVE_PATH**, **CSNOSTOP**, **SFOUTYP** should work, but the following environment variables must be set on the system prior to running csound: **OPCODEDIR**, **OPCODEDIR64**, **CSSTRINGS**, and **CS_LANG**. **CSOUNDRC** can currently (v. 5.02) be set using `--env`, but this behavior is not guaranteed for future versions.

Windows

To set a csound environment on Windows XP and 2000 go to Control Panel->System->Advanced and click on the button 'Environment Variables'. On other versions of Windows earlier than Windows XP and Windows 2000 you set environment variables in the `autoexec.bat` file. Go to 'My Computer', select C: drive, right click on `autoexec.bat`, and select 'Edit'. The statement format is: `SET NAME=VALUE`.

Linux

You can set environment variables on Linux in many ways. You can set them using the *export* shell command, by setting them on `.bashrc` or similar files or by adding them to the `/etc/profile` file.

Mac

If the user has a Mac that shipped with an OS X version prior to 10.3 (includes 10.2 and 10.1) then it is possible that the default shell is the Tenex C-shell (tcsh). If this is the case, then you either have to type:

```
~% setenv OPCODEDIR "/Users/you/your/Csound5/build"
```

or change your `/etc/profile` and or edit your `.tcshrc` file.

If the user has a Mac that shipped with OS X 10.3 or 10.4 then it likely has the "Bourne-again" C-shell (bash) as the default shell. If this is the case, then the user must type something like:

```
~$ export OPCODEDIR=/Users/you/your/Csound5/build
```

in addition if the bash shell is the default, then it is usually easier to edit your .bashrc or /etc/profile.

Note that if users choose one of the above methods, ie editing the .bashrc file then the environment variables are executed when a new shell is created. This can be problematic if your application implements a Quartz or Aqua interface and does not use the commandline.

If this is the case, then the standard solution (up to OS 10.3.9 and unless the application uses the csoundAPI and sets the environ variables directly) is to create an XML property list file (called a .plist file by the OS). This file should nominally be located at ~/.MacOSX/Environment.plist. This has been a solution specifically for the [csoundapi~] object for Pd on OS X. Since Pd uses an OS X native .app style packaging, and runs off of the Aqua interface, the standard means of supplying environment variables to Csound do not work. The solution is to set Csound's environment variables for the Aqua environment.

Likely, most users will not have the hidden folder .MacOSX located in their \$HOME directory (aka ~/) This folder must first be created and the Environment.plist added to this folder. The contents of the Environment.plist file should be something like:

```
<?xml version="1.0" encoding="UTF-8"?>
<!DOCTYPE plist PUBLIC "-//Apple Computer//DTD PLIST 1.0//EN"
    "http://www.apple.com/DTDs/PropertyList-1.0.dtd">
<plist version="1.0">
<dict>
<key>OPCODEDIR</key>
<string>/Library/Frameworks/CsoundLib.framework/Versions/5.1/Resources/Opcodes</string>
<key>OPCODEDIR64</key>
<string>/Volumes/ExternalHD/devel/csound5/lib64</string>
<key>INCDIR</key>
<string>/Volumes/ExternalHD/CSOUND/include</string>
<key>SFDIR</key>
<string>/Volumes/ExternalHD/iTunes/csoundaudio</string>
</dict>
</plist>
```

and so on, using the XML <key> tag for each environment variable required by the API and the <string> tag for it's corresponding path on the system.

Please note that you must login out and login in for these changes to take effect.

Unified File Format for Orchestras and Scores

Description

The Unified File Format, introduced in Csound version 3.50, enables the orchestra and score files, as well as command line flags, to be combined in one file. The file has the extension .csd. This format was originally introduced by Michael Gogins in AXCsound.

The file is a structured data file which uses markup language, similar to any SGML such as HTML. Start tags (<tag>) and end tags (</tag>) are used to delimit the various elements. The file is saved as a text file.

Structured Data File Format

Mandatory Elements

The first tag in the file must be the start tag <CsoundSynthesizer>. The last tag in the file must be

the end tag `</CsoundSynthesizer>`. This element is used to alert the csound compiler to the `.csd` format. All text before the start tag and after the end tag is ignored by Csound. The tag may also be spelled `<CsoundSynthesiser>`.

Options (<CsOptions>)

Csound *command line flags* are put in the Options Element. This section is delimited by the start tag `<CsOptions>` and the end tag `</CsOptions>`. Lines beginning with `#` or `;` are treated as comments.

Orchestra (<CsInstruments>)

The instrument definitions (orchestra) are put into the Instruments Element. The statements and syntax in this section are identical to the Csound *orchestra file*, and have the same requirements, including the header statements (*sr*, *kr*, etc.) This Instruments Element is delimited with the start tag `<CsInstruments>` and the end tag `</CsInstruments>`.

Score (<CsScore>)

Csound score statements are put in the Score Element. The statements and syntax in this section are identical to the Csound *score file*, and have the same requirements. The Score Element is delimited by the start tag `<CsScore>` and the end tag `</CsScore>`.

Optional Elements

Included Base64 Files (<CsFileB>)

Base64-encoded files may be included with the tag `<CsFileB filename=filename>`, where *filename* is the name of the file to be included. The Base64-encoded data should be terminated with a `</CsFileB>` tag. For encoding files, the `csb64enc` and `makecsd` utilities (included with Csound 5.00 and newer) can be used. The file will be extracted to the current directory, and deleted at end of performance. If there is an already existing file with the same name, it is not overwritten, but an error will occur instead.

Base64-encoded MIDI files may be included with the tag `<CsMidifileB filename=filename>`, where *filename* is the name of the file containing the MIDI information. There is no matching end tag. This was added in Csound version 4.07. Note: using this tag is not recommended; use `<CsFileB>` instead.

Base64-encoded sample files may be included with the tag `<CsSampleB filename=filename>`, where *filename* is the name of the file containing the sample. There is no matching end tag. This was added in Csound version 4.07. Note: using this tag is not recommended; use `<CsFileB>` instead.

Version Blocking (<CsVersion>)

Versions of Csound may be blocked by placing one of the following statements between the start tag `<CsVersion>` and the end tag `</CsVersion>`:

Before `##`

or

After `##`

where `##` is the requested Csound version number. The second statement may be written simply as:

`##`

This was added in Csound version 4.09.

Licence Information (<CsLicence> or <CsLicense>)

Licensing details can be included in between the start tag <CsLicence> and the end tag </CsLicence>. There is no format for this information, any text is acceptable. This text will be printed by Csound to the console when the CSD is run.

Example

Below is a sample file, test.csd, which renders a .wav file at 44.1 kHz sample rate containing one second of a 1 kHz sine wave. Displays are suppressed. test.csd was created from two files, tone.orc and tone.sco, with the addition of command line flags.

```
<CsoundSynthesizer>;
; test.csd - a Csound structured data file

<CsOptions>
-W -d -o tone.wav
</CsOptions>

<CsVersion>      ; optional section
Before 4.10      ; these two statements check for
After 4.08       ; Csound version 4.09
</CsVersion>

<CsInstruments>
; originally tone.orc
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
instr 1
a1 oscil p4, p5, 1 ; simple oscillator
out a1
endin
</CsInstruments>

<CsScore>
; originally tone.sco
f1 0 8192 10 1
i1 0 1 20000 1000 ; play one second of one kHz tone
e
</CsScore>

</CsoundSynthesizer>
```

Command Line Parameter File (.csoundrc)

If the file .csoundrc exists, it will be used to set the command line parameters. These can be overridden. Csound 5.00 and newer versions read this file from the HOME directory first (or the full path file name defined by the CSOUNDRC *environment variable*), and then the current directory. If both exist, options in the .csoundrc in the current directory will have higher precedence. It uses the same form as a .csd file, but no tags are needed. Lines beginning with # or ; are treated as comments.

A .csoundrc file can contain something like this:

```
-+rtaudio=portaudio -odac2 -iadc2 -+rtmidi=winmm -M1 -Q1 -m0
```

In this case, csound will generate real-time output and take realtime input from device 2, using the portaudio driver interface. It will input and output realtime MIDI on interface 1. It will print very few messages (-m0). These options will be used by default when other options are not given inside the <CsOptions> of the .csd file or the command line (See *Order of precedence*).

Score File Preprocessing

The Extract Feature

This feature will extract a segment of a sorted numeric score file according to instructions taken from a control file. The control file contains an instrument list and two time points, from and to, in the form:

```
instruments 1 2 from 1:27.5 to 2:2
```

The component labels may be abbreviated as i, f and t. The time points denote the beginning and end of the extract in terms of:

```
[section no.] : [beat no.].
```

Each of the three parts of the argument is optional. The default values for missing i, f or t are:

```
all instruments, beginning of score, end of score.
```

Independent Pre-Processing with Scsort

Although the result of all score preprocessing is retained in the file `score.srt` after orchestra performance (it exists as soon as score preprocessing has completed), the user may sometimes want to run these phases independently. The command

```
scot filename
```

will process the Scot formatted filename, and leave a *standard numeric score* result in a file named `score` for perusal or later processing.

The command

```
scscort < infile > outfile
```

will put a numeric score infile through Carry, Tempo, and Sort preprocessing, leaving the result in outfile.

Likewise *extract*, also normally invoked as part of the *Csound command*, can be invoked as a standalone program:

```
extract xfile < score.sort > score.extract
```

This command expects an already sorted score. An unsorted score should first be sent through Scsort then piped to the extract program:

```
scsort < scorefile | extract xfile > score.extract
```

Using Csound

Csound can be operated in a variety of modes and configurations. The original method for running Csound was as a console program (DOS prompt for Windows, Terminal for Mac OS X). This, of course, still works. Running `csound` without any arguments prints out a list of command-line options, which are more fully explained in the *Command Line Flags (by Category)* section. Normally, the user executes something like:

```
csound myfile.csd
```

or separate orchestra (orc) and score (sco) files can be used:

```
csound myorchestra.orc myscore.sco
```

You can find many .csd files in the examples folder. Most opcode entries in this manual also include simple .csd files showing the usage of the opcode.

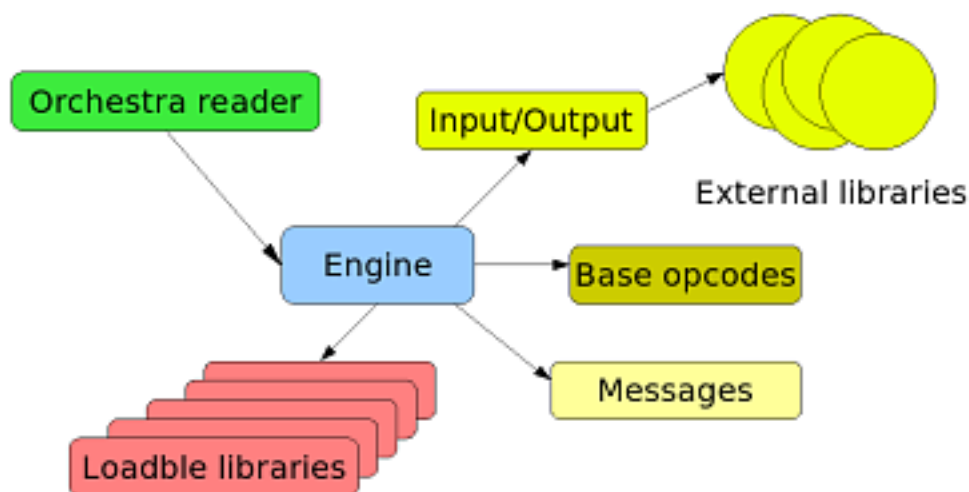
There are also many *Front-Ends* which can be used to run `csound`. A *Front-End* is a graphical program that eases the process of running `csound`, and sometimes provides editing and composing functions.

Csound also has several ways of producing output. It can:

- Read and write soundfiles (off-line rendering) - Using the `-o` and `-i` flags specifying an output filename.
- Read and write digital audio using a sound card (real-time rendering) - Using the `-odac` and `-iadc` flags
- Read and write MIDI files (non-realtime) - Using the `-F` and `--midioutfile` flags.
- Read and write MIDI using a MIDI interface and controller (real-time control) - Using the `-M` and `-Q` flags.

How Csound5 works

Csound processes and generates output using "unit generators" (ugens) called *opcodes*. These opcodes are used to define *instruments* in the *orchestra*. When you run Csound, the engine loads the base Opcodes, and the opcodes contained in separate loadable "opcode libraries". It then interprets the orchestra (through the orchestra reader). The engine sets up an instrument processing chain, which then receives events from the score or in real-time. The processing chain uses the input/output modules to generate output. There are modules that can write to file, or generate *real-time audio output*.



The Csound5 Modular structure.

Csound's processing buffers

Csound processes audio in sample blocks called buffers. There are three separate buffer layers:

1. *spout* = Csound's innermost software buffer, contains *ksmps* sample frames. Csound processes real-time control events once every *ksmps* sample frames.
2. *-b* = Csound's intermediate software buffer (the "software" buffer), in sample frames. Should be (but does not need to be) an integral multiple of *ksmps* (can equal *ksmps* too). Once per *ksmps* sample frames, Csound copies *spout* to the *-b* buffer. Once per *-b* sample frames, Csound copies the *-b* buffer to the *-B* "hardware" buffer.
3. *-B* = The sound card's internal buffer (the "hardware" buffer), in sample frames. Should be (and may need to be) an integral multiple of *-b*. If Csound misses delivering a *-b* one time, the extra *-b* sample frames in *-b* are still there for the sound card to keep playing while Csound catches up. But they can be the same size if you're willing to bet Csound can always keep up with the sound card.

Amplitude values in Csound

Amplitude values in Csound are always relative to a "*Odbfs*" value representing the peak available amplitude before clipping, in either an AD/DA codec, or in a soundfile with a defined range (which both WAVE and AIFF are). In the original Csound, this value was always 32767, corresponding to the bipolar range of a 16bit soundfile or 16bit AD/DA codec, Csound's only possible output back then. This remains the *default* peak amplitude for Csound, for backward compatibility and you will find most of this manual's examples still use this value (hence you find large amplitude values like 10000).

The *Odbfs* value enables Csound to produce appropriately scaled values to whatever output format is being used, whether 24bit integer, 32bit floats, or even 32bit integers. Put another way, the literal amplitude values you write in a Csound instrument only match those written *literally* to the file if the *Odbfs* value in Csound corresponds exactly to that of the output sample format. The consequence of this approach is that you can write a piece with a certain amplitude and have it render correctly

and identically (setting aside of course the better dynamic range of the high-res formats) whether written to an integer or floats file, or rendered in real-time.



Note

The one exception to this is if you choose to write to a "raw" (headerless) file format. In such cases the internal *Odbfs* value is meaningless, and whatever values you use are written unmodified. This does enable arbitrary data to be generated or processed by Csound. It is a relatively exotic thing to do, but some users need it.

You can choose to redefine the *Odbfs* value in the orchestra header, purely for your own convenience or preference. Many people will choose 1.0 (the standard for SAOL, other software like Pure Data, and for many plugin standards such as VST, LADSPA, CoreAudio AudioUnits, etc), but any value is possible.

The common factor in defining amplitudes is the decibel (dB) scale, with 0dB_{FS} always understood as digital peak; hence "0dbfs" means "0dB Full-Scale value". This measure is different to actual amplitude values, since amplitude values are a linear scale which show the actual oscillation around 0, so they can be positive or negative. Decibel values are an absolute logarithmic scale, but can be useful for most opcodes as well. You can convert amplitude to and from decibels using the *ampdb*, *ampdbfs*, *dbamp* and *dbfsamp* functions. This way, Csound enables the programmer to express all amplitudes in dB - lower amplitudes will then be represented by negative dB values. This reflects industry practice (e.g. in level meters in mixers, etc).

For example the same dB level of -6dB (half the amplitude) or -20dB are actually a different linear amplitude according to *Odbfs* like this:

Table 2. dB_{FS} in relation to amplitude

dB_{FS}	0dbfs = 32767 (default)	0dbfs = 1	0dbfs = 1000 (unusual)
0 dB	32767	1	1000
-6 dB	16384	0.5	500
-20 dB	3276.7	0.1	100

Some Csound users might therefore be minded to express all levels in dB_{FS} , and obviate any confusion or ambiguity of level that may otherwise arise when using explicit amplitude values. The decibel scale reflects the response of the ear pretty closely, and that when you want to express a really quiet level, it might be easier and more expressive to write "-46dB" than "0.005" or "163.8".

The reason for using 0dbfs is very simple: digital peak equates to maximum level regardless of sample resolution. If you then define a signal at -110dB you will lose it if rendering to a 16bit file, but retain it (audibly or not) if rendering to 24bit or better. In other words, there is a fixed ceiling, but a moveable floor - you can define sounds as quietly as you like (e.g. envelope tails), in a predictable way, and preserve them or not (without changing the orch code at all), depending on the resolution (file or audio i/o) you render to.



A note on digital amplitude, decibels and dynamic range

A convenient approximation of dynamic range for a certain digital precision is to calculate the decibel interval between the minimum value and the maximum value for a sample. As a rule of thumb, 1 bit (doubling of level) is 6dB, so 16bits = 96dB.

This is not entirely accurate since audio sample values are represented on a bipolar scale with positive and negative values, and 1 bit is used for the sign. Therefore, for 16bit integer samples actually use 1 bit for the sign and 15 bits for the values, so the actual dynamic range is 90dB.

Real-Time Audio

The following information applies mostly to `csound` being run directly from the command line. Front-ends implement these features in different ways, but knowledge of them is necessary in some of them.

The `-i` and `-o` flags can be used to specify realtime output instead of the ordinary non-realtime file output. You should use `-o dac` for realtime output and `-i adc` for realtime input. Naturally, you can use either one or both if your hardware supports it. You can also specify the hardware you want to use by appending a device number or name to the flag (See `-i` and `-o`).

You might also need to use the `+rtaudio` flag to specify the driver interface to be used. Csound defaults to using Portaudio, which is cross-platform and reliable, but for better performance, you might need to use ALSA or JACK on linux, and CoreAudio on Mac. You can use ASIO on Windows if your version of Portaudio has been compiled with ASIO support.

You can see a list of available devices by giving a device number which is out of range, for instance `-o dac99`. This will also reveal if you have ASIO available if you are using PortAudio.

Period & Buffer Sizes

Period and buffer sizes will vary greatly from one machine to another. Lower buffer sizes will result in lower latency, but might cause breakups or clicks in the audio. The Csound flags which control period and buffer sizes are `-b` and `-B`, respectively. Buffer size is hardware dependant, and some experimentation may be necessary to find the optimal balance between low latency performance and uninterrupted audio output. The values given to `-b` and `-B` should be powers of two, and the value of `-B` should be at least one power of two higher than that of `-b`.

Currently, with `-B` set to 512, audio output latency is about 12 milliseconds, fast enough for reasonably responsive keyboard playing. Even shorter latencies, are feasible on some systems.

Control Rate

Low values for `ksmps` will in general give a higher quality of synthesis, but will consume more system resources. There is no hard and fast rule for setting `ksmps` - different orchestras will require different control rates. A waveguide instrument will need a `ksmps` of 1 (and may not be suitable for realtime use), whereas a simple FM synth may be run with a higher `ksmps` without noticeable degradation of sound. If the FM synth were to be used to play a monophonic bassline, a very low `ksmps` may be used, however more complex note clusters will require a higher `ksmps`. A well-tuned Linux system should be capable of running even complex polyphonic synths with `ksmps` values as low as 4 or 8. If full duplex audio is required, `-b` must be an integer multiple of `ksmps`. Bearing this in mind, a rule of thumb might be to only use powers of two for `ksmps`.

Some settings differ according to platform. See further below for information for each platform.

Realtime I/O on Linux

Under Linux, the default PortAudio/PortMidi settings will result in higher latency than that which can be achieved using ALSA and/or JACK. The PortMusic plugins are audio and MIDI servers, which provide an interface to the ALSA drivers, in a manner which is in some respects similar but fundamentally different from that provided by JACK. For a more detailed comparison please refer to:

<http://jackaudio.org/faq>

Using ALSA

The highest level of control and the lowest possible level of latency are to be achieved using the ALSA plugins in combination with the `--sched` flag. Using `--sched` requires that Csound be run as the root user, which may be impossible or undesirable in some circumstances.

The ALSA plugins require the "name" of a "card" and a "device". Unless you have named your "cards" in `~/asoundrc` (or `/etc/asound.conf`), the "names" will actually be numbers. In order to obtain a list of the possible configurations, use the command line utilities "aplay", "arecord" and "amidi". These utilities are included with most Linux distros, or can be downloaded and built from source here:

<ftp://ftp.alsa-project.org/pub/utils/>

Audio Output

Running the following command:

```
aplay -l
```

will give you a list of the audio playback devices available on your system. Typically this list will look something like:

```
[...]
**** List of PLAYBACK Hardware Devices ****
card 0: A5451 [ALI 5451], device 0: ALI 5451 [ALI 5451]
[...]
```

If you have more than one card on your system, or if there is more than one device on your card, the list will of course be more complicated, however in all cases the information that is pertinent is the number/name of the card/device. In order to use the above soundcard for audio output, the following flag would be added to the Csound command line, `~/csoundrc`, or the `<CsOptions>` section of a CSD:

```
--rtaudio=alsa -o dac
```

Output with dmix

If you would like to use Csound with dmix and your soundcard does not support hardware mixing of audio streams, special care is needed in setting up of software (-b) and hardware (-B) buffers. If you get a message from Csound's ALSA driver that looks like the following:

```
ALSA: -B 8192 not allowed on this device; use 7526 instead
```

there is a good chance that you may be using dmix. If you are using dmix, the -b and -B settings of Csound must be synced the `period_size` and `buffer_size` of dmix respectively, using a ratio of the sr for the Csound project to the sample rate that dmix is set up to. The following formula will determine what settings to use for Csound given the settings of dmix:

```
-b = (csound_sr/dmix_sample_rate) * dmix_period_size
-B = (csound_sr/dmix_sample_rate) * dmix_buffer_size
```

For example, if dmix is set to 48000 sample rate, a `period_size` of 1024, and a `buffer_size` of 8192, when running a Csound project with `sr=48000`, the settings for buffers should be "-b 1024 -B8192". If the `sr=24000`, the settings for buffers should be "-b 512 -B4096".

Because of this relationship, if a Csound project's `sr` does not evenly divide into the `sample_rate` used by dmix, then it may be difficult if not impossible to set the correct setting for -b and -B due to rounding errors. It is suggested then that if you are using sample rates different than what your setting is for dmix, then you may want to configure dmix to have a `period_size` and `buffer_size` that can be evenly divided by the ratio between the `csound sr` and `dmix sample_rate`. For example, to run a project with `sr=16000`, the following dmix setting:

```
pcm.amix {
```

```
type dmix
ipc_key 50557
slave {
    pcm "hw:0,0"
    period_time 0
    #period_size 1024
    #buffer_size 8192
    period_size 1536
    buffer_size 12288
}
bindings {
    0 0
    1 1
}
}

# route ALSA software through pcm.amix
pcm.!default {
    type plug
    slave.pcm "amix"
}
```

with period_size 1536 and buffer_size 12288 will divide nicely by 3 (the ratio of the csound sr to the dmix sample_rate) to get "-b 512 -B4096" ($(16000/48000) * 1536 = 512$, $(16000/48000) * 12288 = 4096$).



Note

For most soundcards that this affects, the default sample rate for the card will be 48000 and the defaults for dmix will be 1024 and 8192.

Audio Input

Typically the same card will be used for both input and output, so to continue using the foregoing example, the flag:

```
-i adc:hw:0,0
```

would be added for audio input from Card 0 Device 0. To use the default card employ one of the following flags, with the forementioned warning that this will not necessarily work:

```
-i adc
```

If you wish to use a different card or device for input, running the following utility from the command line will provide a list of input devices:

```
arecord -l
```

If, by way of an example, you wanted to use a USB audio interface, which is the second "card" in your system, for output, but wanted to use your internal soundcard, the first card in your setup, for input, you would put the following flags somewhere useful:

```
--rtaudio=alsa -i adc:hw:0,0 -o dac:hw:1,0
```

If you wanted to use the second device on your USB interface, to send audio to a specific channel, for instance, you would use the following flags:

```
--rtaudio=alsa -i adc:hw:0,0 -o dac:hw:1,1
```

MIDI Input

Csound does not automatically create its own ALSA sequencer port. It creates an ALSA raw midi

port each time it runs. In order to enable your orchestra to receive MIDI input you can use VirMIDI or MIDITHru, whichever you prefer. Setting up these virtual MIDI ports is a topic that has been covered extensively elsewhere, see The Linux MIDI how-to [<http://www.midi-howto.com/>] or browse your distro's documentation or the ALSA documentation for instructions on how to install and configure either VirMIDI or MIDITHru (seqdummy). Once you have done so run:

```
amidi -l
```

for a list of available devices. Typically this will look something like the following:

```
[...]
Device  Name
hw:1,0  Virtual Raw MIDI (16 subdevices)
hw:1,1  Virtual Raw MIDI (16 subdevices)
hw:1,2  Virtual Raw MIDI (16 subdevices)
hw:1,3  Virtual Raw MIDI (16 subdevices)
hw:2,0,0 PCR MIDI
hw:2,0,1 PCR 1
```

In this example, Csound can connect to any of the four available Virtual Raw MIDI ports, where it will listen for MIDI input. The following flag instructs Csound to listen on the first of these ports:

```
-+rtmidi=alsa -Mhw:1,0
```

You will then need to connect your hardware or software controller to the port which is hosting your Csound synthesizer. The simplest way to do this is using the "aconect" utility. Run:

```
aconect -li
```

for a list of available input devices, and:

```
aconect -lo
```

for a list of available output devices (including the port to which Csound has been connected). These should give something like the following:

```
#aconect -li
client 0: 'System' [type=kernel]
  0 'Timer'
  1 'Announce'
    Connecting To: 15:0
client 20: 'Virtual Raw MIDI 1-0' [type=kernel]
  0 'VirMIDI 1-0'
client 21: 'Virtual Raw MIDI 1-1' [type=kernel]
  0 'VirMIDI 1-1'
client 22: 'Virtual Raw MIDI 1-2' [type=kernel]
  0 'VirMIDI 1-2'
client 23: 'Virtual Raw MIDI 1-3' [type=kernel]
  0 'VirMIDI 1-3'
client 24: 'PCR' [type=kernel]
  0 'PCR MIDI'
  1 'PCR 1'
  2 'PCR 2'
```

```
#aconect -lo
```

```
client 20: 'Virtual Raw MIDI 1-0' [type=kernel]
0 'VirMIDI 1-0'
client 21: 'Virtual Raw MIDI 1-1' [type=kernel]
0 'VirMIDI 1-1'
client 22: 'Virtual Raw MIDI 1-2' [type=kernel]
0 'VirMIDI 1-2'
client 23: 'Virtual Raw MIDI 1-3' [type=kernel]
0 'VirMIDI 1-3'
client 24: 'PCR' [type=kernel]
0 'PCR MIDI'
1 'PCR 1'
```

In the following example, the USB keyboard which is listed above as client 24 will be connected to a Csound synthesizer which is listening on the first VirMIDI port. The keyboard has three output ports. The first (24:0) transmits messages received on the MIDI in port, the second (24:1) transmits keyboard and controller messages, and the third (24:2) transmits system exclusive messages. The following command connects the second port of the keyboard to the Csound synthesizer:

```
aconnect 24:1 20:0
```

Remember that Csound acts as a raw MIDI device and is not an ALSA sequencer client. This means that Csound will not appear in MIDI device listings and will not be available for use directly with *aconnect*, so you must connect to a virtual device (like 'virtual raw MIDI' or 'MIDI through') for persistent connections, or connect directly to the destination using command line flags.

MIDI Output

Csound can be connected to any device which shows up on the ALSA sequencer list of output ports, obtained by "amidi -l" as above. In order to connect a Csound synthesizer to the MIDI out port of the keyboard listed above, the following flag would be used:

```
-Qhw:2,0,0
```

Scheduling

If you are able to run Csound as the root user, using the "--sched" flag will dramatically improve real-time performance, when using ALSA, however you may hang your system if you do something stupid. DO NOT use "--sched" if you are using JACK for audio output. JACK controls scheduling for the audio applications connected to it, and also tries to run at the highest possible priority. If the "--sched" flag is used, Csound and JACK will be competing rather than cooperating, resulting in extremely poor performance.

Using JACK

The simplest way to use the JACK plugin enabling input and output is as follows:

```
++rtaudio=jack -i adc -o dac
```

Additionally, there are some command line options specific to JACK:

JACK Command-line Flags

--jack_client=[client_name]

The client name used by Csound, defaults to 'csound5'. If multiple instances of Csound connect to the JACK server, different client names need to be used to avoid name conflicts.

name prefix], - +jack_outportname=[output port name prefix]	Name prefix of Csound JACK input/output ports; the default is 'input' and 'output'. The actual port name is the channel number appended to the name prefix. Example: with the above default settings, a stereo orchestra will create these ports in full duplex operation: <div style="display: flex; justify-content: space-between;"><div>csound5:input1 csound5:input2 csound5:output1 csound5:output2</div><div>(record left) (record right) (playback left) (playback right)</div></div>
-+jack_sleep_time=[sleep time in microseconds]	As of Csound version 5.01, this option is deprecated and ignored.

Connecting Csound to other JACK clients

By default, no connections are made (you need to use `jack_connect`, `qjackctl`, or a similar utility); however, the plugin can connect to ports specified as `'-iadc:portname_prefix'` or `'-odac:portname_prefix'`, where `portname_prefix` is the full name of a port without a channel number, such as `'alsa_pcm:capture_'` (for `-i adc`), or `'alsa_pcm:playback_'` (for `-o dac`).

Notes on buffer sizes

Audio data is received from and sent to the JACK server by Csound using a ring buffer that is controlled by the `-b` and `-B` flags. `-B` is the total size of the buffer, while `-b` is the size of a single period. These values are rounded so that the total size is an integer multiple of, and greater than the period size. The difference of the Csound buffer and period size must be greater than or equal to the JACK period size.

If both `-iadc` and `-odac` are used at the same time, the `-b` option should be set to an integer multiple of `ksmps`.

An example of buffer settings for low latency on a fast system:

```
jackd -d alsa -P -r 48000 -p 64 -n 4 -zt &  
csound --rtaudio=jack -b 64 -B 256 [...]
```

with real time scheduling (as root):

```
jackd -R -P 90 -d alsa -P -r 48000 -p 64 -n 2 -zt &  
csound --sched=80,90,10 -d --rtaudio=jack -b 64 -B 192 [...]
```

To improve performance, use `ksmps` values like 32 and 64.

The sample rate of the orchestra must be the same as that of the JACK server.

Using Pulseaudio

Support for Pulseaudio [<http://www.pulseaudio.org/>] was added in Csound 5.09. You can specify the following settings:

1. Sink names: it's possible to use a number instead of the full name, so `-odac:1` would select your second device (count starts at 0).
2. Server name: it's possible to connect to a specific server by using `++server=<server_string>` where `server_string` is a name of a server or a more complex server selection string (see [pulseaudio.org](http://www.pulseaudio.org/) [<http://www.pulseaudio.org/>] on server strings). This should be network transparent and should allow connections to remote machines.
3. Stream names: it is possible to label the streams generated by `csound`, by using `-+output_stream=<stream-name>` and `++input_stream=<stream-name>`

Here's an example command line:

```
csound -odac:1 examples/trapped.csd -+rtaudio=pulse -+server=unix:/tmp/pulse-victor/native -+output_stream=trapped
```

Windows

Real-time Audio

Windows users can use either the default *PortAudio* Realtime module, or the *winmm* Realtime Audio Module. The *winmm* module is a native windows module which provides great stability, but latency will usually be too high for realtime interaction. To activate a realtime module, you can use the *-+rtaudio* flag with value of *portaudio* or *winmme*. The default value is *portaudio*, which is activated by default without specifying it.

You also need to specify the sound device you want to use, and specify that you want to generate real-time audio output instead of soundfile to disk output. To do this, you must use the *-odac* or *-odac* flag, which tells *csound* to output to the Digital-to-Analog converters instead of a file. By adding a number after the flag (e.g. *-odac2*), you can choose the device number you want. To find out available devices in your system, you can use a large out of range number (e.g. *-odac99*), and *csound* will report an error, and list available devices.

When choosing the device number under Portaudio, you are also choosing the driver interface, since Portaudio supports WinMME, DirectX and ASIO. If you have an ASIO capable interface or an ASIO driver emulator like ASIO4ALL [<http://www.asio4all.com>], the device will show multiple times, once for each driver interface. ASIO will give you the best latency on your system, so if available it should be your choice for realtime audio output.

Enabling realtime audio input is done using *-iadc*, which makes *csound* listen to the realtime audio outputs. You can again select the device by its number, and check for available devices using an out of range number. Note that for input you use 'adc' instead of 'dac'. Make sure you have the appropriate input selected in your soundcard's control panel.

Real-time MIDI

To enable Real-time MIDI on Windows, you can use the *-M* flag for MIDI input and the *-Q* flag for MIDI output. You might need to specify the device number after the flag (e.g. *-M2*), and again, you can find the available devices by giving an out of range number.

Csound will use PortMidi as the default MIDI module, but there's also a native *winmme* module, which can be activated with the flag:

```
-+rtmidi=winmme
```

A typical set of flags to enable Real-time Audio and MIDI I/O can look like:

```
-+rtmidi=winmme -M1 -Q1 -+rtaudio=portaudio -odac3 -iadc3
```

Mac

Coming Soon...

Optimizing Audio I/O Latency

To achieve the lowest latency possible without audio break ups, a combination of variables needs to be tweaked. The final values will be platform and system dependent, and will also depend on the complexity of the audio calculations performed. You need to adjust *ksmps* in the orchestra, as well as the software (*-b*) and hardware buffer (*-B*) sizes.

Usually the simplest solution is the following:

1. Set *ksmps* to a value with a good tradeoff between quality and performance, without adjusting *-B* at all.
2. Set *-b* to a negative power of two of this value.

To get the optimal values, start with something you think is going to be too low, ie -1, and then continue "upwards", -2, -4 and so on, until you stop getting x-runs (glitches). The real value of *-b* will be the absolute value of $-b * ksmps$.

3. Reduce the hardware buffer (*-B*). Bring it down from the default (1024 on Linux, 4096 on Mac OS X, 16384 on Windows), halving it each time, until you start to get x-runs (glitches) again. Then take it back up again until performance is continuous.

This process assumes you have a 16-bit soundcard. If you have a 24-bit soundcard, then *-B* should be 3/2, or 3 times *-b*, rather than 2 or 4 times. Csound works with 32-bit floats, or 64-bit doubles whereas most soundcards are 16 or 24-bit integer. *-b* is the internal buffer, so it's dealing with the 32 or 64-bit side of things, whereas *-B* is the hardware buffer, so it's dealing with the 16 or 24-bit side. The csound default for floats is $-B = 4 * -b$. This is a sane value for a 16 bit card. You can usually get away with $-B = 2 * -b$, but this is the absolute minimum. For example, if you set *-b1024 -B2048*, csound will tell you that:

```
audio buffered in 1024 sample-frame blocks
writing 4096-byte blocks to dac
```

4096 bytes is 32768 bits. $32768/32 = 1024$, our sample-frame size, $1024 * 32/16 = 2048$, our buffer size. Were we to reduce the value of *-B*, we would need to reduce the value of *-b* by a corresponding amount in order to continue to write 16-bit integers to dac. The minimum size of *-b* is $(-B * bitrate)/32$. That is to say that the minimum ratio of *-b* to *-B* should be:

- 16-bit: 1:2
- 24-bit: 2:3
- 32-bit: 1:1

While there is no theoretical maximum ratio, it makes no sense to have a very high ratio here, as the software buffer has to fill the hardware buffer before returning. If the ratio is high, it will take a long time, defeating the purpose of setting a small value for *-b*.

The value of *-b* is something that will need to be varied depending on the complexity of the instrument you're working with, but because it's intimately related to the value of *ksmps*, it's better to synchronise it with *ksmps* and go from there. One way to do it is to decide how long the release on your envelopes might need to be at maximum (for desired effect), set the release on all envelopes to maximum, give yourself a generous value for *-b*, and then play. If it breaks up, double *ksmps*, repeat until smooth, then bring the value of *-b* down as far as possible.

The value of *-B* is primarily determined by operating system and soundcard. Figure out (using above method) how low you can go, and use that value (or one higher for safety). If you have problems you'll know that it's probably because of an inappropriate value for *ksmps*, too low a value for *-b*, or denormals (see *denorm*).

Configuring

Once you have either unpacked a binary distribution, or built Csound from sources, you will need to configure Csound so that it will run properly on your system. Installers usually perform these steps for you automatically.

On all platforms, make sure the directory or directories containing Csound's plugin libraries are in an `OPCODEDIR` or `OPCODEDIR64` environment variable depending on the precision of the compiled binary.

The Python opcodes currently require at least Python 2.4, which can be downloaded from www.python.org [<http://www.python.org>] if it is not already on your system. You can check if it is available by typing 'python' on a command prompt or DOS window.

Windows

On Windows, make sure the directory or directories (normally the `C:\Program Files\Csound` directory) containing the Csound executables directory are in your `PATH` variable, or else copy all the executable files to your Windows `system32` directory. Depending on your installation method, you might also need to set the `OPCODEDIR` and `OPCODEDIR64` environment variables. Assuming that Csound is installed to the default location of `C:\Program Files\Csound` you can use (otherwise set the paths accordingly):

```
set OPCODEDIR=C:\Program Files\Csound\plugins
set OPCODEDIR64=C:\Program Files\Csound\plugins64
set PATH=%PATH%;C:\Program Files\Csound\bin
```



Missing python24.dll or python25.dll

If you get a pop-up about the missing Python library (`python24.dll` or `python25.dll`) and don't need the python opcodes, just delete `C:\Program Files\Csound\plugins\py.dll` and `C:\Program Files\Csound\plugins64\py.dll`, and the pop-up about the missing Python library should be gone.

Unix and Linux

On Unix and Linux, either install the Csound program in one of the system `bin` directories, typically `/usr/local/bin`, and the Csound and plugin shared libraries in places like `/usr/local/lib/csound/plugins` or `/usr/local/lib/csound/plugins64` and make sure that `OPCODEDIR` and `OPCODEDIR64` environment variable are set correctly.

CsoundAC

CsoundAC requires some additional configuration. On all platforms, CsoundAC requires that you have Python installed on your computer. The directory containing the `_csoundAC` shared library and the `CsoundAC.py` file must be in your `PYTHONPATH` environment variable, so that the Python runtime knows how to load these files.

Syntax of the Orchestra

The Csound orchestra (.orc) or the *<CsInstruments>* section of a csd file, contains:

- A *header section*, which specifies global options for instrument performance
- A list of *User defined opcodes* and *instrument blocks* containing UDO and instrument definitions.

The orchestra header, instrument blocks, and UDOs contain *Orchestra statements*. An *orchestra statement* in Csound has the format:

```
label:    result opcode argument1, argument2, ... ;comments
```

The label is optional and identifies the basic statement that follows as the potential target of a go-to operation (see *Program Flow Control*). A label has no effect on the statement per se.

Depending on their function, some opcodes produce no output, so they have no result value. Others take no arguments and only produce a result.

Every orchestra statement must be on a single line, however long lines can be wrapped to a new line using the `\` character. This character indicates that the next line is part of the current one, this way you can split a line for easier reading, like this:

```
a2  oscbnk kcps, 1.0, kfmd1, 0.0, 40, 203, 0.1, 0.2, kamfr, kamfr2, 148, \
      0, 0, 0, 0, 0, 0, -1, \
      kfnum, 3, 4
```

Comments are optional and are for the purpose of letting the user document his orchestra code. Comments begin with a semicolon (;) and extend to the end of the line. Comments can optionally be in C-style, spanning multiple lines like this:

```
/* Anything in here -----
   is a comment which can span
   several lines ----- */
```

The remainder (result, opcode, and arguments) form the basic statement. This also is optional, i.e. a line may have only a label or comment or be entirely blank. If present, the basic statement must be complete on one line, and is terminated by a carriage return and line feed.

The opcode determines the operation to be performed; it usually takes some number of input values (or arguments, with a maximum value of about 800); and it usually has a result field variable to which it sends output values at some fixed rate. There are four possible rates:

1. once only, at orchestra setup time (effectively a permanent assignment)
2. once at the beginning of each note (at initialization (init) time: i-rate)
3. once every performance-time control loop (perf-time control rate, or k-rate)
4. once each sound sample of every control loop (perf-time audio rate, or a-rate)

Orchestra Header Statements

The *Orchestra Header* contains global information that applies to all instruments and defines aspects of Csound output. It is sometimes referred to as *instr 0*, because it behaves as an instrument,

but without k- or a-rate processing (i.e. only opcodes and instructions that work at i-rate are allowed).

An *orchestra header statement* operates once only, at orchestra setup time. It is most commonly an assignment of some value to a *global reserved symbol*, e.g. `sr = 20000`. All orchestra header statements belong to a pseudo instrument 0, an *init* pass of which is run prior to all other instruments at score time 0. Any *ordinary statement* can serve as an orchestra header statement, eg. `gifreq = cpspch(8.09)` provided it is an init-time only operation. Statements that are normally placed in an orchestra header are:

- *Odbfs*
- *ctrlinit*
- *ftgen*
- *kr*
- *ksmps*
- *massign*
- *nchnls*
- *pgmassign*
- *pset*
- *seed*
- *sr*
- *strset*

For example, a Csound header may look like:

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
Odbfs = 1
massign 1, 10
```

Instrument and Opcode Block Statements

An instrument block is comprised of *ordinary statements* that set values, control the logical flow, or invoke the various signal processing subroutines that lead to audio output. Statements that define an instrument block are:

- *instr*
- *endin*

An instrument block looks like this:

```
instr 1 ;A simple sine wave oscillator
aout oscils 10000, 440, 0
out aout
endin
```

Statements that define a user defined opcode (UDO) block are

- *opcode*
- *endop*

See the *UDO* section for more information.

Ordinary Statements

An *ordinary statement* runs at either init time or performance time or both. Operations which produce a result formally run at the rate of that result (that is, at init time for i-rate results; at performance time for k- and a-rate results), with the sole exception of the *init* opcode. Most generators and modifiers, however, produce signals that depend not only on the instantaneous value of their arguments but also on some preserved internal state. These performance-time units therefore have an implicit init-time component to set up that state. The run time of an operation which produces no result is apparent in the opcode.

Arguments are values that are sent to an operation. Most arguments will accept arithmetic expressions composed of constants, variables, reserved symbols, value converters, arithmetic operations, and conditional values.

Constants and Variables

Constants are floating point numbers, such as 1, 3.14159, or -73.45. They are available continuously and do not change in value.

Variables are named cells containing numbers. They are available continuously and may be updated at one of the four update rates (setup only, i-rate, k-rate, or a-rate). i- and k-rate variables are scalars (i.e. they take on only one value at any given time) and are primarily used to store and recall controlling data, that is, data that changes at the note rate (for i-rate variables) or at the control rate (for k-rate variables). i- and k-variables are therefore useful for storing note parameter values, pitches, durations, slow-moving frequencies, vibratos, etc. a-rate variables, on the other hand, are arrays or vectors of information. Though renewed on the same perf-time control pass as k-rate variables, these array cells represent a finer resolution of time by dividing the control period into sample periods (see *ksmps*). a-rate variables are used to store and recall data changing at the audio sampling rate (e.g. output signals of oscillators, filters, etc.).

A further distinction is that between local and global variables. *local* variables are private to a particular instrument, and cannot be read from or written into by any other instrument. Their values are preserved, and they may carry information from pass to pass (e.g. from initialization time to performance time) within a single instrument. Local variable names begin with the letter *p*, *i*, *k*, or *a*. The same local variable name may appear in two or more different instrument blocks without conflict.

Global variables are cells that are accessible by all instruments. The names are either like local names preceded by the letter *g*, or are special reserved symbols. Global variables are used for broadcasting general values, for communicating between instruments (semaphores), or for sending sound from one instrument to another (e.g. mixing prior to reverberation).

Given these distinctions, there are eight forms of local and global variables:

Table 3. Types of Variables

Type	When Renewable	Local	Global
reserved symbols	permanent	--	rsymbol
score pfields	i-time	p number	--
init variables	i-time	i name	gi name
control signals	p-time, k-rate	k name	gk name

Type	When Renewable	Local	Global
audio signals	p-time, k-rate (all audio samples in a k-pass)	a name	ga name
spectral data types	k-rate	w name	--
streaming spectral data types	k-rate	f name	gf name
string variables	i-time and optionally k-rate	S name	gS name

Where *rsymbol* is a special reserved symbol (e.g. *sr*, *kr*), *number* is a positive integer referring to a score pfield or sequence number, and *name* is a string of letters, the underscore character, and/or digits with local or global meaning. As might be apparent, score parameters are local i-rate variables whose values are copied from the invoking score statement just prior to the init pass through an instrument, while MIDI controllers are variables which can be updated asynchronously from a MIDI file or MIDI device.

Variable Initialization

Opcodes that let one initialize variables are:

- *assign*
- *divz*
- *init*
- *tival*

Predefined Math Constant Macros

Csound defines several important math constants as *Macros*. You can see the full list *here*.

Expressions

Expressions may be composed to any depth. Each part of an expression is evaluated at its own proper rate. For instance, if the terms within a sub-expression all change at the control rate or slower, the sub-expression will be evaluated only at the control rate; that result might then be used in an audio-rate evaluation. For example, in

```
k1 + abs(int(p5) + frac(p5) * 100/12 + sqrt(k1))
```

the 100/12 would be evaluated at orch init, the p5 expressions evaluated at note i-time, and the remainder of the expression evaluated every k-period. The whole might occur in a unit generator argument position, or be part of an assignment statement.

Directories and Files

Many generators and the Csound command itself specify filenames to be read from or written to. These are optionally full pathnames, whose target directory is fully specified. When not a full path, filenames are sought in several directories in order, depending on their type and on the setting of certain environment variables. The latter are optional, but they can serve to partition and organize the directories so that source files can be shared rather than duplicated in several user directories. The environment variables can define directories for soundfiles SFDIR, sound samples SSDIR,

sound analysis SADIR, and include files for orchestra and score files INCDIR.

In Csound version 5.00 and later, these environment variables can specify multiple directories as a ; separated list. If a file is found in more than one location, the first one has the highest precedence.

The search order is:

1. Soundfiles being written are placed in SFDIR (if it exists), else the current directory.
2. Soundfiles for reading are sought in the current directory. If default paths are not disabled, files will next be sought for relative to the CSD/ORC/SCO file. Finally they will be sought in SSDIR and then SFDIR.
3. Analysis control files for reading are sought in the current directory. If default paths are not disabled, files will next be sought for relative to the CSD/ORC/SCO file. Finally they will be sought in SADIR.
4. MIDI files for reading are sought in the current directory. If default paths are not disabled, files will next be sought for relative to the CSD/ORC/SCO file. Finally they will be sought in MFDIR, SSDIR and SFDIR.
5. Files of code to be included in orchestra and score files (with *#include*) are sought first in the current directory, then in the same directory as the orchestra or score file (as appropriate), then finally INCDIR.

Nomenclature

Throughout this document, opcodes are indicated in **boldface** and their argument and result mnemonics, when mentioned in the text, are given in *italics*. Argument names are generally mnemonic (*amp*, *phs*), and the result is usually denoted by the letter *r*. Both are preceded by a type qualifier *i*, *k*, *a*, or *x* (e.g. *kamp*, *iphs*, *ar*). The prefix *i* denotes scalar values valid at note init time; prefixes *k* or *a* denote control (scalar) and audio (vector) values, modified and referenced continuously throughout performance (i.e. at every control period while the instrument is active). Arguments are used at the prefix-listed times; results are created at their listed times, then remain available for use as inputs elsewhere. With few exceptions, argument rates may not exceed the rate of the result. The validity of inputs is defined by the following:

- arguments with prefix *i* must be valid at init time;
- arguments with prefix *k* can be either control or init values (which remain valid);
- arguments with prefix *a* must be vector inputs;
- arguments with prefix *x* may be either vector or scalar (the compiler will distinguish).

All arguments, unless otherwise stated, can be expressions whose results conform to the above. Most opcodes (such as **linen** and **oscil**) can be used in more than one mode, which one being determined by the prefix of the result symbol.

Throughout this manual, the term "opcode" is used to indicate a command that usually produces an a-, k-, or i-rate output, and always forms the basis of a complete Csound orchestra statement. Items such as "+" or "*sin(x)*" or "(a >= b ? c : d)" are called "operators."

Macros

Orchestra macros work like C preprocessor macros, and replace the content of the macro in the orchestra before it is compiled. The opcodes one can use to create, call, or undefine orchestra macros are:

- *#define*
- *\$NAME*
- *#ifdef*
- *#ifndef*
- *#end*
- *#else*
- *#include*
- *#undef*

Orchestra macros can also be defined using the command line flag *--omacro:*.

More information and examples on the usage of orchestra macros can be found in the entry for *#define*.

These opcodes refer to orchestra macros; for score macros, refer to *Score Macros*.

Named Instruments

As a recent addition to the orchestra syntax, instruments can be defined with string names. Such named instruments are callable from the score, and are supported by a number of opcodes.

Syntax

A named instrument is declared as shown below:

```
instr Name[ , Name2[ , Name3[ , ... ] ] ]  
[ ... ]  
endin
```

A single instrument can have any number of names, and any of these names can be used to call the instrument. Additionally, it is possible to use numbers as name, denoting a standard numbered instrument, so the following declaration is also valid:

```
instr 100, Name1, 99, Name2, 1, 2, 3
```

An instrument name may consist of any number of letters, digits, and the underscore (`_`) character, however, the first character must not be a digit. Optionally, the instrument name may be prefixed with the '+' character (see below), for example:

```
instr +Reverb
```

For all instrument names, a number is automatically assigned (note: if the message level (`-m`) is not zero, these numbers are printed to the console during orchestra compilation), following these rules:

- any unused instrument numbers are taken up in ascending order, starting from 1
- the numbers are assigned in the order of instrument name definition, so named instruments that are defined later will always have a higher number (except if the '+' modifier is used)
- if the instrument name was prefixed with '+', the assigned number will be higher than that of any

of the (both numbered and named) other instruments without '+'. If there are multiple '+' instruments, the numbering of these will follow the order of definition, according to the above rule.

Using '+' is mainly useful for global output or effect instruments, that must be performed after the other instruments.

An example for instrument numbers:

```
instr 1, 2
endin

instr Instr1
endin

instr +Effect1, Instr2
endin

instr 100, Instr3, +Effect2, Instr4, 5
endin
```

In this example, the instrument numbers are assigned as follows:

```
Instr1: 3
Effect1: 101
Instr2: 4
Instr3: 6
Effect2: 102
Instr4: 7
```

Using Named Instruments

Named instruments can be called by using the name in double quotes as the instrument number (note: the '+' character should be omitted). Currently (as of Csound 4.22.4), named instruments are supported by:

- 'i' and 'q' score events



Notes

1. in score files, unmatched quotes, and spaces or other invalid characters in the strings should be avoided, otherwise (at least with current version) unpredictable behavior may occur (this problem does not exist for -L line events). However, there is checking for undefined instruments, and in such cases, the event is simply ignored with a warning.
2. Stand-alone utilities (score sort and extract) do not support named instruments. It is still possible to sort such scores by using the -t0 option of the main Csound executable)

- real-time line events (-L)
- event, schedkwhen, subinstr, and subinstrinit opcodes
- massign, pgmassign, prealloc, and mute opcodes

Additionally, there is a new opcode (nstrnum) that returns the number of a named instrument:

```
insno nstrnum "name"
```

With the above example, nstrnum "Effect1" would return 101. If an instrument with the specified

name does not exist, an init error occurs, and -1 is returned.

Example

```
; ---- orchestra ----
sr      = 44100
ksmps   = 10
nchnls  = 1

prealloc "SineWave", 20
prealloc "MIDISineWave", 20
massign 1, "MIDISineWave"

gaOutSend      init 0

instr +OutputInstr

out gaOutSend
clear gaOutSend

endin

instr SineWave
a1 oscils p4, p5, 0
vincr gaOutSend, a1

endin

instr MIDISineWave

iamp      veloc
inote     notnum
icps      = cpsoct(inote / 12 + 3)
a1        oscils iamp * 100, icps, 0
vincr     gaOutSend, a1

endin

; ---- score ----
i "SineWave" 0 2 12000 440
i "OutputInstr" 0 3
e
```

Author

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2002

User Defined Opcodes (UDO)

Csound allows the definition of opcodes inside the orchestra header using the opcodes *opcode* and *endop*. The defined opcode may run with a different number of control samples (*ksmps*) using *setksmps*.

To connect inputs and outputs for the UDO, use *xin* and *xout*.

An UDO looks like this:

```
opcode Lowpass, a, akk

setksmps 1                ; need sr=kr
ain, kal, ka2 xin         ; read input parameters
aout      init 0           ; initialize output
aout      = ain*ka1 + aout*ka2 ; simple tone-like filter
xout      aout             ; write output

endop
```

This UDO called *Lowpass* takes 3 inputs (the first is a-rate, and the next two are k-rate), and delivers 1 a-rate output. Notice the use of *xin* to receive inputs and *xout* to deliver outputs. Also note the use of *setksmps*, which is needed for the filter to work properly.

To use this UDO within an instrument, you would do something like:

```
afiltered Lowpass asource, kvalue1, kvalue2
```

See the entry for *opcode* for detailed information on UDO definition.

You can find many ready made UDO's (or contribute your own) at *Csounds.com* [<http://www.csounds.com/>]'s *User Defined Opcode Database* [<http://www.csounds.com/udo/>].

The Standard Numeric Score

The score section contains events that instantiate instruments from the orchestra. There are various score statements that enable complex score building within the csound language.

Currently, the maximum length of the score is $2^{31}-1$ control periods. For example, with `kr=1500`, you can run a score for a maximum of about 16.5 days before problems occur due to overflowing signed 32-bit integer variables.

Note also that when using single precision floats (i.e. the 'f' installers rather than the 'd' ones), the accuracy of timing becomes worse after performing for a long time.

Preprocessing of Standard Scores

A *Score* (a collection of score statements) is divided into time-ordered sections by the *s statement*. Before being read by the orchestra, a score is preprocessed one section at a time. Each section is normally processed by 3 routines: *Carry*, *Tempo*, and *Sort*.

Carry

Within a group of consecutive *i statements* whose p1 whole numbers correspond, any pfield left empty will take its value from the same pfield of the preceding statement. An empty pfield can be denoted by a single point (.) delimited by spaces. No point is required after the last nonempty pfield. The output of Carry preprocessing will show the carried values explicitly. The Carry Feature is not affected by intervening comments or blank lines; it is turned off only by a non-*i statement* or by an *i statement* with unlike p1 whole number.

Three additional features are available for p2 alone: +, ^+x, and ^-x. The symbol + in p2 will be given the value of p2 + p3 from the preceding i statement. This enables note action times to be automatically determined from the sum of preceding durations. The + symbol can itself be carried. It is legal only in p2. E.g.: the statements

```
i1  0      .5      100
i  .  +
i
```

will result in

```
i1  0      .5      100
i1  .5     .5      100
i1  1      .5      100
```

The symbols ^+x and ^-x determine the current p2 by adding or subtracting, respectively, the value of x from the preceding p2. These may be used in p2 only and are *not* carried like the + symbol. Note also that there should be no spaces following the ^, the +, or the - parts of these symbols -- the number must come directly after as in ^+2.3. If the example above had been

```
i1  0      .5      100
i  .  ^+1
i  .  ^+1
```

the result would instead be

```
i1  0      .5      100
```

```
i1 1 .5 100
i1 2 .5 100
```

The Carry feature should be used liberally. Its use, especially in large scores, can greatly reduce input typing and will simplify later changes.

There can sometimes be circumstances where you do not want "missing" pfields after the last one entered to be implicitly carried. An example would be an instrument that is designed to take a variable number of pfields. Beginning with Csound 5.08, you can prevent the implicit carrying of pfields at the end of an *i* statement by using the symbol ! (called the "no-carry symbol"). The ! must appear at the end of an *i* statement and it cannot be used in p1, p2, or p3, since these pfields are required. Here is an example:

```
i1 0 .5 100
i . +
i . . . !
i
```

This score would be interpreted as

```
i1 0 .5 100
i1 .5 .5 100
i1 1 .5 ; no p4
i1 1.5 .5 ; only p1 to p3 are carried here
```

Tempo

This operation time warps a score section according to the information in a *t statement*. The tempo operation converts p2 (and, for *i statements*, p3) from original beats into real seconds, since those are the units required by the orchestra. After time warping, score files will be seen to have orchestra-readable format demonstrated by the following:

```
i p1 p2beats p2seconds p3beats p3seconds p4 p5 ....
```

Sort

This routine sorts all action-time statements into chronological order by p2 value. It also sorts coincident events into precedence order. Whenever an *f statement* and an *i statement* have the same p2 value, the *f statement* will precede. Whenever two or more *i statements* have the same p2 value, they will be sorted into ascending p1 value order. If they also have the same p1 value, they will be sorted into ascending p3 value order. Score sorting is done section by section (see *s statement*). Automatic sorting implies that score statements may appear in any order within a section.



Note

The operations Carry, Tempo and Sort are combined in a 3-phase single pass over a score file, to produce a new file in orchestra-readable format (see the Tempo example). Processing can be invoked either explicitly by the *Scsort* command, or implicitly by Csound which processes the score before calling the orchestra. Source-format files and orchestra-readable files are both in ASCII character form, and may be either perused or further modified by standard text editors. User-written routines can be used to modify score files before or after the above processes, provided the final orchestra-readable statement format is not violated. Sections of different formats can be sequentially batched; and sections of like format can be merged for automatic sorting.

Score Statements

The statements used in scores are:

- *a* - Advance score time by a specified amount
- *b* - Resets the clock
- *e* - Marks the end of the last section of the score
- *f* - Causes a *GEN subroutine* to place values in a stored function table
- *i* - Makes an instrument active at a specific time and for a certain duration
- *m* - Sets a named mark in the score
- *n* - Repeats a section
- *q* - Used to quiet an instrument
- *r* - Starts a repeated section
- *s* - Marks the end of a section
- *t* - Sets the tempo
- *v* - Provides for locally variable time warping of score events
- *x* - Skip the rest of the current section
- *{* - Begins a non-sectional, nestable loop.
- *}* - Ends a non-sectional, nestable loop.

Next-P and Previous-P Symbols

At the close of any of the operations *Carry*, *Tempo*, and *Sort*, three additional score features are interpreted during file writeout: next-p, previous-p, and *ramping*.

i statement pfields containing the symbols *np_x* or *pp_x* (where *x* is some integer) will be replaced by the appropriate pfield value found on the next *i statement* (or previous *i statement*) that has the same p1. For example, the symbol *np7* will be replaced by the value found in p7 of the next note that is to be played by this instrument. *np* and *pp* symbols are recursive and can reference other *np* and *pp* symbols which can reference others, etc. References must eventually terminate in a real number or a *ramp symbol*. Closed loop references should be avoided. *np* and *pp* symbols are illegal in p1, p2 and p3 (although they may reference these). *np* and *pp* symbols may be Carried. *np* and *pp* references cannot cross a Section boundary. Any forward or backward reference to a non-existent note-statement will be given the value zero.

E.g.: the statements

```
i1  0    1    10   np4  pp5
i1  1    1    20
i1  1    1    30
```

will result in

```
i1  0    1    10   20   0
i1  1    1    20   30   20
```

```
i1  2  1  30  0  30
```

np and *pp* symbols can provide an instrument with contextual knowledge of the score, enabling it to glissando or crescendo, for instance, toward the pitch or dynamic of some future event (which may or may not be immediately adjacent). Note that while the *Carry* feature will propagate *np* and *pp* through unsorted statements, the operation that interprets these symbols is acting on a time-warped and fully sorted version of the score.

Ramping

i statement pfields containing the symbol < will be replaced by values derived from linear interpolation of a time-based ramp. Ramps are anchored at each end by the first real number found in the same pfield of a preceding and following note played by the same instrument. E.g.: the statements

```
i1  0  1  100
i1  1  1  <
i1  2  1  <
i1  3  1  400
i1  4  1  <
i1  5  1  0
```

will result in

```
i1  0  1  100
i1  1  1  200
i1  2  1  300
i1  3  1  400
i1  4  1  200
i1  5  1  0
```

Ramps cannot cross a Section boundary. Ramps cannot be anchored by an *np* or *pp* symbol (although they may be referenced by these). Ramp symbols are illegal in p1, p2 and p3. Ramp symbols may be Carried. Note, however, that while the Carry feature will propagate ramp symbols through unsorted statements, the operation that interprets these symbols is acting on a time-warped and fully sorted version of the score. In fact, time-based linear interpolation is based on warped score-time, so that a ramp which spans a group of accelerating notes will remain linear with respect to strict chronological time.

Starting with Csound version 3.52, using the symbols (or) will result in an exponential interpolation ramp, similar to *expon*. Using the symbol ~ (a tilde) will result in uniform, random distribution between the first and last values of the ramp. Use of these functions must follow the same rules as the linear ramp function.

Score Macros

Description

Macros are textual replacements which are made in the score as it is being presented to the system. The macro system in Csound is a very simple one, and uses the characters # and \$ to define and call macros. This can allow for simpler score writing, and provide an elementary alternative to full score generation systems. The score macro system is similar to, but independent of, the macro system in the orchestra language.

#define NAME -- defines a simple macro. The name of the macro must begin with a letter and can consist of any combination of letters and numbers. Case is significant. This form is limiting, in that the variable names are fixed. More flexibility can be obtained by using a macro with arguments, described below.

#define NAME(*a' b' c'*) -- defines a macro with arguments. This can be used in more complex situations. The name of the macro must begin with a letter and can consist of any combination of letters and numbers. Within the replacement text, the arguments can be substituted by the form: \$A. In fact, the implementation defines the arguments as simple macros. There may be up to 5 arguments, and the names may be any choice of letters. Remember that case is significant in macro names.

\$NAME. -- calls a defined macro. To use a macro, the name is used following a \$ character. The name is terminated by the first character which is neither a letter nor a number. If it is necessary for the name not to terminate with a space, a period, which will be ignored, can be used to terminate the name. The string, *\$NAME.*, is replaced by the replacement text from the definition. The replacement text can also include macro calls.

#undef NAME -- undefines a macro name. If a macro is no longer required, it can be undefined with *#undef* NAME.

Syntax

```
#define NAME # replacement text #

#define NAME(a' b' c') # replacement text #

$NAME.

#undef NAME
```

Initialization

replacement text # -- The replacement text is any character string (not containing a #) and can extend over multiple lines. The replacement text is enclosed within the # characters, which ensure that additional characters are not inadvertently captured.

Performance

Some care is needed with textual replacement macros, as they can sometimes do strange things. They take no notice of any meaning, so spaces are significant. This is why, unlike the C programming language, the definition has the replacement text surrounded by # characters. Used carefully, this simple macro system is a powerful concept, but it can be abused.

Another Use For Macros. When writing a complex score it is sometimes all too easy to forget to what the various instrument numbers refer. One can use macros to give names to the numbers. For example

```
#define Flute #i1#
#define Whoop #i2#

$Flute. 0 10 4000 440
$Whoop. 5 1
```

Examples

Example 1. Simple Macro

A note-event has a set of p-fields which are repeated:

```
#define ARGS # 1.01 2.33 138#
```

```
i1 0 1 8.00 1000 $ARGS
i1 0 1 8.01 1500 $ARGS
i1 0 1 8.02 1200 $ARGS
i1 0 1 8.03 1000 $ARGS
```

This will get expanded before sorting into:

```
i1 0 1 8.00 1000 1.01 2.33 138
i1 0 1 8.01 1500 1.01 2.33 138
i1 0 1 8.02 1200 1.01 2.33 138
i1 0 1 8.03 1000 1.01 2.33 138
```

This can save typing, and is makes revisions easier. If there were two sets of p-fields one could have a second macro (there is no real limit on the number of macros one can define).

```
#define ARGS1 # 1.01 2.33 138#
#define ARGS2 # 1.41 10.33 1.00#
i1 0 1 8.00 1000 $ARGS1
i1 0 1 8.01 1500 $ARGS2
i1 0 1 8.02 1200 $ARGS1
i1 0 1 8.03 1000 $ARGS2
```

Example 2. Macros with arguments

```
#define ARG(A) # 2.345 1.03 $A 234.9#
i1 0 1 8.00 1000 $ARG(2.0)
i1 + 1 8.01 1200 $ARG(3.0)
```

which expands to

```
i1 0 1 8.00 1000 2.345 1.03 2.0 234.9
i1 + 1 8.01 1200 2.345 1.03 3.0 234.9
```

Credits

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Bath, UK

April, 1998 (New in Csound version 3.48)

Multiple File Score

Description

Using the score in more than one file.

Syntax

```
#include "filename"
```


Performance

It is sometimes convenient to have the score in more than one file. This use is supported by the *#include* facility which is part of the macro system. A line containing the text

```
#include "filename"
```

where the character " can be replaced by any suitable character. For most uses the double quote symbol will probably be the most convenient. The file name can include a full path.

This takes input from the named file until it ends, when input reverts to the previous input. There is currently a limit of 20 on the depth of included files and macros.

A suggested use of *#include* would be to define a set of macros which are part of the composer's style. It could also be used to provide repeated sections.

```
S
#include :section1:
;; Repeat that
S
#include :section1:
```

Alternative methods of doing repeats, use the *r statement*, *m statement*, and *n statement*.

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April, 1998 (New in Csound version 3.48)

Thanks to Luis Jure for pointing out the incorrect syntax in multiple file include statement.

Evaluation of Expressions

In earlier versions of Csound the numbers presented in a score were used as given. There are occasions when some simple evaluation would be easier. This need is increased when there are macros. To assist in this area the syntax of arithmetic expressions within square brackets [] has been introduced. Expressions built from the operations +, -, *, /, % ("modulo"), and ^ ("power of") are allowed, together with grouping with (). Unary minus and plus are also supported. The expressions can include numbers, and naturally macros whose values are numeric or arithmetic strings. All calculations are made in floating point numbers. The usual precedence rules are followed when evaluating: expressions within parantheses () are evaluated first and ^ is evaluated before *, /, and % which are evaluated before + and -.

In addition to arithmetic operations, the following bitwise logical operators are also available: & (AND), | (OR), and # (XOR, exclusive-OR). These operators round their operands to the nearest (long) integer before evaluating. The logical operators have the same precedence as the *, /, and % arithmetic operators.

New in Csound version 3.56 are @*x* (next power-of-two greater than or equal to *x*) and @@*x* (next power-of-two-plus-one greater than or equal to *x*).

Finally, the tilde symbol ~ can be used in an expression wherever a number is permissible to use. Each ~ will evaluate to a random value between zero (0) and one (1).

Example

```
r3 CNT
i1 0 [0.3*$CNT.]
i1 + [($CNT./3)+0.2]
e
```

As the three copies of the section have the macro \$CNT. with the different values of 1, 2 and 3, this expands to

```
s
i1 0 0.3
i1 0.3 0.533333
s
i1 0 0.6
i1 0.6 0.866667
s
i1 0 0.9
i1 0.9 1.2
e
```

This is an extreme form, but the evaluation system can be used to ensure that repeated sections are subtly different.

Here are some simple examples of each operator:

```
i1 0 1 [ 110 + 220 ] ; evaluates to 330
i1 + . [ 330 - 55 ] ; 275
i1 + . [ 44 * 10 ] ; 440
i1 + . [ 1100 / 2 ] ; 550
i1 + . [ 5 ^ 4 ] ; 625
i1 + . [ 5660 % 1000 ] ; 660
i1 + . [ 110 & 220 ] ; 76
i1 + . [ 110 | 220 ] ; 254
i1 + . [ 110 # 220 ] ; 178
i1 + . [ ~ ] ; random between 0-1
i1 + . [ ~ * 4 + 1 ] ; random between 1-5
i1 + . [ ~ * 95 + 5 ] ; random between 5-100

i1 + . [ 8 / 2 * 3 ] ; 12
i1 + . [ 4 + 3 - 2 + 1 ] ; 6
i1 + . [ 4 + 3 * 2 + 1 ] ; 11
i1 + . [ (4 + 3)*(2 + 1) ] ; 21

i1 + . [ 2 * 2 & 3 ] ; 4
i1 + . [ 3 & 2 * 2 ] ; 0
i1 + . [ 4 | 3 * 3 ] ; 13
```

Credits

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Bath, UK

April, 1998 (New in Csound version 3.48)

Strings in p-fields

You can pass a string as a p-field instead of a number, like this:

i 1 0 10 "A4"

The string can be received by the instrument and further processed using the *string opcodes*.



Note

Currently only one p-field can contain a string (i.e. no more than one string per line is allowed). You can overcome this using *strset* and *strget*.

Front Ends

Here's a (far from complete) list of front-ends available for Csound.

Csound5GUI

Csound5GUI is a cross-platform, versatile GUI which is part of the standard Csound distribution. It implements most configuration features of Csound.

CSDplayer

This is a simple java program to play csd files. It is included in the standard distribution.

Winsound

Also part of the main Csound tree (though not available in all distributions), Winsound is cross-platform FLTK port of Barry Vercoe's original front-end for csound.

WinXoundPro

A convenient front-end for windows with syntax highlighting. You can get it at the WinXound Front Page [<http://www.ibiart.it/winxound/index.html>].

Csound Editor

A convenient front-end for windows with syntax highlighting. You can get it at the Flavio Tordini's Home Page [<http://flavio.tordini.org/csound-editor/>].

MacCsound

More than a front-end for the Mac at MacCsound Page [<http://www.csounds.com/matt/MacCsound/>].

Cabel

Cabel is a graphical user interface for building csound instruments by patching modules similar to modular synthesizers. Cross-platform, written in Python. At <http://cabel.sourceforge.net/>.

Blue

Composition oriented front-end written in Java. It's interface is much like a digital multitrack, but differs in that there timelines within timelines (polyObjects). This allows for a compositional organization in time that seems to me to be very intuitive, informative, and flexible. Get it at: Blue Home Page [<http://csounds.com/stevenyi/blue/>].

CsoundAC

Python Scripting

You can use CsoundAC as a Python extension module. You can do this in a standard Python interpreter, such as Python command line or the Idle Python GUI.

To use CsoundAC in a standard Python interpreter, import CsoundAC.

```
import CsoundAC
```

The CsoundAC module automatically creates an instance of CppSound named csound, which provides an object-oriented interface to the Csound API. In a standard Python interpreter, you can load a Csound .csd file and perform it like this:

```
C:\Documents and Settings\mkg>python
Python 2.3.3 (#51, Dec 18 2003, 20:22:39) [MSC v.1200 32 bit (Intel)] on win32
Type "help", "copyright", "credits" or "license" for more information.
>>> import CsoundAC
>>> csound.load("c:/projects/csound5/examples/trapped.csd")
1
>>> csound.exportForPerformance()
1
>>> csound.perform()
BEGAN CppSound::perform(5, 988ee0)...
BEGAN CppSound::compile(5, 988ee0)...
Using default language
0dBFS level = 32767.0
Csound version 5.00 beta (float samples) Jun 7 2004
libsndfile-1.0.10pre6
orchname: temp.orc
scorename: temp.sco
orch compiler:
398 lines read
instr 1
instr 2
instr 3
instr 4
instr 5
instr 6
instr 7
instr 8
instr 9
instr 10
instr 11
instr 12
instr 13
instr 98
instr 99
sorting score ...
... done
Csound version 5.00 beta (float samples) Jun 6 2004
displays suppressed
0dBFS level = 32767.0
orch now loaded
audio buffered in 16384 sample-frame blocks
SFDIR undefined. using current directory
writing 131072-byte blks of shorts to test.wav
WAV
SECTION 1:
ENDED CppSound::compile.
ftable 1:
ftable 2:
ftable 3:
ftable 4:
ftable 5:
ftable 6:
ftable 7:
ftable 8:
ftable 9:
ftable 10:
ftable 11:
ftable 12:
ftable 13:
ftable 14:
ftable 15:
ftable 16:
ftable 17:
ftable 18:
ftable 19:
ftable 20:
ftable 21:
ftable 22:
new alloc for instr 1:
B 0.000 .. 1.000 T 1.000 TT 1.000 M: 32.7 0.0
new alloc for instr 1:
B 1.000 .. 3.600 T 3.600 TT 3.600 M: 207.6 0.1
...
B 93.940 .. 94.418 T 98.799 TT281.799 M: 477.6 85.0
B 94.418 ..100.000 T107.172 TT290.172 M: 118.9 11.5
```

```
end of section 4          sect peak amps:  25950.8  26877.4
inactive allocs returned to freespace
end of score.            overall amps:  32204.8  31469.6
overall samples out of range:      0      0
0 errors in performance
782 131072-byte soundblks of shorts written to test.wav WAV
Elapsed time = 13.469000 seconds.
ENDED CppSound::perform.
1
>>>
```

The `koch.py` script shows how to use Python to do algorithmic composition for Csound. You can use Python triple-quoted string literals to hold your Csound files right in your script, and assign them to Csound:

```
csound.setOrchestra('''sr = 44100
kr = 441
ksmps = 100
nchnls = 2
0dbfs = .1
instr 1,2,3,4,5 ; FluidSynth General MID
I; INITIALIZATION
; Channel, bank, and program determine the preset, that is, the actual sound.
ichannel      =          p1
iprogram      =          p6
ikey          =          p4
ivelocity     =          p5 + 12
ijunk6        =          p6
ijunk7        =          p7
; AUDIO
istatus       =          144;
print         iprogram, istatus, ichannel, ikey, ivelocity, aleft, aright
fluid         "c:/projects/csound5/samples/VintageDreamsWaves-v2.sf2", \\
iprogram, istatus, ichannel, ikey, ivelocity, 1
outs          aleft, arightendin''')
csound.setCommand("csound --opcode-lib=c:/projects/csound5/fluid.dll \\
-RWdfo ./koch.wav ./temp.orc ./temp.sco")
csound.exportForPerformance()
csound.perform()
```

CsoundVST

CsoundVST is a multi-function front end for Csound, based on the Csound API. CsoundVST runs as a stand-alone graphical user interface to Csound, and it also runs as a VST instrument or effect plugin in VST hosts such as Cubase with the same user interface. CsoundVST is part of the main csound source tree, but is not included in standard distributions, due to licensing limitations of Steinberg's VST SDK.

Standalone

To run CsoundVST as a stand-alone front end to Csound, execute CsoundVST. When the program has loaded, you will see a graphical user interface with a row of buttons along the top. Click on the *Open...* button to load a `.csd` file. You can also click on the *Open...* button and load a `.orc` file, then click on the *Import...* button to add a `.sco` file. You can edit the Csound command, the orchestra file, or the score file in the respective tabs of the user interface. When all is satisfactory, click on the *Perform* button to run Csound. You can stop a performance at any time by clicking on the *Stop* button.

VST Plugin

The following instructions are for Cubase 4.0. You would follow roughly similar procedures in other hosts.

Use the *Devices* menu, *Plug-In Information* dialog, *VST Plug-Ins* tab, *VST 2.x Plug-in Paths* dialog, *Add* button to add your `csound/bin` directory to Cubase's plugin path. You can have multiple directories separated by semicolons. Then select the CsoundVST path and click on the *Set as Shared Folder* button.

Quit Cubase, and start it again.

Use the *File* menu, *New Project* dialog to create a new song.

Use the *Project* menu, *Add Track* submenu, to add a new MIDI track.

Use the pencil tool to draw a *Part* on the track a few measures long. Write some music in the *Part* using the *Event* editor or the *Score* editor.

Use the *Devices* menu (or the F11 key) to open the *VST Instruments* dialog.

Click on one of the *No VST Instrument* labels, and select *CsoundVST* from the list that pops up.

Click on the *e* (for edit) button to open the *CsoundVST* dialog.

On the Settings page, check the *Instrument* box in the VST Plugin group, and the *Classic* box in the *Csound performance mode* group. Then click on the *Apply* button.

Click on the *Open* button to bring up the file selector dialog. Navigate to a directory containing a Csound csd file suitable for MIDI performance, such as `csound/examples/CsoundVST.csd`. Click on the OK button to load the file. You can also open and import a suitable `.orc` and `.sco` file as described above.

In any event, the command line in the *Classic Csound command line* text box must specify `-+rtmidi=null -M0`, and should read something like this:

```
csound -f -h -+rtmidi=null -M0 -d -n -m7 --midi-key-oct=4 --midi-velocity=5 temp.orc temp.sco
```

Click on the *VST Instruments* dialog's on/off button to turn it on. This should compile the Csound orchestra.

In the *Cubase Track Inspector*, click on the *out: Not Assigned* label and select *CsoundVST* from the list that pops up.

On the ruler at the top of the *Arrangement* window, select the loop end point and drag it to the end of your part, then click on the loop button to enable looping.

Click on the *play* button on the *Transport* bar. You should hear your music played by CsoundVST.

Try assigning your track to different channels; a different Csound instrument will perform each channel.

When you save your song, your Csound orchestra will be saved as part of the song and re-loaded when you re-load the song.

You can click on the *Orchestra* tab and edit your Csound instruments while CsoundVST is playing. To hear your changes, just click on the *CsoundVST Perform* button to recompile the orchestra.

You can assign up to 16 channels to a single CsoundVST plugin.

TclCsound

TclCsound was introduced to provide a simple scripting interface to Csound. Tcl is a simple language that is easy to extend and provide nice facilities such as easy file access and TCP networking. With its Tk component, it can also handle a graphic and event interface. TclCsound provides three 'points of contact' with Tcl:

1. a csound-aware tcl interpreter (cstclsh)
2. a csound-aware windowing shell (cswish)
3. a csound commands module for Tcl/Tk (tclcsound dynamic lib)

The Tcl interpreter: cstclsh

With cstclsh, it is possible to have interactive control over a csound performance. The command starts an interactive shell, which holds an instance of Csound. A number of commands can then be used to control it. For instance, the following command can compile csound code and load it in memory ready for performance:

```
csCompile -odac orchestra score -m0
```

Once this is done, performance can be started in two ways: using csPlay or csPerform . The command

```
csPlay
```

will start the Csound performance in a separate thread and return to the cstclsh prompt. A number of commands can then be used to control Csound. For instance,

```
csPause
```

will pause performance; and

```
csRewind
```

will rewind to the beginning of the note-list. The csNote, csTable and csEvent commands can be used to add Csound score events to the performance, on-the-fly. The csPerform command, as opposed to csPlay , will not launch a separate thread, but will run Csound in the same thread, returning only when the performance is finished. A variety of other commands exist, providing full control of Csound.

Cswish: the windowing shell

With Cswish, Tk widgets and commands can be used to provide graphical interface and event handling. As with cstclsh, running the cswish command also opens an interactive shell. For instance, the following commands can be used to create a transport control panel for Csound:

```
frame .fr
button .fr.play -text play -command csPlay
button .fr.pause -text pause -command csPause
button .fr.rew -text rew -command csRewind
pack .fr .fr.play .fr.pause .fr.rew
```

Similarly, it is possible to bind keys to commands so that the computer keyboard can be used to play Csound.

Particularly useful are the control channel commands that TclCsound provides. For instance, named IO channels can be registered with TclCsound and these can be used with the invalue, outvalue opcodes. In addition, the Csound API also provides a complete software bus for audio, control and

string channels. It is possible in TclCsound to access control and string bus channels (the audio bus is not implemented, as Tcl is not able to handle such data). With these TclCsound commands, Tk widgets can be easily connected to synthesis parameters.

A Csound server

In Tcl, setting up TCP network connections is very simple. With a few lines of code a csound server can be built. This can accept connections from the local machine or from remote clients. Not only Tcl/Tk clients can send commands to it, but TCP connections can be made from other software, such as, for instance, Pure Data (PD). A Tcl script that can be run under the standard tclsh interpreter is shown below. It uses the Tclcsound module, a dynamic library that adds the Csound API commands to Tcl.

```
# load tclcsound.so
#(OSX: tclcsound.dylib, Windows: tclcsound.dll)
load tclcsound.so Tclcsound
set forever 0

# This arranges for commands to be evaluated
proc ChanEval { chan client } {
  if { [catch { set rtn [eval [gets $chan]] } err] } {
    puts "Error: $err"
  } else {
    puts $client $rtn
    flush $client
  }
}

# this arranges for connections to be made

proc NewChan { chan host port } {
  puts "Csound server: connected to $host on port $port ($chan)"
  fileevent $chan readable [list ChanEval $chan $host]
}

# this sets up a server to listen for
# connections

set server [socket -server NewChan 40001]
set sinfo [fconfigure $server -sockname]
puts "Csound server: ready for connections on port [lindex $sinfo 2]"
vwait forever
```

With the server running, it is then possible to set up clients to control the Csound server. Such clients can be run from standard Tcl/Tk interpreters, as they do not evaluate the Csound commands themselves. Here is an example of client connections to a Csound server, using Tcl:

```
# connect to server
set sock [socket localhost 40001]

# compile Csound code
puts $sock "csCompile -odac orchestra score"
flush $sock

# start performance
puts $sock "csPlay"
flush $sock

# stop performance
puts $sock "csStop"
flush $sock
```

As mentioned before, it is possible to set up clients using other software systems, such as PD. Such clients need only to connect to the server (using a netsend object) and send messages to it. The first item of each message is taken to be a command. Further items can optionally be added to it as arguments to that command.

A Scripting Environment

With TclCsound, it is possible to transform the popular text editor e-macs into a Csound scripting/performing environment. When in Tcl mode, the editor allows for Tcl expressions to be evaluated by selection and use of a simple escape sequence (Ctrl-C Ctrl-X). This facility allows the integrated editing and performance of Csound and Tcl/Tk code.

In Tcl it is possible to write score and orchestra files that can be saved, compiled and run by the same script, under the e-macs environment. The following example shows a Tcl script that builds a csound instrument and then proceeds to run a csound performance. It creates 10 slightly detuned parallel oscillators, generating sounds similar to those found in Risset's *Inharmonique*.

```
load tclcsound.so Tclcsound

# set up some intermediary files

set orcfile "tcl.orc"
set scofile "tcl.sco"
set orc [open $orcfile w]
set sco [open $scofile w]

# This Tcl procedure builds an instrument
proc MakeIns { no code } {
  global orc sco
  puts $orc "instr $no"
  puts $orc $code
  puts $orc "endin"
}

# Here is the instrument code
append ins "asum init 0 \n"
append ins "ifreq = p5 \n"
append ins "iamp = p4 \n"

for { set i 0 } { $i < 10 } { incr i } {
  append ins "a$i oscili iamp,
ifreq+ifreq*[expr $i * 0.002], 1\n"
}

for { set i 0 } { $i < 10 } { incr i } {
  if { $i } {
    append ins " + a$i"
  } else {
    append ins "asum = a$i "
  }
}

append ins "\nk1 linen 1, 0.01, p3, 0.1 \n"
append ins "out asum*k1"

# build the instrument and a dummy score

MakeIns 1 $ins
puts $sco "f0 10"
close $orc
close $sco
```

```
# compile
csCompile $orcfile $scofile -odac -d -m0

# set a wavetable
csTable 1 0 16384 10 1 .5 .25 .2 .17 .15 .12 .1

# send in a sequence of events and perform it
for {set i 0} { $i < 60 } { incr i } {
  csNote 1 [expr $i * 0.1] .5 \
  [expr ($i * 10) + 500] [expr 100 + $i * 10]
}
csPerform

# it is possible to run it interactively as
# well
csNote 1 0 10 1000 200
csPlay
```

The use of such facilities as provided by e-macs can emulate an environment not unlike the one found under the so-called 'modern synthesis systems', such as SuperCollider (SC). In fact, it is possible to run Csound in a client-server set-up, which is one of the features of SC3. A major advantage is that Csound provides about three or four times the number of unit generators found in that language (as well as providing a lower-level approach to signal processing, in fact these are but a few advantages of Csound).

TclCsound as a language wrapper

It is possible to use TclCsound at a slightly lower level, as many of the C API functions have been wrapped as Tcl commands. For instance it is possible to create a 'classic' Csound command-line frontend completely written in Tcl. The following script demonstrates this:

```
#!/usr/local/bin/cstclsh

set result 1
csCompileList $argv
while { $result != 0 } {
  set result csPerformKsmpls
}
```

TclCsound Command Reference

Performance control commands:

csCompile [csound command-line] : compiles an orc/sco/csd + any options

csCompileList arglist : compiles an orc/sco/csd + options given as a Tcl list 'arglist'

csPerform : plays the score, returning when finished

csPerformKsmpls : performs one ksmpls block of audio samples, returning when finished

csPerformBuffer : performs one buffersize block of audio samples, returning when finished

csPlay : starts asynchronous performance in a separate thread, returning immediately

csPause : pauses playback

csStop : stops performance and resets csound

csRewind : rewinds the score

csOffset secs : offsets score playback by secs

csGetoffset : returns the score offset in secs

csGetScoreTime : returns the score time in secs

Event commands:

csNote [p-fields] : sends in a i-statement event

csTable [p-fields] : sends in a f-statement event

csEvent opcode [p-fields] : sends in a score event defined by 'opcode' plus p-fields

csNoteList arglist : sends in a i-statement event with p-fields as a Tcl list 'arglist'

csTableList arglist : sends in a f-statement event with p-fields as a Tcl list 'arglist'

csEventList arglist : sends in a score event defined by 'opcode' plus p-fields as a Tcl list 'arglist'

Invalue, outvalue, pvsin, pvsout control and string channel commands:

csInChannel name : registers a csound invalue channel

csOutChannel name : registers a csound outvalue channel and creates tcl global variable 'name'

csInValue channel value : sets the value of a csound invalue channel

csOutValue channel : returns the value of a csound outvalue channel

csPvsIn number [size olaps wsize wtype]: registers a pvs in bus channel, optionally initialising fsig values for fftsize to 'size' (default:1024), overlaps to 'olaps' (def.: size/4), window size to 'wsize' (def.: size) and window type to 'wtype' (def.: 1, Hanning window, see manual page for pvsanal). Works with pvsin opcode (PVS_AMP_FREQ format only).

csPvsOut number [size olaps wsize wtype]: registers a pvs out bus channel. Works with opcode pvsout (PVS_AMP_FREQ format only).

csPvsInSet channel bin amp freq: sets the amp and freq of a bin of the pvs in channel number.

csPvsOutGet channel bin [isFreq]: returns the amp or freq of a bin of the pvs out channel number. The optional argument 'isFreq' (default: 0) controls whether the returned value is the bin amp (0) or freq (1).

csSetControlChannel channel value : sets the value of control channel 'channel', creating it if it does not exist

csGetControlChannel channel : returns the value of control channel 'channel'; creates the channel it if it does not exist

csSetStringChannel channel string : sets the string channel 'channel', creating it if it does not exist

csGetStringChannel channel : returns the string in channel 'channel'; creates the channel it if it does not exist

Message commands:

csMessageOutput var: appends all csound messages to the tcl variable var.

Table commands:

csGetTableSize ftn : returns the size of function table ftn (-1 if non-existent)

csSetTable ftn index value : sets the value of position 'index' to 'value' in function table 'ftn'

csGetTable ftn index : returns the value of position 'index' in function table 'ftn'

Environment variable commands:

csOpcodedir opcodedir : sets the opcode directory

csSetenv envvar value : sets any environment variable (eg. SFDIR, SADIR)

Building Csound

Csound has become a complex project and can involve many dependencies. Unless you are a Csound developer or need to develop Csound plugins, you should try to use one of the precompiled distributions from <http://www.sourceforge.net/projects/csound>. However, building from source is probably the best option on GNU/Linux.

This section focuses on the main Csound 5 build system, which uses SCons [<http://www.scons.org>], a Python program that replaces *make* for cross-platform configuration and building.

When building Csound from source instead of using a precompiled package, you first need to obtain the sources for a release of Csound at <http://www.sourceforge.net/projects/csound>. The source packages have either a zip or tar.gz extension.

The latest (possibly unstable) Csound source code is also available through the Concurrent Versions System (CVS). It's likely (if you're running Mac OS X or Linux) that you already have CVS installed on your machine. If not, it can be downloaded from (<http://www.cvshome.org>). There are many graphical front ends for cvs, but you can easily get the sources using the command line version.

The Csound CVS front page is located at: http://sourceforge.net/cvs/?group_id=81968. Information about accessing the CVS repository may be found in the SourceForge document <http://sourceforge.net/docs/E04/>. To download Csound sources using CVS, run the following commands (from a terminal or DOS shell):

```
cvs -d:pserver:anonymous@csound.cvs.sourceforge.net:/cvsroot/csound login
cvs -z3 -d:pserver:anonymous@csound.cvs.sourceforge.net:/cvsroot/csound co -P csound5
```

To update the Csound5 sources you already have in your csound5 directory, change to that directory, and then type:

```
cvs -z3 update -d
```

To update only a single file, change to the sources directory, and then type:

```
cvs -z3 update filename
```

Basic requirements to build Csound 5 on any platform

- Install libsndfile version 1.0.13 or later from www.mega-nerd.com/libsndfile [<http://www.mega-nerd.com/libsndfile>].
- Install Python from www.python.org [<http://www.python.org>]. In most cases it is best to install the most recent stable version. Python is needed for SCons to run.
- Install the SCons build system from www.scons.org [<http://www.scons.org>].

These are the minimum requirements for a build, but csound has many optional components which enhance functionality and add opcodes which may require additional libraries.

Optional configurations (ALL platforms)

In most cases it is best to install the most recent stable versions of the optional libraries.

- Real-time audio can use the cross-platform PortAudio library (trunk version or devel-19 branch) from www.portaudio.com/usingcvs.html [<http://www.portaudio.com/usingcvs.html>]. Please note that stable version 18 will not work. Csound can also use several platform specific audio APIs like ALSA, JACK, CoreAudio and the Windows multimedia library, see each platform notes for details.
- Real-time MIDI can use the cross-platform PortMidi library from www.cs.cmu.edu/~music/portmusic [<http://www.cs.cmu.edu/~music/portmusic>]
- For GUI widgets, install FLTK 1.1 or 1.3 from www.fltk.org [<http://www.fltk.org>]. You must configure and build FLTK with `--enable-shared --enable-threads`.
- For generating Python and Java interfaces, install the Software Interface and Wrapper Generator (SWIG) from <http://www.swig.org>.
- *CsoundAC* requires FLTK and the boost C++ template libraries for random numbers and linear algebra, from <http://www.boost.org>. *CsoundAC* requires at least version 1.32.1.
- The fluid opcodes require the Fluidsynth library from <http://savannah.nongnu.org/download/fluid>.
- The OSC opcodes require the latest version of the liblo library from <http://plugin.org.uk/liblo>. On Windows, liblo requires a Windows version of the POSIX thread library (pthreads) which is available from <http://sourceware.org/pthreads-win32>; copy `libpthreadGC2.a` to `libpthread.a`. You may also need the latest version of autoconf from MinGW.
- The STK opcodes require STK source code from <http://ccrma.stanford.edu/software/stk>, copied into `csound5/Opcodes/stk`.
- The Loris opcodes require Loris source code from <http://sourceforge.net/projects/loris>, copied into `csound5/Opcodes/Loris`.

Windows

The following is needed to build on Windows (more complete build instructions for Windows may be found in the `csound-build.tex` document (`csound-build.pdf`)):

- Install a compiler like gcc or Microsoft Visual Studio (there is also support for the Intel C++ compiler). If using MinGW (gcc), install all of the current release of MinGW using the Automated MinGW Installer from www.mingw.org [<http://www.mingw.org>], for example into `c:/mingw`. This should install gcc, g++, GNU binutils, the MinGW runtime, and the win32 API. Then install the current release of MSys.

On Windows you can use Microsoft Visual C++ (except for *CsoundAC*). The free Express Edition, from <http://www.microsoft.com/express/vc/> works fine. You will need to obtain a copy of the `direct.h` header file for Windows, e.g. from <http://www.softgalleria.net/direct.php>. You may also need to obtain the `bufferoverflowu.lib` library from Microsoft and put it into the Visual C++ `lib` directory. Then open a shell in which to compile Csound, (usually called Visual Studio Command Prompt command, within the Visual C++ program menu).

Optional configurations for Windows include the following:

- Real-time audio and MIDI can use the Windows multimedia library. This module will be built automatically if the headers are found.
- The VST Host opcodes require both the Steinberg VST headers.

Linux

Optional configurations for Linux include the following:

- Real-time audio on Linux can use ALSA (www.alsa-project.org [<http://www.alsa-project.org>]) and JACK (www.jackaudio.org/ [<http://www.jackaudio.org/>]) in addition to PortAudio. Distributions usually provide the appropriate dev packages for these systems through their repositories.
- The DSSI Host opcodes require both the LADSPA and DSSI headers.

Mac OS X

Optional configurations for Mac OS X include the following:

- Real-time audio can use CoreAudio (OSX builtin native audio system) and Jack, apart from PortAudio.
- The DSSI Host opcodes require both the LADSPA and DSSI headers.

Building Csound 5 with SCons

When you have all the necessary packages and their sources (or -dev packages) to support your particular requirements on your hardware platform, execute "scons -h" to discover the current configuration options.

Building is made considerably easier if, when installing, the downloaded headers and libraries are installed in their default locations. To modify the default build, in particular to handle non-standard options for third-party dependencies, such as where headers and libraries are to be found:

- On Windows, when building with Microsoft Visual C++, modify custom-msvc.py.
- On Windows, when building with MinGW/MSys, modify custom-mingw.py
- On Linux and Mac OSX edit custom.py

If you do modify this file, you should make it read-only (or otherwise protect it) so that CVS will not overwrite it when you next update the sourcefiles. Avoid modifying the SConstruct file.

Execute scons with the optional custom variables you desire. For example:

```
scons buildOSC=1 buildCsound5GUI=1 buildPythonOpcodes=1 useOSC=1 buildLoris=0
```



Note

It is important that you set the environment variable `OPCODEDIR` to the directory where plugin libraries are installed; in the case of a double precision build, `OPCODEDIR64` should be set instead. Installers usually take care of this, but it is necessary when building from source so Csound can find its plugin libraries.

Current build options

Table 4. Current SCons Build Options

Custom Variable	Effect if set to 1
buildCsoundVST	Build CsoundVST. Needs CsoundAC, FLTK,

Custom Variable	Effect if set to 1
	boost, Python, SWIG.
buildCsoundAC	Build CsoundAC. Needs FLTK, boost, Python, SWIG.
buildCsound5GUI	Build FLTK GUI frontend. Requires FLTK 1.1.7 or later.
buildCSEditor	Build the Csound syntax highlighting text editor. Requires FLTK headers and libs.
buildDSSI	Build DSSI/LADSPA host opcodes.
buildImageOpcodes	Build image opcodes. 1 by default. Set to 0 to avoid.
buildInterfaces	Build interface library for Python, JAVA, Lua, C++, and other languages.
buildJavaWrapper	Build Java wrapper for the interface library.
buildLoris	Build the Loris Python extension and opcodes.
buildNewParser	Enable building new parser. Requires Flex/Bison.
buildOSXGUI	Build the basic GUI frontend. OSX only.
buildPDCClass	Build csoundapi~ PD class. Needs m_pd.h in the standard places.
buildPythonOpcodes	Build Python opcodes
buildRelease	Build for release. Implies noDebug.
buildSDFT	Build SDFT code. 1 by default. Set to 0 to avoid.
buildTclcsound	Build Tclcsound frontend (cstclsh, cswish and telcsound dynamic module). Requires Tcl/Tk headers and libs.
buildUtilities	Build stand-alone executables for utilities that can also be used with -U.
buildVirtual	Build Virtual MIDI keyboard. Requires FLTK 1.1.7 or later headers and libs.
buildvst4cs	Build vst4cs plugins. Requires Steinberg VST headers.
buildWinsound	Build Winsound frontend. Requires FLTK headers and libs.
dynamicCsoundLibrary	Build dynamic Csound library instead of libcsound.a.
gcc3opt	Enable gcc 3.3.x or later optimizations for the specified CPU architecture (e.g. pentium3); implies noDebug.
gcc4opt	Enable gcc 4.0 or later optimizations for the specified CPU architecture (e.g. pentium3); implies noDebug.
generateTags	Generate TAGS.
generatePdf	Generate PDF documentation.
install	Enables the Install targets.
Lib64	Build for lib64 rather than lib.
noDebug	Build without debugging information.
noFLTKThreads	Disable use of a separate thread for FLTK widgets.
useAltiVec	On OSX use the gcc AltiVec optimisation flags.
useALSA	ALSA for real-time audio and MIDI input and

Custom Variable	Effect if set to 1
	output.
useCoreAudio	use CoreAudio for real-time audio input and output.
useDouble	Use double-precision floating point for audio samples.
useFLTK	Use FLTK for graphs and widget opcodes.
useGettext	Use the GNU internationalisation/localisation scheme
useGprof	Build with profiling information (-pg).
usePortAudio	Use PortAudio for real-time audio input and output.
usePortMIDI	Build PortMidi plugin for real time MIDI input and output.
useJack	Used if you compiled PortAudio to use Jack; also builds Jack plugin.
useLrint	Use lrint() and lrintf() for converting floating point values to integers.
useOSC	For OSC support.
useUDP	For UDP support. 1 by default. Set to 0 to avoid.
withICL	Build with the Intel C++ Compiler (also requires Microsoft Visual C++), Set to 0 to build with MinGW. Windows only.
withMSVC	Build with Microsoft Visual C++, or set to 0 to build with MinGW. Windows only.
Word64	Build for 64bit computer.
pythonVersion	Set to the Python version to be used.

Csound Links

Csound's "home page" is maintained by Richard Boulanger at <http://csounds.com>.

The Csound source code is maintained by John ffitch and others at <http://www.sourceforge.net/projects/csound>. The most recent versions and precompiled packages for most platforms also can be downloaded here [http://sourceforge.net/project/showfiles.php?group_id=81968].

A Csound mailing list exists to discuss Csound. It is run by John ffitch of Bath University, UK. To have your name put on the mailing list send an empty message to: csound-subscribe@lists.bath.ac.uk [<mailto:csound-subscribe@lists.bath.ac.uk>]. You can also subscribe to the digest (1 message per day) by sending an empty email to: csound-digest-subscribe@lists.bath.ac.uk [<mailto:csound-digest-subscribe@lists.bath.ac.uk>]. Posts sent to csound@lists.bath.ac.uk [<mailto:csound@lists.bath.ac.uk>] go to all subscribed members of the list. You can browse the csound mailing list archives here [http://agencities.cs.bath.ac.uk/%7ebwillkie/list_arch.php]

Similarly, the Csound-devel mailing list exists to discuss Csound development. For more information on this list, go to <http://lists.sourceforge.net/lists/listinfo/csound-devel>. Posts sent to csound-devel@lists.sourceforge.net [<mailto:csound-devel@lists.sourceforge.net>] go to all subscribed members of the list.

Part II. Opcodes Overview

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Signal Generators

Additive Synthesis/Resynthesis

The opcodes for additive synthesis and resynthesis are:

- *adsyn*
- *adsynt*
- *adsynt2*
- *hsboscil*

See the section *Spectral processing* for more information and further additive/resynthesis opcodes.

Basic Oscillators

The basic oscillator opcodes are: (note that opcodes that end with 'i' implement linear interpolation and those that end with '3' implement cubic interpolation)

- Oscillator Banks: *oscbnk*
- Simple table oscillators: *oscil*, *oscil3* and *oscili*.
- Simple, fast sine oscillator: *oscils*
- Precision oscillators: *poscil* and *poscil3*.
- More flexible oscillators: *oscilikt*, *osciliktp*, *oscilikts* and *osciln* (also called *oscilx*).

Oscillators can also be constructed from generic table read opcodes. See the *Table Read/Write operations* section.

LFOs

- *lfo*
- *vibr*
- *vibrato*

See the section *Table access* for other table reading opcodes that can be used as oscillators. Also see the section *Dynamic spectrum Oscillators*.

Dynamic Spectrum Oscillators

The opcodes that generate dynamic spectra are:

- Harmonic spectra: *buzz* and *gbuzz*
- Impulse generator: *mpulse*

- Band limited oscillators (analog modelled): *vco* and *vco2*

The following opcodes can be used to generate band-limited waveforms for use with *vco2* and other oscillators:

- *vco2init*
- *vco2ft*
- *vco2ift*

FM Synthesis

The FM synthesis opcodes are:

- *foscil*
- *foscili*

FM instrument models

- *fmb3*
- *fmbell*
- *fmmetal*
- *fmpercfl*
- *fmrhode*
- *fmvoice*
- *fmwurlie*

Granular Synthesis

The granular synthesis opcodes are:

- *diskgrain*
- *fof*
- *fof2*
- *fog*
- *grain*
- *grain2*
- *grain3*
- *granule*

- *partikkel*
- *partikkelsync*
- *sndwarp*
- *sndwarpst*
- *syncgrain*
- *syncloop*
- *vosim*

Hyper Vectorial Synthesis

- *vphaseseg*
- *hvs1*
- *hvs2*
- *hvs3*

Linear and Exponential Generators

The opcodes that generate linear or exponential curves or segments are:

- *expon*
- *expcurve*
- *expseg*
- *expsega*
- *expsegr*
- *gainslider*
- *jspline*
- *line*
- *linseg*
- *linsegr*
- *logcurve*
- *loopseg*
- *loopsegp*
- *lpshold*

- *lpsholdp*
- *rspline*
- *scale*
- *transeg*

Envelope Generators

The following envelope generators are available:

- *adsr*
- *madsr*
- *mxadsr*
- *xadsr*
- *linen*
- *linenr*
- *envlpx*
- *envlpxr*

Consult the *Linear and exponential generators* section for additional methods to create envelopes.

Models and Emulations

The following opcodes model or emulate the sounds of other instruments (some based on the STK toolkit by Perry Cook):

- *bamboo*
- *barmodel*
- *cabasa*
- *crunch*
- *dripwater*
- *gogobel*
- *guiro*
- *mandol*
- *marimba*
- *moog*
- *sandpaper*
- *sekere*
- *shaker*

- *sleighbells*
- *stix*
- *tambourine*
- *vibes*
- *voice*

Other models and emulations

- *lorenz*
- *planet*
- *prepiano*
- Fractal Number (Mandelbrot set) generator: *mandel*
- *chuap*

Phasors

The opcodes that generate a moving phase value:

- *phasor*
- *phasorbnk*
- *syncphasor*

These opcodes are useful in combination with the *Table access* opcodes.

Random (Noise) Generators

Opcodes that generate random numbers are:

- *betarnd*
- *bexprnd*
- *cauchy*
- *cuserrnd*
- *duserrnd*
- *exprand*
- *gauss*
- *linrand*
- *noise*
- *pcauchy*

- *pinkish*
- *poisson*
- *rand*
- *randh*
- *randi*
- *rnd31*
- *random*
- *randomh*
- *randomi*
- *trirand*
- *unirand*
- *urd*
- *weibull*
- *jitter*
- *jitter2*
- *trandom*

See *seed* which sets the global seed value for all x-class noise generators, as well as other opcodes that use a random call, such as *grain*. *rand*, *randh*, *randi*, *rnd(x)* and *birnd(x)* are not affected by *seed*.

See also functions which generate random numbers in the section *Random Functions*.

Sample Playback

Opcodes that implement sample playback and looping are:

- *bbcutm*
- *bbcuts*
- *flooper*
- *flooper2*
- *loscil*
- *loscil3*
- *loscilx*
- *lphasor*
- *lposcil*
- *lposcil3*

- *lposcila*
- *lposcilsa*
- *lposcilsa2*
- *sndloop*
- *waveset*

See also the *Signal Input* section for other ways to input sound.

Soundfonts

Fluid Opcodes

The fluid family of opcodes wraps Peter Hannape's SoundFont 2 player, FluidSynth: *fluidEngine* for instantiating a FluidSynth engine, *fluidSetInterpMethod* for setting interpolation method for a channel in a FluidSynth engine, *fluidLoad* for loading SoundFonts, *fluidProgramSelect* for assigning presets from a SoundFont to a FluidSynth engine's MIDI channel, *fluidNote* for playing a note on a FluidSynth engine's MIDI channel, *fluidCCi* for sending a controller message at i-time to a FluidSynth engine's MIDI channel, *fluidCCk* for sending a controller message at k-rate to a FluidSynth engine's MIDI channel, *fluidControl* for playing and controlling loaded Soundfonts (using 'raw' MIDI messages), *fluidOut* for receiving audio from a single FluidSynth engine, and *fluidAllOut* for receiving audio from all FluidSynth engines.

- *fluidAllOut*
- *fluidCCi*
- *fluidCCk*
- *fluidControl*
- *fluidEngine*
- *fluidLoad*
- *fluidNote*
- *fluidOut*
- *fluidProgramSelect*
- *fluidSetInterpMethod*

Old Soundfont opcodes

These opcodes can also use soundfonts to generate sound. The usage of the fluid Opcodes (above) is recommended instead of these opcodes.

- *sfilist*
- *sfinstr*
- *sfinstr3*
- *sfinstr3m*
- *sfinstrm*

- *sfload*
- *sfpassign*
- *sfplay*
- *sfplay3*
- *sfplay3m*
- *sfplaym*
- *sflooper*
- *sfplist*
- *sfpreset*

Scanned Synthesis

Scanned synthesis is a variant of physical modeling, where a network of masses connected by springs is used to generate a dynamic waveform. The opcode *scanu* defines the mass/spring network and sets it in motion. The opcode *scans* follows a predefined path (trajectory) around the network and outputs the detected waveform. Several *scans* instances may follow different paths around the same network.

These are highly efficient mechanical modelling algorithms for both synthesis and sonic animation via algorithmic processing. They should run in real-time. Thus, the output is useful either directly as audio, or as controller values for other parameters.

The Csound implementation adds support for a scanning path or matrix. Essentially, this offers the possibility of reconnecting the masses in different orders, causing the signal to propagate quite differently. They do not necessarily need to be connected to their direct neighbors. Essentially, the matrix has the effect of “molding” this surface into a radically different shape.

To produce the matrices, the table format is straightforward. For example, for 4 masses we have the following grid describing the possible connections:

	1	2	3	4
1				
2				
3				
4				

Whenever two masses are connected, the point they define is 1. If two masses are not connected, then the point they define is 0. For example, a unidirectional string has the following connections: (1,2), (2,3), (3,4). If it is bidirectional, it also has (2,1), (3,2), (4,3)). For the unidirectional string, the matrix appears:

	1	2	3	4
1	0	1	0	0
2	0	0	1	0
3	0	0	0	1
4	0	0	0	0

The above table format of the connection matrix is for conceptual convenience only. The actual val-

ues shown in te table are obtained by *scans* from an ASCII file using *GEN23*. The actual ASCII file is created from the table model row by row. Therefore the ASCII file for the example table shown above becomes:

0100001000010000

This matrix example is very small and simple. In practice, most scanned synthesis instruments will use many more masses than four, so their matrices will be much larger and more complex. See the example in the *scans* documentation.

Please note that the generated dynamic wavetables are very unstable. Certain values for masses, centering, and damping can cause the system to “blow up” and the most interesting sounds to emerge from your loudspeakers!

The supplement to this manual contains a tutorial on scanned synthesis. The tutorial, examples, and other information on scanned synthesis is available from the Scanned Synthesis page at cSounds.com.

Scanned synthesis developed by Bill Verplank, Max Mathews and Rob Shaw at Interval Research between 1998 and 2000.

Opcodes that implement scanned synthesis are:

- *scanhammer*
- *scans*
- *scantable*
- *scanu*
- *xscanmap*
- *xscans*
- *xscansmap*
- *xscanu*

Table Access

The opcodes that access tables are:

- *oscil1*
- *oscil1i*
- *osciln*
- *oscilx*
- *table*
- *table3*
- *tablei*

Opcodes ending in 'i' implement linear interpolation and opcodes ending in '3' implement cubic interpolation.

The following opcodes implement fast table reading/writing without boundary checks:

- *tab*
- *tab_i*
- *tabw*
- *tabw_i*

See the sections *Table Queries*, *Read/Write Operations* and *Table Reading with Dynamic Selection* for other table operations.



Note

Although tables with a size which is not a power of two can be created using a negative size (see *fscore statement*), some opcodes will not accept them.

Wave Terrain Synthesis

The opcode that uses wave terrain synthesis is *wterrain*.

Waveguide Physical Modeling

The opcodes that implement waveguide physical modeling are:

- *pluck*
- *repluck*
- *wgbow*
- *wgbowedbar*
- *wgbrass*
- *wgclar*
- *wgflute*
- *wgpluck*
- *wgpluck2*
- *wguide1*
- *wguide2*

Signal Input and Output

File Input and Output

The opcodes for file input and output are:

- File open/close: *fiopen* and *ficlose*.
- File output: *dumpk*, *dumpk2*, *dumpk3*, *dumpk4*, *fout*, *fouti*, *foutir* and *foutk*
- File input: *readk*, *readk2*, *readk3*, *readk4*, *fin*, *fini* and *fink*
- Utilities for use with the *fout* opcodes: *clear*, *vincr*
- Printing to a file: *fprints* and *fprintks*

Signal Input

The opcodes that receive audio signals are:

- Synchronous input: *in*, *in32*, *inch*, *inh*, *ino*, *inq*, *inrg*, *ins* and *inx*
- File streaming: *diskin*, *diskin2* and *soundin*
- User defined channel input: *invalue*
- Streaming input: *soundin*
- Direct to zak input: *inz*

See the section *Software Bus* for input and output through the API.

Signal Output

The opcodes that write audio signals are:

- Synchronous output: *out*, *out32*, *outc*, *outch*, *outh*, *outo*, *outrg*, *outq*, *outq1*, *outq2*, *outq3*, *outq4*, *outs*, *outs1*, *outs2* and *outx*
- Streaming output: *soundout* and *soundouts*
- User defined channel output: *outvalue*
- Direct from zak output: *outz*

The opcode *monitor* can be used for monitoring the complete output of *csound* (the output spout frame).

See the section *Software Bus* for input and output through the API.

Software Bus

Csound implements a software bus for internal routing or routing to external software calling the

Csound API.

The opcodes to use the software bus are:

- *chn_k*
- *chn_a*
- *chn_S*
- *chnclear*
- *chnexport*
- *chnmix*
- *chnparams*

Printing and Display

Opcodes for printing and displaying values are:

- *dispfft*
- *display*
- *flashtxt*
- *print*
- *printf*
- *printf_i*
- *printk*
- *printk2*
- *printks*
- *prints*

Sound File Queries

The opcodes that query information about files are:

- *filelen*
- *filenchnls*
- *filepeak*
- *filesr*

Signal Modifiers

Amplitude Modifiers and Dynamic processing

The opcodes that modify amplitude are:

- *balance*
- *compress*
- *clip*
- *dam*
- *gain*

The opcode *Odbfs* facilitates the use of amplitude by removing the need to use of explicit sample values.

Convolution and Morphing

The opcodes that convolve and morph signals are:

- *convolve* also called *convle*
- *cross2*
- *dconv*
- *ftconv*
- *ftmorf*
- *pconvolve*

Delay

Fixed delays

- *delay*
- *delay1*
- *delayk*

Delay Lines

- *delayr*
- *delayw*

- *deltap*
- *deltap3*
- *deltapi*
- *deltapn*
- *deltapx*
- *deltapxw*

Variable delays

- *vdelay*
- *vdelay3*
- *vdelayx*
- *vdelayxs*
- *vdelayxq*
- *vdelayxw*
- *vdelayxwq*
- *vdelayxws*

Multitap delays

- *multitap*

Panning and Spatialization

Amplitude spatialization

- *locsend*
- *locsig*
- *pan*
- *pan2*
- *space*
- *spdist*
- *spsend*

3D spatialization with simulation of room acoustics

- *spat3d*
- *spat3di*
- *spat3dt*

Vector Base Amplitude Panning

- *vbap16*
- *vbap16move*
- *vbap4*
- *vbap4move*
- *vbap8*
- *vbap8move*
- *vbaplsinit*
- *vbapz*
- *vbapzmove*

Binaural spatialization

- *hrtfer*
- *hrtfmove*
- *hrtfmove2*
- *hrtfstat*

Ambisonics

- *bformdec*
- *bformenc*

Reverberation

The opcodes one can use for reverberation are:

- *alpass*
- *babo*
- *comb*
- *freeverb*

- *nestedap*
- *nreverb* (also called *reverb2*)
- *reverb*
- *reverbsc*
- *valpass*
- *vcomb*

Sample Level Operators

The opcodes one may use to modify signals are:

- $a(k)$
- *denorm*
- *diff*
- *downsamp*
- *fold*
- $i(k)$
- *integ*
- *interp*
- $i(k)$
- *ntrpol*
- *samphold*
- *upsamp*
- *vaget*
- *vaset*

Signal Limiters

Opcodes that can be used to limit signals are:

- *limit*
- *mirror*
- *wrap*

Special Effects

Opcodes that generate special effects are:

- *distort*
- *distort1*
- *flanger*
- *harmon*
- *phaser1*
- *phaser2*

Standard Filters

Resonant Low-pass filters

- *areson*
- *lowpass2*
- *lowres*
- *lowresx*
- *lpf18*
- *moogvcf*
- *moogladder*
- *reson*
- *resonr*
- *resonx*
- *resony*
- *resonz*
- *rezy*
- *statevar*
- *svfilter*
- *tbvcf*
- *vlowres*
- *bqrez*

Standard filters

- Hi-pass filters: *atone*, *atonex*
- Low-pass filters: *tone*, *tonex*

- Biquad filters: *biquad* and *biquada*.
- Butterworth filters: *butterbp*, *butterbr*, *butterhp*, *butterlp* (which are also called *butbp*, *butbr*, *buthp*, *butlp*)
- General filters: *clfilt*

Control signal filters

- *aresonk*
- *atonek*
- *lineto*
- *port*
- *portk*
- *resonk*
- *resonxk*
- *tlineto*
- *tonek*

Specialized Filters

High pass filters

- *dcblock*
- *dcblock2*

Parametric EQ

- *pareq*
- *rbjeq*
- *eqfil*

Other filters

- *nlfilt*
- *filter2*
- *fofilter*
- *hilbert*

- *zfilter2*

Waveguides

The opcodes that use waveguides to modify a signal are:

- *streson*
- *wguide1*
- *wguide2*

Waveshaping and Phase Distortion

These opcodes can perform dynamic waveshaping or phaseshaping (a.k.a. phase distortion). They differ from traditional table-based methods of waveshaping by directly calculating the transfer function with one or more variable parameters for affecting the amount or results of the shaping. Most of these opcodes could be used on either an audio signal (for waveshaping) or a phasor (for phaseshaping) but tend to work best for one of these applications.

These opcodes are good for waveshaping:

- *chebyshevpoly*
- *clip*
- *distort*
- *distort1*
- *polynomial*
- *powershape*

These opcodes are good for phaseshaping:

- *pdclip*
- *pdhalf*
- *pdhalfy*

Instrument Control

Clock Control

The opcodes to start and stop internal clocks are:

- *clockoff*
- *clockon*

These clocks count CPU time. There are 32 independent clocks available. You can use the opcode *readclock* to read current values of a clock. See *Time Reading* for other timing opcodes.

Conditional Values

The opcodes for conditional values are ==, >=, >, <, <=, and !=.

Duration Control Statements

The opcodes one can use to manipulate a note's duration are:

- *ihold*
- *turnoff*
- *turnoff2*
- *turnon*

For other realtime instrument control see *Real-time Performance Control* and *Instrument Invocation*.

FLTK Widgets and GUI controllers

Widgets allow the design of a custom Graphical User Interface (GUI) to control an orchestra in real-time. They are derived from the open-source library FLTK (Fast Light Tool Kit). This library is one of the fastest graphic libraries available, supports OpenGL and should be source compatible with different platforms (Windows, Linux, Unix and Mac OS). The subset of FLTK implemented in Csound provides the following types of objects:

Containers

FLTK Containers are widgets that contain other widgets such as panels, windows, etc. Csound provides the following container objects:

- Panels
- Scroll areas
- Pack
- Tabs
- Groups

Valuators	<p>The most useful objects are named <i>FLTK Valuators</i>. These objects allow the user to vary synthesis parameter values in real-time. Csound provides the following valuator objects:</p> <ul style="list-style-type: none">• Sliders• Knobs• Rollers• Text fields• Joysticks• Counters
Other widgets	<p>There are <i>other FLTK widgets</i> that are not valuators nor containers:</p> <ul style="list-style-type: none">• Buttons• Button banks• Labels• Keyboard and Mouse sensing

Also there are some other opcodes useful to modify the *widget appearance*:

- Updating widget value.
- Setting primary and selection colors of a widget.
- Setting font type, size and color of widgets.
- Resizing a widget.
- Hiding and showing a widget.

There are also these *general opcodes* that allow the following actions:

- Running the widget thread: *FLrun*
- Loading snapshots containing the status of all valuators of an orchestra: *FLgetsnap* and *FLloadsnap*.
- Saving snapshots containing the status of all valuators of an orchestra: *FLsavesnap* and *FLsetsnap*
- Setting the snapshot group of a declared valuator: *FLsetSnapGroup*

Below is a simple example of Csound code to create a window. Notice that all opcodes are init-rate and must be called only once per session. The best way to use them is to place them in the header section of an orchestra, before any instrument. Even though placing them inside an instrument is not prohibited, unpredictable results can occur if that instrument is called more than once.

Each container is made up of a couple of opcodes: the first indicating the start of the container block and the last indicating the end of that container block. Some container blocks can be nested but they must not be crossed. After defining all containers, a widget thread must be run by using the special *FLrun* opcode that takes no arguments.

```

<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o linseg.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>
;*****
sr=48000
kr=480
ksmps=100
nchnls=1

;*** It is recommended to put almost all GUI code in the
;*** header section of an orchestra

        FLpanel      "Panel1",450,550 ;***** start of container
; some widgets should be contained here
        FLpanelEnd    ;***** end of container

        FLrun         ;***** runs the widget thread, it is always required!
instr 1
;put some synthesis code here
endin
;*****
</CsInstruments>
<CsScore>
f 0 3600 ;dummy table for realtime input
e

</CsScore>
</CsoundSynthesizer>

```

The previous code simply creates a panel (an empty window because no widgets are defined inside the container).

The following example creates two panels and inserts a slider inside each of them:

```

<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o linseg.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>
;*****
sr=48000
kr=480
ksmps=100
nchnls=1

gk1,iha    FLpanel      "Panel1",450,550,100,100 ;***** start of container
           FLslider     "FLslider 1", 500, 1000, 0 ,1, -1, 300,15, 20,50
           FLpanelEnd    ;***** end of container

gk2,ihb    FLpanel      "Panel2",450,550,100,100 ;***** start of container
           FLslider     "FLslider 2", 100, 200, 0 ,1, -1, 300,15, 20,50
           FLpanelEnd    ;***** end of container

           FLrun         ;***** runs the widget thread, it is always required!

instr 1
; gk1 and gk2 variables that contain the output of valuator
; widgets previously defined, can be used inside any instrument
printk2 gk1
printk2 gk2 ;print the values of the valuator whenever they change
endin
;*****
</CsInstruments>
<CsScore>
f 0 3600 ;dummy table for realtime input
e

</CsScore>
</CsoundSynthesizer>

```

All widget opcodes are init-rate opcodes, even if valuator output k-rate variables. This happens because an independent thread is run based on a callback mechanism. It consumes very few processing resources since there is no need of polling. (This differs from other MIDI based controller opcodes.) So you can use any number of windows and valuator without degrading the real-time performance.

FLTK Containers

The opcodes for FLTK containers are:

- *FLgroup*
- *FLgroupEnd*
- *FLpack*
- *FLpackEnd*
- *FLpanel*
- *FLpanelEnd*
- *FLscroll*
- *FLscrollEnd*
- *FLtabs*
- *FLtabsEnd*

FLTK Valuator

The opcodes for FLTK valuator are:

- *FLcount*
- *FLjoy*
- *FLknob*
- *FLroller*
- *FLslider*
- *FLtext*

Other FLTK Widgets

Other FLTK widget opcodes are:

- *FLbox*
- *FLbutBank*
- *FLbutton*
- *FLkeyIn*
- *FLhvsBox*

- *FLhvsBoxSetValue*
- *FLmouse*
- *FLprintk*
- *FLprintk2*
- *FLprintk2*
- *FLslidBnk*
- *FLslidBnk2*
- *FLslidBnkGetHandle*
- *FLslidBnkSet*
- *FLslidBnk2Set*
- *FLslidBnk2Setk*
- *FLvalue*
- *FLvslidBnk*
- *FLvslidBnk2*
- *FLxyin*

Modifying FLTK Widget Appearance

The following opcodes modify FLTK widget appearance:

- *FLcolor*
- *FLcolor2*
- *FLhide*
- *FLlabel*
- *FLsetAlign*
- *FLsetBox*
- *FLsetColor*
- *FLsetColor2*
- *FLsetFont*
- *FLsetPosition*
- *FLsetSize*
- *FLsetText*
- *FLsetTextColor*
- *FLsetTextSize*

- *FLsetTextType*
- *FLsetVal_i*
- *FLsetVal*
- *FLshow*

General FLTK Widget-related Opcodes

The general FLTK widget-related opcodes are:

- *FLgetsnap*
- *FLloadsnap*
- *FLrun*
- *FLsavesnap*
- *FLsetsnap*
- *FLupdate*
- *FLsetSnapGroup*

Instrument Invocation

The opcodes one can use to create score events from within an orchestra are:

- *event*
- *event_i*
- *scoreline_i*
- *scoreline*
- *schedule*
- *schedwhen*
- *schedkwhen*
- *schedkwhennamed*

The *mute* opcode can be used to mute/unmute instruments during a performance.

Instruments definitions can be removed using the *remove* opcode.

Program Flow Control

The opcodes to manipulate which orchestra statements are executed are:

- *cgoto*

- *cigoto*
- *ckgoto*
- *cngoto*
- *elseif*
- *else*
- *endif*
- *goto*
- *if*
- *igoto*
- *kgoto*
- *tigoto*
- *timeout*

Opcodes to create looping constructions are:

- *loop_ge*
- *loop_gt*
- *loop_le*
- *loop_lt*



Warning

Some of these opcodes work at i-rate even if they contain k- or a- rate comparisons. See the *Reinitialization* section.

Real-time Performance Control

Opcodes that monitor and control real-time performance are:

- *active*
- *cpuprc*
- *maxalloc*
- *prealloc*
- *jacktransport*

The running csound process can be terminated using *exitnow*.

Initialization and Reinitialization

Opcodes used for the initialization of variables:

- *init*
- *tival*
- *=*
- *pset*

The opcodes that can generate another initialization pass are:

- *reinit*
- *rigoto*
- *rireturn*

The opcode *p* can be used to find score p-fields at i- or k-rate.

nstrnum returns the instrument number for a named instrument.

Sensing and Control

TCL/TK widgets

- *button*
- *checkbox*
- *control*
- *setctrl*

Keyboard and mouse sensing

- *sensekey* (also called *sense*)
- *xyin*

Envelope followers

- *follow*
- *follow2*
- *peak*
- *rms*

Tempo and Pitch estimation

- *ptrack*
- *pitch*
- *pitchamdf*
- *tempest*

Tempo and Sequencing

- *tempo*
- *miditempo*
- *tempoval*
- *seqtime*
- *seqtime2*
- *trigger*
- *trigseq*
- *timedseq*
- *changed*

System

- *getcfig*

Score control

- *rewindscore*
- *setscorepos*

Stacks

Csound implements a global stack that can be accessed with the following opcodes:

- *stack*
- *pop*

- *push*
- *pop_f*
- *push_f*

Sub-instrument Control

These opcodes let one define and use a sub-instrument:

- *subinstr*
- *subinstrinit*

See also the UDO and *Orchestra Macros* Macros section for similar functionality.

Time Reading

Opcodes one can use to read time values are:

- *readclock*
- *rtclock*
- *timeinstk*
- *timeinsts*
- *times*
- *timek*

You can obtain the system date using:

- *date* - Returns the number seconds since 1 January 1970.
- *dates* - Returns as a string the date and time specified.

You can also set up counters using *clockoff* and *clockon*.

Function Table Control

Refer to the *f score statement*, *ftgen*, *figentmp* and the *GEN Routines* section for information on creating tables.

Tables can be removed from memory using the *ftfree* opcode.

For information on table access, consult the section *Table Access*.

Tables for use with the *loscilx* opcode can be loaded using *sndload*.

Tables by default, require a size which is a power of two. However tables with any size can be generated by specifying the size as a negative number (see *f score statement*).



Note

Not all opcodes accept tables whose size is not a power of two, as this may be a requirement for internal processing.

Table Queries

Opcodes the query tables for information are:

- For tables loaded from a sound file (using *GEN01*): *ftchnls*, *ftlen*, *ftlptim* and *ftsr*
- For any table: *nsamp*, *ftlen*, *tbleng*

Read/Write Operations

Opcodes that read and write to a table are:

- *ftloadk*
- *ftload*
- *ftsavek*
- *ftsave*
- *tablecopy*
- *tablegpw*
- *tableicopy*
- *tableigpw*
- *tableimix*
- *tableiw*
- *tablemix*
- *tablera*
- *tablew*

- *tablewa*
- *tablewkt*
- *tabmorph*
- *tabmorpha*
- *tabmorphak*
- *tabmorphi*
- *tabrec*
- *tabplay*
- *ftmorf*

Table values can be accessed within expressions using the *tb* family of opcodes.

Many oscillators are in fact specialized table readers. See the *Basic oscillators* section.

Table Reading with Dynamic Selection

Opcodes that let one dynamically (at k-rate) select tables are:

- *tableikt*
- *tablekt*
- *tablexkt*

Mathematical Operations

Amplitude Converters

Opcodes to convert between different amplitude measurements are:

- *ampdb*
- *ampdbfs*
- *db*
- *dbamp*
- *dbfsamp*

Use *rms* to find the rms value of a signal. See also *0dbfs* for another way to handle amplitudes in *csound*.

Arithmetic and Logic Operations

Opcodes that perform arithmetic and logic operations are -, +, &&, //, *, /, ^, and %.

See the *Conditional Values* section and the *if* family of opcodes for usage of logical operators.

Comparators and Accumulators

The following opcodes perform comparisons between signals at a-rate or k-rate, find maxima or minima, or accumulate the results of several computations or comparisons:

- *max*
- *max_k*
- *maxabs*
- *maxabsaccum*
- *maxaccum*
- *min*
- *minabs*
- *minabsaccum*
- *minaccum*
- *vincr*
- *clear*

Mathematical Functions

Opcodes that perform mathematical functions are:

- *abs*
- *ceil*
- *exp*
- *floor*
- *frac*
- *int*
- *log*
- *log10*
- *logbtwo*
- *pow*
- *powershape*
- *powoftwo*
- *round*
- *sqrt*

Opcode Equivalents of Functions

Opcodes that perform the equivalent of mathematical functions are:

- *chebyshevpoly*
- *divz*
- *mac*
- *maca*
- *polynomial*
- *pow*
- *product*
- *sum*
- *taninv2*

Random Functions

Opcodes that perform random functions are:

- *birnd*
- *rnd*

See the section *Random (Noise) Generators* for opcodes that generate random signals.

Trigonometric Functions

Opcodes that perform trigonometric functions are:

- *cos*, *cosh* and *cosinv*
- *sin*, *sinh* and *sininv*
- *tan*, *tanh*, *taninv*, and *taninv2*.

Linear Algebra Opcodes

Linear Algebra Opcodes — Scalar, vector, and matrix arithmetic on real and complex values.

Description

These opcodes implement many linear algebra operations, from scalar, vector, and matrix arithmetic up to and including QR based eigenvalue decompositions. The opcodes are designed for digital signal processing, and of course other mathematical operations, in the Csound orchestra language.

The numerical implementation uses the gmm++ library from home.gna.org/getfem/gmm_intro [http://home.gna.org/getfem/gmm_intro].



Warning

For applications with f-sig variables, array arithmetic must be performed only when the f-sig is "current," because f-rate is some fraction of k-rate; currency can be determined with the `la_k_current_f` opcode.

For applications using assignments between real vectors and a-rate variables, array arithmetic must be performed only when the vectors are "current", because the size of the vector may be some integral multiple of ksmpls; currency can be determined by means of the `la_k_current_vr` opcode.

Table 5. Linear Algebra Data Types

Mathematical Type	Code	Corresponding Csound Type or Types
real scalar	r	i-rate or k-rate variable
complex scalar	c	pair of i-rate or k-rate variables, e.g. "kr, ki"
real vector	vr	i-rate variable holding address of array
real vector	a	a-rate variable
real vector	t	function table number
complex vector	vc	i-rate variable holding address of array
complex vector	f	fsig variable
real matrix	mr	i-rate variable holding address of array
complex matrix	mc	i-rate variable holding address of array

All arrays are 0-based; the first index iterates rows to give columns, the second index iterates columns to give elements.

All arrays are general and dense; banded, Hermitian, symmetric and sparse routines are not implemented.

An array can be of type code vr, vc, mr, or mc and is stored in an i-rate object. In orchestra code, an array is passed as a MYFLT i-rate variable that contains the address of the array object, which is actually stored in the allocator opcode instance. Although array variables are i-rate, of course their val-

ues and even shapes may change at i-rate or k-rate.

All operands must be pre-allocated; except for the creation opcodes, no opcode ever allocates any arrays. This is true even if the array appears on the left-hand side of an opcode! However, some operations may reshape arrays to hold results.

Arrays are automatically deallocated when their instrument is deallocated.

Not only for more efficient performance, but also to make it easier to remember opcode names, the performance rate, output value types, operation names, and input value types are deterministically encoded into the opcode name:

1. "la" for "linear algebra opcode family".
2. "i" or "k" for performance rate.
3. Type code(s) (see above table) for output value(s), but only if the type is not implicit from the input values.
4. Operation name: common mathematical name (preferred) or abbreviation.
5. Type code(s) for input values, if not implicit.

For additional details, see the `gmm++` documentation at <http://download.gna.org/getfem/doc/gmmuser.pdf>.

Syntax

Array Creation

```
ivv                                la_i_vr_create    irows
```

Create a real vector with irows rows.

```
ivc                                la_i_vc_create    irows
```

Create a complex vector with irows rows.

```
imr                                la_i_mr_create    irows, icolumns [, odiagonal]
```

Create a real matrix with irows rows and icolumns columns, with an optional value on the diagonal.

```
imc                                la_i_mc_create    irows, icolumns [, odiagonal_r, odiagonal_i]
```

Create a complex matrix with irows rows and icolumns columns, with an optional complex value on the diagonal.

Array Introspection

```
irows                                la_i_size_vr    ivr
```

Return the number of rows in real vector ivr.

```
irows                                la_i_size_vc    ivc
```

Return the number of rows in complex vector ivc.

```
irows, icolumns                    la_i_size_mr    imr
```

Return the number of rows and columns in real matrix imr.

```
irows, icolumns      la_i_size_mc      imc
```

Return the number of rows and columns in complex matrix imc.

```
kfiscurrent          la_k_current_f      fsig
```

Return 1 if fsig is current, that is, if the value of fsig will change on the next kperiod.

```
kvriscurrent          la_k_current_vr      ivr
```

Return 1 if the real vector ivr is current, that is, if Csound's current audio sample frame stands at index 0 of the vector.

```
la_i_print_vr      ivr
```

Print the value of real vector ivr.

```
la_i_print_vc      ivc
```

Print the value of complex vector ivc.

```
la_i_print_mr      imr
```

Print the value of real matrix imr.

```
la_i_print_mc      imc
```

Print the value of complex matrix imc.

Array Assignment and Conversion

```
ivr      la_i_assign_vr      ivr
```

Assign the value of the real vector on the right-hand side to the real vector on the left-hand side, at i-rate.

```
ivr      la_k_assign_vr      ivr
```

Assign the value of the real vector on the right-hand side to the real vector on the left-hand side, at k-rate.

```
ivc      la_i_assign_vc      ivc
```

```
ivc      la_k_assign_vc      ivr
```

```
imr      la_i_assign_mr      imr
```

```
imr      la_k_assign_mr      imr
```

```
imc      la_i_assign_mc      imc
```

```
imc      la_k_assign_mc      imr
```



Warning

Assignments to vectors from tables or fsigs may resize the vectors.

Assignments to vectors from a-rate variables, or to a-rate variables from vectors, will

be performed incrementally, one chunk of ksmps elements per kperiod. Therefore, array arithmetic on such vectors should only be performed when the vectors are current, as determined by the la_k_current_vr opcode.

ivr	la_k_assign_a	asig
ivr	la_i_assign_t	itablenumber
ivr	la_k_assign_t	itablenumber
ivc	la_k_assign_f	fsig
asig	la_k_a_assign	ivr
itablenum	la_i_t_assign	ivr
itablenum	la_k_t_assign	ivr
fsig	la_k_f_assign	ivc

Fill Arrays with Random Elements

ivr	la_i_random_vr	[ifill_fraction]
ivr	la_k_random_vr	[kfill_fraction]
ivc	la_i_random_vc	[ifill_fraction]
ivc	la_k_random_vc	[kfill_fraction]
imr	la_i_random_mr	[ifill_fraction]
imr	la_k_random_mr	[kfill_fraction]
imc	la_i_random_mc	[ifill_fraction]
imc	la_k_random_mc	[kfill_fraction]

Array Element Access

ivr	la_i_vr_set	irow, ivalue
kvr	la_k_vr_set	krow, kvalue
ivc	la_i_vc_set	irow, ivalue_r, ivalue_i
kvc	la_k_vc_set	krow, kvalue_r, kvalue_i
imr	la_i_mr_set	irow, icolumn, ivalue
kmr	la_k_mr_set	krow, kcolumn, ivalue
imc	la_i_mc_set	irow, icolumn, ivalue_r, ivalue_i
kmc	la_k_mc_set	krow, kcolumn, kvalue_r, kvalue_i

ivalue	<code>la_i_get_vr</code>	ivr, irow
kvalue	<code>la_k_get_vr</code>	ivr, krow
ivalue_r, ivalue_i	<code>la_i_get_vc</code>	ivc, irow
kvalue_r, kvalue_i	<code>la_k_get_vc</code>	ivc, krow
ivalue	<code>la_i_get_mr</code>	imr, irow, icolumn
kvalue	<code>la_k_get_mr</code>	imr, krow, kcolumn
ivalue_r, ivalue_i	<code>la_i_get_mc</code>	imc, irow, icolumn
kvalue_r, kvalue_i	<code>la_k_get_mc</code>	imc, krow, kcolumn

Single Array Operations

imr	<code>la_i_transpose_mr</code>	imr
imr	<code>la_k_transpose_mr</code>	imr
imc	<code>la_i_transpose_mc</code>	imc
imc	<code>la_k_transpose_mc</code>	imc
ivr	<code>la_i_conjugate_vr</code>	ivr
ivr	<code>la_k_conjugate_vr</code>	ivr
ivc	<code>la_i_conjugate_vc</code>	ivc
ivc	<code>la_k_conjugate_vc</code>	ivc
imr	<code>la_i_conjugate_mr</code>	imr
imr	<code>la_k_conjugate_mr</code>	imr
imc	<code>la_i_conjugate_mc</code>	imc
imc	<code>la_k_conjugate_mc</code>	imc

Scalar Operations

ir	<code>la_i_norm1_vr</code>	ivr
kr	<code>la_k_norm1_vr</code>	ivc
ir	<code>la_i_norm1_vc</code>	ivc
kr	<code>la_k_norm1_vc</code>	ivc
ir	<code>la_i_norm1_mr</code>	imr
kr	<code>la_k_norm1_mr</code>	imr
ir	<code>la_i_norm1_mc</code>	imc

kr	la_k_norm1_mc	imc
ir	la_i_norm_euclid_vr	ivr
kr	la_k_norm_euclid_vr	ivr
ir	la_i_norm_euclid_vc	ivc
kr	la_k_norm_euclid_vc	ivc
ir	la_i_norm_euclid_mr	mvr
kr	la_k_norm_euclid_mr	mvr
ir	la_i_norm_euclid_mc	mvc
kr	la_k_norm_euclid_mc	mvc
ir	la_i_distance_vr	ivr
kr	la_k_distance_vr	ivr
ir	la_i_distance_vc	ivc
kr	la_k_distance_vc	ivc
ir	la_i_norm_max	imr
kr	la_k_norm_max	imc
ir	la_i_norm_max	imr
kr	la_k_norm_max	imc
ir	la_i_norm_inf_vr	ivr
kr	la_k_norm_inf_vr	ivr
ir	la_i_norm_inf_vc	ivc
kr	la_k_norm_inf_vc	ivc
ir	la_i_norm_inf_mr	imr
kr	la_k_norm_inf_mr	imr
ir	la_i_norm_inf_mc	imc
kr	la_k_norm_inf_mc	imc
ir	la_i_trace_mr	imr
kr	la_k_trace_mr	imr
ir, ii	la_i_trace_mc	imc
kr, ki	la_k_trace_mc	imc

ir	la_i_lu_det	imr
kr	la_k_lu_det	imr
ir	la_i_lu_det	imc
kr	la_k_lu_det	imc

Elementwise Array-Array Operations

ivr	la_i_add_vr	ivr_a, ivr_b
ivc	la_k_add_vc	ivc_a, ivc_b
imr	la_i_add_mr	imr_a, imr_b
imc	la_k_add_mc	imc_a, imc_b
ivr	la_i_subtract_vr	ivr_a, ivr_b
ivc	la_k_subtract_vc	ivc_a, ivc_b
imr	la_i_subtract_mr	imr_a, imr_b
imc	la_k_subtract_mc	imc_a, imc_b
ivr	la_i_multiply_vr	ivr_a, ivr_b
ivc	la_k_multiply_vc	ivc_a, ivc_b
imr	la_i_multiply_mr	imr_a, imr_b
imc	la_k_multiply_mc	imc_a, imc_b
ivr	la_i_divide_vr	ivr_a, ivr_b
ivc	la_k_divide_vc	ivc_a, ivc_b
imr	la_i_divide_mr	imr_a, imr_b
imc	la_k_divide_mc	imc_a, imc_b

Inner Products

ir	la_i_dot_vr	ivr_a, ivr_b
kr	la_k_dot_vr	ivr_a, ivr_b
ir, ii	la_i_dot_vc	ivc_a, ivc_b
kr, ki	la_k_dot_vc	ivc_a, ivc_b
imr	la_i_dot_mr	imr_a, imr_b
imr	la_k_dot_mr	imr_a, imr_b
imc	la_i_dot_mc	imc_a, imc_b

imc	la_k_dot_mc	imc_a, imc_b
ivr	la_i_dot_mr_vr	imr_a, ivr_b
ivr	la_k_dot_mr_vr	imr_a, ivr_b
ivc	la_i_dot_mc_vc	imc_a, ivc_b
ivc	la_k_dot_mc_vc	imc_a, ivc_b

Matrix Inversion

imr, icondition	la_i_invert_mr	imr
imr, kcondition	la_k_invert_mr	imr
imc, icondition	la_i_invert_mc	imc
imc, kcondition	la_k_invert_mc	imc

Matrix Decompositions and Solvers

ivr	la_i_upper_solve_mr	imr [, j_l_diagonal]
ivr	la_k_upper_solve_mr	imr [, j_l_diagonal]
ivc	la_i_upper_solve_mc	imc [, j_l_diagonal]
ivc	la_k_upper_solve_mc	imc [, j_l_diagonal]
ivr	la_i_lower_solve_mr	imr [, j_l_diagonal]
ivr	la_k_lower_solve_mr	imr [, j_l_diagonal]
ivc	la_i_lower_solve_mc	imc [, j_l_diagonal]
ivc	la_k_lower_solve_mc	imc [, j_l_diagonal]
imr, ivr_pivot, isize	la_i_lu_factor_mr	imr
imr, ivr_pivot, ksize	la_k_lu_factor_mr	imr
imc, ivr_pivot, isize	la_i_lu_factor_mc	imc
imc, ivr_pivot, ksize	la_k_lu_factor_mc	imc
ivr_x	la_i_lu_solve_mr	imr, ivr_b
ivr_x	la_k_lu_solve_mr	imr, ivr_b
ivc_x	la_i_lu_solve_mc	imc, ivc_b
ivc_x	la_k_lu_solve_mc	imc, ivc_b
imr_q, imr_r	la_i_qr_factor_mr	imr
imr_q, imr_r	la_k_qr_factor_mr	imr

<code>imc_q, imc_r</code>	<code>la_i_qr_factor_mc</code>	<code>imc</code>
<code>imc_q, imc_r</code>	<code>la_k_qr_factor_mc</code>	<code>imc</code>
<code>ivr_eig_vals</code>	<code>la_i_qr_eigen_mr</code>	<code>imr, i_tolerance</code>
<code>ivr_eig_vals</code>	<code>la_k_qr_eigen_mr</code>	<code>imr, k_tolerance</code>
<code>ivr_eig_vals</code>	<code>la_i_qr_eigen_mc</code>	<code>imc, i_tolerance</code>
<code>ivr_eig_vals</code>	<code>la_k_qr_eigen_mc</code>	<code>imc, k_tolerance</code>



Warning

Matrix must be Hermitian in order to compute eigenvectors.

<code>ivr_eig_vals, imr_eig_vecs</code>	<code>la_i_qr_sym_eigen_mr</code>	<code>imr, i_tolerance</code>
<code>ivr_eig_vals, imr_eig_vecs</code>	<code>la_k_qr_sym_eigen_mr</code>	<code>imr, k_tolerance</code>
<code>ivc_eig_vals, imc_eig_vecs</code>	<code>la_i_qr_sym_eigen_mc</code>	<code>imc, i_tolerance</code>
<code>ivc_eig_vals, imc_eig_vecs</code>	<code>la_k_qr_sym_eigen_mc</code>	<code>imc, k_tolerance</code>

Credits

Michael Gogins

New in Csound version 5.09

Pitch Converters

Functions

Opcodes that provide common pitch functions are:

- *cent*
- *cpsmidinn*
- *cpsoct*
- *cpspch*
- *octave*
- *octcps*
- *octmidinn*
- *octpch*
- *pchmidinn*
- *pchoct*
- *semitone*

Tuning Opcodes

Opcodes that provide tuning functions are:

- *cps2pch*
- *cpsxpch*
- *cpstun*
- *cpstuni*

Real-time MIDI Support

Csound supports realtime MIDI input and output, as well as input from MIDI files. Realtime MIDI input is activated using the `-M` (or `--midi-device=DEVICE`) command line flag. You must specify the device number or name after the `-M`. For example to use device number 2, you would use something like:

```
csound -M2 myrtmidi.csd
```

You can find out the available devices by using an out of range device:

```
csound -M99 myrtmidi.csd
```



Note

This will only work if the MIDI module can be accessed by device number. For alsa, you must first find the device name using:

```
cat /proc/asound/cards
```

You must then use something like:

```
csound --rtmidi=alsa -M hw:3 myrtmidi.csd
```

Realtime MIDI output is activated using `-Q`, using device number or names as shown above.

You can also load a MIDI file using the `-F` or `--midifile=FILE` command line flag. The MIDI file is read in realtime, and behaves as if it was being performed or recieved in realtime. So the csound program is not aware if MIDI input comes from a MIDI file or directly from a MIDI interface.

Once realtime MIDI input and/or output has been activated, opcodes like *MIDI Input* and *MIDI Output* will have effect.

When MIDI input is enabled (with `-M` or `-F`), each incoming *noteon* message will generate a note event for an instrument which has the same number as the channel of the event (see *massign* and *pg-massign* to change this behavior). This means that MIDI controlled instruments are polyphonic by default, since each note will generate a new instance of the instrument.

See the *MIDI/Score Interoperability* opcodes for information on designing instruments which can be used from the score or driven by MIDI.

There are several realtime MIDI modules available, you must use the `--rtmidi` flag (See `--rtmidi`), to specify the module. The default module is *portmidi* which provides adequate MIDI I/O on all platforms, however for improved performance and reliability some platform specific modules are also provided.

Currently the midi modules available are:

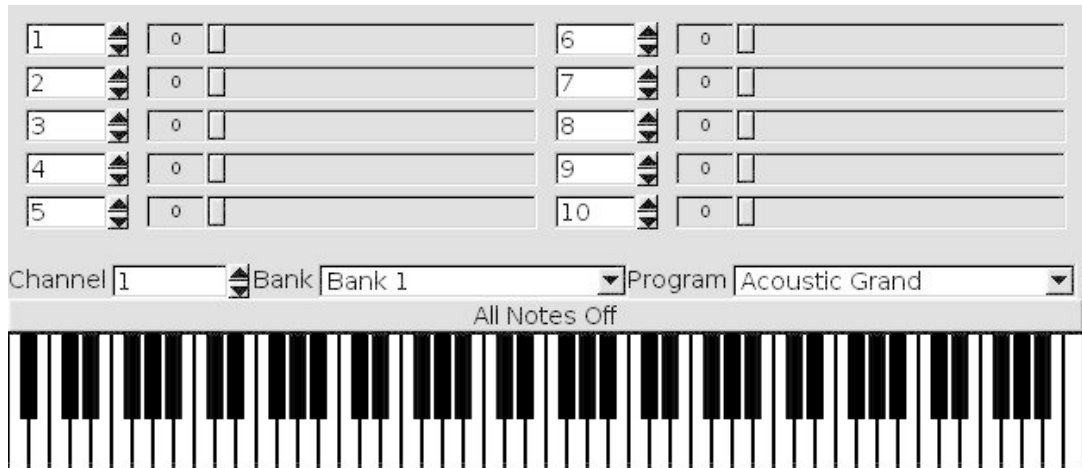
- *alsa* - To use the ALSA midi system (Linux only)
- *winmm* - To use the windows MME system (Windows only)
- *portmidi* - To use the portmidi system (all platforms). This is the default setting.
- *virtual* - To use a virtual graphical keyboard (See below) as MIDI input (all platforms)



Tip

When csound runs, it will process the score and then quit. If there are no events in the score, Csound will exit immediately. If you want to use only MIDI events instead of score events, you need to tell Csound to run for a certain amount of time. This can be done with a dummy *f-statement* like "f 0 3600".

Virtual MIDI Keyboard



Virtual MIDI keyboard.

The virtual MIDI keyboard module (activated using `--rtmidi=virtual` on the command line flags) provides a way of sending realtime MIDI information to Csound without the need of a MIDI device. It can send note information, control changes, bank and program changes on a specified channel. The MIDI information from the virtual keyboard is processed by Csound in exactly the same way as MIDI information that comes from the other MIDI drivers, so if your Csound orchestra is designed to work with hardware MIDI devices, this will also work.

For the device flag (`-M`), the virtual keyboard uses this to take in the name of a keyboard mapping files. Like all MIDI drivers, a device must be given to activate the driver. If you would like to just use the default settings of the keyboard, simply passing in 0 (i.e. `-M0`) and the virtual keyboard will use its default settings. If instead of the 0 a name of a file is given, the keyboard will attempt to load the file as a keyboard mapping. If the file could not be opened or read correctly, the default settings will be used.

Keyboard Mapping files allow the user to customize the name and number of banks as well as the name and number of programs per bank. The following example keyboard mapping (named `key-board.map`) has inline comments on the file format. This file is also available with the Csound source distribution in the `InOut/virtual_keyboard` folder.

```
# Custom Keyboard Map for Virtual Keyboard
# Steven Yi
#
# USAGE
#
# When using the Virtual Keyboard, you can supply a filename for a mapping
# of banks and programs via the -M flag, for example:
#
# csound --rtmidi=virtual -Mkeyboard.map my_project.csd
#
# INFORMATION ON THE FORMAT
#
# -lines that start with '#' are comments
# -lines that have [] start new bank definitions,
#   the contents are bankNum=bankName, with bankNum=[1,16384]
# -lines following bank statements are program definitions
#   in the format programNum=programName, with programNum=[1,128]
# -bankNumbers and programNumbers are defined in this file
```

```
# starting with 1, but are converted to midi values (starting
# with 0) when read
#
# NOTES
#
# -if an invalid bank definition is found, all program
# definitions that follow will be ignored until a new
# valid bank definition is found
# -if a valid bank is defined by no valid programs found
# for that bank, it will default to General MIDI program
# definitions
# -if an invalid program definition is found, it will be
# ignored

[1=My Bank]
1=My Test Patch 1
2=My Test Patch 2
30=My Test Patch 30

[2=My Bank2]
1=My Test Patch 1(bank2)
2=My Test Patch 2(bank2)
30=My Test Patch 30(bank3)
```

The ten sliders up top are by default set to MIDI Controller number 1-10 though they can be changed to whatever one wishes to use. The controller numbers and values of each slider are set per channel, so one may use different settings and values for each channel.

By default there are 128 banks and for each bank 128 patches defaulting to General Midi names. The MIDI bank standard uses 14-bit resolution to support 16384 possible banks, but the bank numbers by default are 0-127. To use values higher than 127, one should use a custom keyboard map and set the desired bank number value for the bank name. The virtual keyboard will correctly transmit the bank number as MSB and LSB with controller numbers 0 and 32.

Beyond the input available from interacting with the GUI via mouse, one may also trigger off MIDI notes by using the ASCII keyboard when the virtual keyboard window is focused. The layout is done much like a tracker and offers two octaves and a major third to trigger, starting from Middle-C (MIDI note 60). The ASCII keyboard MIDI note values are given in the following table.

Table 6. ASCII Keyboard MIDI Note Values

Keyboard Key	MIDI Value
z	60
s	61
x	62
d	63
c	64
v	65
g	66
b	67
h	68
n	69
j	70
m	71
q	72
2	73
w	74
3	75
e	76
r	77

Keyboard Key	MIDI Value
5	78
t	79
6	80
y	81
7	82
u	83
i	84
9	85
o	86
0	87
p	88

Here's an example of usage of the virtual MIDI keyboard. It uses the file *virtual.csd* [examples/virtual.csd].

```

<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in    Virtual MIDI    -M0 is needed anyway
-odac        -iadc      -+rtmidi=virtual -M0
</CsOptions>
<CsInstruments>
; By Mark Jamerson 2007

sr=44100
ksmps=10
nchnls=2

massign 1,1
prealloc 1,10

instr 1 ;Midi FM synth

inote cpsmidi
iveloc ampmidi 10000
idur = 2
      xtratism 1

kgate oscil 1,10,2
anoise noise 100*inote,.99
acps  samphold anoise,kgate
aosc oscili 1000,acps,1
aout = aosc

; Use controller 7 to control volume
kvol ctrl7 1, 7, 0.2, 1

outs kvol * aout, kvol * aout

endin

</CsInstruments>
<CsScore>
f0 3600
f1 0 1024 10 1
f2 0 16 7 1 8 0 8
f3 0 1024 10 1 .5 .6 .3 .2 .5

e
</CsScore>
</CsoundSynthesizer>

```

MIDI input

The following opcodes can receive MIDI information:

- MIDI information for any instruments: *aftouch*, *chanctrl* and *polyaft*, *pchbend*.
- MIDI information for MIDI-triggered instruments: *veloc*, *midictrl* and *notnum*. See also *Converters*.
- MIDI Controller input for any instrument: *midic7*, *midic14* and *midic21*.
- MIDI Controller input for MIDI-triggered instruments: *ctrl7*, *ctrl14* and *ctrl21*.
- MIDI controller value initialization: *initc7*, *initc14*, *initc21* and *ctrlinit*.

massign can be used to specify the csound instrument to be triggered by a particular MIDI channel. *pgmassign* can be use to assign a csound instrument to a specific MIDI program.

MIDI Message Output

Opcodes that produce MIDI output are:

- *mdelay*
- *nrpn*
- *outiat*
- *outic*
- *outic14*
- *outipat*
- *outipb*
- *outipc*
- *outkat*
- *outkc*
- *outkc14*
- *outkpat*
- *outkpb*
- *outkpc*

Generic Input and Output

Opcodes for generic MIDI input and output are *midiin* and *midiout*.

Converters

The following opcodes can convert MIDI information from a MIDI-triggered instrument instance:

- MIDI note number to frequency converters: *cpsmidi*, *cpsmidib*, *cpstmid*, *octmidi*, *octmidib*, *pchmidi* and *pchmidib*.

- MIDI velocity to amplitude converters: *ampmidi*.

Event Extenders

Opcodes that let one extend the duration of an event are:

- *release*
- *xtratim*

Note-on/Note-off Output

Opcodes to output MIDI note on or off messages are:

- *midion*
- *midion2*
- *moscil*
- *noteoff*
- *noteon*
- *noteondur*
- *noteondur2*

MIDI/Score Interoperability opcodes

The following opcodes can be used to design instruments that work interchangeably for real-time MIDI and score events:

- *midichannelaftertouch*
- *midichn*
- *midicontrolchange*
- *mididefault*
- *midinoteoff*
- *midinoteoncps*
- *midinoteonkey*
- *midinoteonoct*
- *midinoteonpch*
- *midipitchbend*
- *midipolyaftertouch*
- *midiprogramchange*.



Adapting a score-activated Csound instrument.

To adapt an ordinary Csound instrument designed for score activation for score/MIDI interoperability:

- Change all *linen*, *linseg*, and *expseg* opcodes to *linenr*, *linsegr*, and *expsegr*, respectively, except for a de-clicking or damping envelope. This will not materially change score-driven performance.
- Add the following lines at the beginning of the instrument definition:

```
; Ensures that a MIDI-activated instrument  
; will have a positive p3 field.  
mididefault 60, p3  
; Puts MIDI key translated to cycles per  
; second into p4, and MIDI velocity into p5  
midinoteoncps p4, p5
```

Obviously, *midinoteoncps* could be changed to *midinoteonoct* or any of the other options, and the choice of p-fields is arbitrary.



MIDI Realtime Input/Ouput command line options

New *MIDI I/O flags* in Csound 5.02, can replace most uses of these MIDI interop opcodes, and make usage easier.

System Realtime Messages

Opcodes for System Realtime MIDI messages are: *mclock* and *mrtmsg*.

Slider Banks

Opcodes for slider banks of MIDI controls are:

- *slider8*
- *slider8f*
- *slider16*
- *slider16f*
- *slider32*
- *slider32f*
- *slider64*
- *slider64f*
- *s16b14*
- *s32b14*
- *sliderKawai*

Opcodes for storing slider banks of MIDI controls to tables are:

- *slider8table*
- *slider8tablef*
- *slider16table*
- *slider16tablef*
- *slider32table*
- *slider32tablef*
- *slider64table*
- *slider64tablef*

Spectral Processing

See the section *Additive Synthesis/Resynthesis* for the basic resynthesis opcodes.

Short-time Fourier Transform (STFT) Resynthesis



Use of PVOC-EX files with the old Csound pvoc opcodes

All the original pvoc opcodes can now read a PVOC-EX file, as well as the native non-portable file format. As the PVOC-EX file uses a double-size analysis window, users may find that this gives a useful improvement in quality, for some sounds and processes, despite the fact that the resynthesis does not use the same window size.

Apart from the window size parameter, the main difference between the original .pv format and PVOC-EX is in the amplitude range of analysis frames. While rescaling is applied, so that no significant difference in output level is experienced, whichever file format is used, some slight loss of amplitude can still arise, as the double window usage itself modifies frame amplitudes, of which the resynthesis code is unaware. Note that all the original pvoc opcodes expect a mono analysis file, and multi-channel PVOC-EX files will accordingly be rejected.

Opcodes that implement STFT resynthesis are:

- *tableseg*
- *pvadd*
- *pvbufread*
- *pvcross*
- *pvinterp*
- *pvoc*
- *pvread*
- *tableseg*
- *tablexseg*
- *vpvoc*

Use the utility *PVANAL* to generate pv analysis files.

Linear Predictive Coding (LPC) Resynthesis

The linear predictive coding resynthesis opcodes are:

- *lpfreson*
- *lpinterp*
- *lpread*

- *lpreson*
- *lpslot*

LPC analysis files can be created using the *LPANAL* utility.

Non-standard Spectral Processing

These units generate and process non-standard signal data types, such as down-sampled time-domain control signals and audio signals, and their frequency-domain (spectral) representations. The data types (*d-*, *w-*) are self-defining, and the contents are not processable by any other Csound units. These unit generators are experimental, and subject to change between releases, they will also be joined by others later.

The opcodes for non-standard spectral processing are *specaddm*, *specdiff*, *specdisp*, *specfilt*, *spechist*, *specptrk*, *specscal*, *specsum*, and *spectrum*.

Tools for Real-time Spectral Processing (pvs opcodes)

With these opcodes, two new core facilities are added to Csound. They offer improved audio quality, and fast performance, enabling high-quality analysis and resynthesis (together with transformations) to be applied in real-time to live signals. The original Csound phase vocoder remains unaltered; the new opcodes use an entirely separate set of functions based on “pvoc.c” in the CARL distribution, written by Mark Dolson.

The Csound *dnoise* and *srconv* utilities (also by Dolson, from CARL) also use this pvoc engine. CARL pvoc is also the basis for the phase vocoder included in the Composer's Desktop Project. A few small but important modifications have been made to the original CARL code to support real-time streaming.

1. Support for the new PVOC-EX analysis file format. This is a fully portable (cross-platform) open file format, supporting three analysis formats, and multi-channel signals. Currently only the standard amplitude+frequency format has been implemented in the opcodes, but the file format itself supports amplitude+phase and complex (real-imaginary) formats. In addition to the new opcodes, the original Csound pvoc opcodes have been extended (and thereby with enhanced audio quality in some cases) to read PVOC-EX files as well as the original (non-portable) format.

Full details of the structure of a PVOC-EX file are available via the website: <http://www.cs.bath.ac.uk/~jpff/NOS-DREAM/researchdev/pvocex/pvocex.html>. This site also gives details of the freely available console programs pvocex and pvocex2 which can be used to create PVOC-EX files in all supported formats.

2. A new frequency-domain signal type, fully streamable, with *f* as the leading character. In this document it is conveniently referred to as an *fsig*. Primary support for fsigs is provided by the opcodes *pvsanal* and *pvsynth*, which perform conventional phase vocoder overlap-add analysis and resynthesis, independently of the orchestra control-rate. The only requirement is that the control-rate *kr* be higher than or equal to the analysis rate, which can be expressed by the requirement that $k_{\text{smpls}} \leq \text{overlap}$, where *overlap* is the distance in samples between analysis frames, as specified for *pvsanal*. As *overlap* is typically at least 128, and more usually 256, this is not an onerous restriction in practice. The opcode *pvsinfo* can be used at init time to acquire the properties of an *fsig*.

The *fsig* enables the nominal separation between the analysis and resynthesis stages of the phase vocoder to be exposed to the Csound programmer, so that not only can alternatives be employed for either or both of these stages (not only oscillator-bank resynthesis, but also the generation of synthetic *fsig* streams), but opcodes, operating on the *fsig* stream, can themselves become more

elemental. Thus the fsig enables the creation of a true streaming plugin framework for frequency domain signals. With the old pvoc opcodes, each opcode is required to act as a resynthesiser, so that facilities such as pitch scaling are duplicated in each opcode; and in many cases the opcodes are parameter-rich. The separation of analysis and synthesis stages by means of the fsig encourages the development of a wide range of simple building-block opcodes implementing one or two functions, with which more elaborate processes can be constructed.

This is very much a preliminary and experimental release, and it is possible that the precise definition of the opcodes may change, in response to user feedback. Also, clearly, many new possibilities for opcodes are opened up; these factors may also have a retrospective influence on the opcodes presented here.

Note that some opcode parameters currently have restricted or missing implementation. This is at least in part in order to keep the opcodes simple at this stage, and also because they highlight important design issues on which no decision has yet been made, and on which opinions from users are sought.

One important point about the new signal type is that because the analysis rate is typically much lower than kr, new analysis frames are not available on each k-cycle. Internally, the opcodes track ksmpts, and also maintain a frame counter, so that frames are read and written at the correct times; this process is generally transparent to the user. However, it means that k-rate signals only act on an fsig at the analysis rate, not at each k-cycle. The opcode pvsftw returns a k-rate flag that is set when new fsig data is valid.

Because of the nature of the overlap-add system, the use of these opcodes incurs a small but significant delay, or latency, determined by the window size ($\max(\text{ifftsize}, \text{iwinsize})$). This is typically around 23msecs. In this first release, the delay is slightly in excess of the theoretical minimum, and it is hoped that it can be reduced, as the opcodes are further optimized for real-time streaming.

The opcodes for real-time spectral processing are *pvsadsyn*, *pvsanal*, *pvsacross*, *pvsfread*, *pvsftr*, *pvsftw*, *pvsinfo*, *pvsmaska*, and *pvsynth*.

In addition there are a number of opcodes available as plugins in Csound5. These are *pvsdiskin*, *pvscent*, *pvsdemix*, *pvsfreeze*, *pvsbuffer*, *pvsbufread*, *pvscale*, *pvsshift*, *pvsifd*, *pvsinit*, *pvsin*, *pvsout*, *pvsosc*, *pvsbin*, *pvsdisp*, *pvsfwrite*, *pvmix*, *pvssmooth*, *pvsfilter*, *pvsblur*, *pvsstencil*, *pvsarp*, *pvsvoc*, *pvsrmorph*, *pvsbandp*, *pvsbandr*

A number of opcodes are designed to generate and process streaming partials tracks data. these are *partials*, *trcross*, *trfilter*, *trsplit*, *trmix*, *trscale*, *trshift*, *trlowest*, *trhighest*, *tradsyn*, *sinsyn*, *resyn*, *binit*

See the *Stacks* section for information on the stack opcodes which can stack f-signals.

ATS Spectral Processing

These opcodes can read, transform and resynthesize ATS analysis files. Please note that you need the ATS application to produce analysis files. From the ATS Reference Manual:

"ATS is a software library of functions for spectral Analysis, Transformation, and Synthesis of sound based on a sinusoidal plus critical-band noise model. A sound in ATS is a symbolic object representing a spectral model that can be sculpted using a variety of transformation functions."

For more information on ATS visit: <http://www-ccrma.stanford.edu/~juan/ATS.html>.

ATS analysis files can be produced using the ATS software or the csound utility *ATSA*.

The opcodes for ATS processing are:

- *ATSinfo*: reads data out of the header of an ATS file.
- *ATSread*, *ATSreadnz*, *ATSbufread*, *ATSinterpread*, *ATSpartialtap*: read data from an ATS file or

buffer.

- *ATSadd*, *ATSaddnz*, *ATScross*, *ATSSinnoi*: Resynthesize sound.

Credits

Author: Alex Norman
Seattle, Washington
2004

Loris Opcodes



Note

These opcodes are an optional component of Csound5. You can check if they are installed by using the command 'csound -z' which lists all available opcodes.

The Loris family of opcodes wraps: *lorisread* which imports a set of bandwidth-enhanced partials from a SDIF-format data file, applying control-rate frequency, amplitude, and bandwidth scaling envelopes, and stores the modified partials in memory; *lorismorph*, which morphs two stored sets of bandwidth-enhanced partials and stores a new set of partials representing the morphed sound. The morph is performed by linearly interpolating the parameter envelopes (frequency, amplitude, and bandwidth, or noisiness) of the bandwidth-enhanced partials according to control-rate frequency, amplitude, and bandwidth morphing functions, and *lorisplay*, which renders a stored set of bandwidth-enhanced partials using the method of Bandwidth-Enhanced Additive Synthesis implemented in the Loris software, applying control-rate frequency, amplitude, and bandwidth scaling envelopes.

For more information about sound morphing and manipulation using Loris and the Reassigned Bandwidth-Enhanced Additive Model, visit the Loris web site at www.cerlsoundgroup.org/Loris [<http://www.cerlsoundgroup.org/Loris>].

Examples

Example 3. Play the partials without modification

```
;
; Play the partials in clarinet.sdif
; from 0 to 3 sec with 1 ms fadetime
; and no frequency , amplitude, or
; bandwidth modification.
;
instr 1
  ktime      linseg      0, p3, 3.0      ; linear time function from 0 to 3 seconds
             lorisread   ktime, "clarinet.sdif", 1, 1, 1, 1, .001
  asig       lorisplay   1, 1, 1, 1
             out         asig
endin
```

Example 4. Add tuning and vibrato

```
; Play the partials in clarinet.sdif
; from 0 to 3 sec with 1 ms fadetime
; adding tuning and vibrato, increasing the
; "breathiness" (noisiness) and overall
; amplitude, and adding a highpass filter.
;
```

```

instr 2
  ktime    linseg    0, p3, 3.0    ; linear time function from 0 to 3 seconds
  ; compute frequency scale for tuning
  ; (original pitch was G#4)
  ifscale  =         cpspch(p4)/cpspch(8.08)

  ; make a vibrato envelope
  kvenv    linseg    0, p3/6, 0, p3/6, .02, p3/3, .02, p3/6, 0, p3/6, 0
  kvib     oscil      kvenv, 4, 1    ; table 1, sinusoid

  kbwenv   linseg    1, p3/6, 1, p3/6, 2, 2*p3/3, 2
  a1       lorisread  ktime, "clarinet.sdif", 1, 1, 1, 1, .001
  a2       lorisplay  1, ifscale+kvib, 2, kbwenv
  out      atone      a1, 1000      ; highpass filter, cutoff 1000 Hz
  out      a2

endin

```

The instrument in the first example synthesizes a clarinet tone from beginning to end using partials derived from reassigned bandwidth-enhanced analysis of a three-second clarinet tone, stored in a file, `clarinet.sdif`. The instrument in Example 2 adds tuning and vibrato to the clarinet tone synthesized by instr 1, boosts its amplitude and noisiness, and applies a highpass filter to the result. The following score can be used to test both of the instruments described above.

```

; make sinusoid in table 1
f 1 0 4096 10 1

; play instr 1
;   strt  dur
i 1 0 3
i 1 + 1
i 1 + 6
s

; play instr 2
;   strt  dur  ptch
i 2 1 3 8.08
i 2 3.5 1 8.04
i 2 4 6 8.00
i 2 4 6 8.07
e

```

Example 5. Morph partials

```

; Morph the partials in clarinet.sdif into the
; partials in flute.sdif over the duration of
; the sustained portion of the two tones (from
; .2 to 2.0 seconds in the clarinet, and from
; .5 to 2.1 seconds in the flute). The onset
; and decay portions in the morphed sound are
; specified by parameters p4 and p5, respectively.
; The morphing time is the time between the
; onset and the decay. The clarinet partials are
; shifted in pitch to match the pitch of the flute
; tone (D above middle C).
;
instr 1
  ionset   =          p4
  idecay   =          p5
  itmorph  =          p3 - (ionset + idecay)
  ipshift  =          cpspch(8.02)/cpspch(8.08)

  ; clarinet time function, morph from .2 to 2.0 seconds
  ktcl     linseg     0, ionset, .2, itmorph, 2.0, idecay, 2.1
  ; flute time function, morph from .5 to 2.1 seconds
  ktfl     linseg     0, ionset, .5, itmorph, 2.1, idecay, 2.3
  kmurph   linseg     0, ionset, 0, itmorph, 1, idecay, 1
  lorisread ktcl, "clarinet.sdif", 1, ipshift, 2, 1, .001
  lorisread ktfl, "flute.sdif", 2, 1, 1, 1, .001
  lorisread kmurph, kmurph, kmurph
  asig     lorisplay  3, 1, 1, 1
  out      out

endin

```

Example 6. More morphing

```

; Morph the partials in trombone.sdif into the
; partials in meow.sdif. The start and end times
; for the morph are specified by parameters p4
; and p5, respectively. The morph occurs over the
; second of four pitches in each of the sounds,
; from .75 to 1.2 seconds in the flutter-tongued
; trombone tone, and from 1.7 to 2.2 seconds in
; the cat's meow. Different morphing functions are
; used for the frequency and amplitude envelopes,
; so that the partial amplitudes make a faster
; transition from trombone to cat than the frequencies.
; (The bandwidth envelopes use the same morphing
; function as the amplitudes.)
;
instr 2
  ionset = p4
  imorph = p5 - p4
  irelease = p3 - p5

  kttbn linseg 0, ionset, .75, imorph, 1.2, irelease, 2.4
  ktmeow linseg 0, ionset, 1.7, imorph, 2.2, irelease, 3.4

  kmfreq linseg 0, ionset, 0, .75*imorph, .25, .25*imorph, 1, irelease, 1
  kmamp linseg 0, ionset, 0, .75*imorph, .9, .25*imorph, 1, irelease, 1

  lorisread kttbn, "trombone.sdif", 1, 1, 1, 1, .001
  lorisread ktmeow, "meow.sdif", 2, 1, 1, 1, .001
  lorismorph 1, 2, 3, kmfreq, kmamp, kmamp
  asig lorisplay 3, 1, 1, 1
  out asig
endin

```

The instrument in the first morphing example performs a sound morph between a clarinet tone and a flute tone using reassigned bandwidth-enhanced partials stored in `clarinet.sdif` and `flute.sdif`.

The morph is performed over the sustain portions of the tones, 2. seconds to 2.0 seconds in the case of the clarinet tone and .5 seconds to 2.1 seconds in the case of the flute tone. The time index functions, `ktcl` and `ktfl`, align the onset and decay portions of the tones with the specified onset and decay times for the morphed sound, specified by parameters `p4` and `p5`, respectively. The onset in the morphed sounds is purely clarinet partial data, and the decay is purely flute data. The clarinet partials are shifted in pitch to match the pitch of the flute tone (D above middle C).

The instrument in the second morphing example performs a sound morph between a flutter-tongued trombone tone and a cat's meow using reassigned bandwidth-enhanced partials stored in `trombone.sdif` and `meow.sdif`. The data in these SDIF files have been channelized and distilled to establish correspondences between partials.

The two sets of partials are imported and stored in memory locations labeled 1 and 2, respectively. Both of the original sounds have four notes, and the morph is performed over the second note in each sound (from .75 to 1.2 seconds in the flutter-tongued trombone tone, and from 1.7 to 2.2 seconds in the cat's meow). The different time index functions, `kttbn` and `ktmeow`, align those segments of the source and target partial sets with the specified morph start, morph end, and overall duration parameters. Two different morphing functions are used, so that the partial amplitudes and bandwidth coefficients morph quickly from the trombone values to the cat's-meow values, and the frequencies make a more gradual transition. The morphed partials are stored in a memory location labeled 3 and rendered by the subsequent `lorisplay` instruction. They could also have been used as a source for another morph in a three-way morphing instrument. The following score can be used to test both of the instruments described above.

```

; play instr 1
;   strt dur onset decay
i 1 0 3 .25 .15
i 1 + 1 .10 .10
i 1 + 6 1. 1.
s

; play instr 2

```



```
;      strt  dur  morph_start  morph_end  
i 2      0      4      .75      2.75  
e
```

Credits

This implementation of the Loris unit generators was written by Kelly Fitz (loris@cerlsoundgroup.org [mailto:loris@cerlsoundgroup.org]).

It is patterned after a prototype implementation of the *lorisplay* unit generator written by Corbin Champion, and based on the method of Bandwidth-Enhanced Additive Synthesis and on the sound morphing algorithms implemented in the Loris library for sound modeling and manipulation. The opcodes were further adapted as a plugin for Csound 5 by Michael Gogins.

Strings

String variables are variables with a name starting with S or gS (for a local or global string variable, respectively), and can store any string with a maximum length defined by the `--max_str_len` command line flag (255 characters by default). These variables can be used as input argument to any opcode that expects a quoted string constant, and can be manipulated at initialization or performance time with the opcodes listed below.

It is also possible to use string p-fields. The string p-field can be used by many orchestra opcodes directly, or it can be copied to a string variable first:

```
a1    diskin2 p5, 1
```

```
Sname strget p5  
a1    diskin2 Sname, 1
```

Strings within Csound can be expressed using traditional double quotes (" "), and also using {{ }}. The second method is useful to allow ';' and '\$' characters within the string without having to use ASCII codes.



Note

String variables and related opcodes are not available in Csound versions older than 5.00.

Strings can also be linked to a number using *strset* and *strget*.

Csound 5 also has improvements in parsing string constants. It is possible to specify a multi-line string by enclosing it within {{ and }} instead of the usual double quote characters (note that the length of string constants is not limited, and is not affected by the `--max_str_len` option), and the following escape sequences are automatically converted:

- `\a` alert bell
- `\b` backspace
- `\n` new line
- `\r` carriage return
- `\t` tab
- `\\` a single backslash character
- `\nnn` the character of which the ASCII code (in octal) is nnn

It can be useful together with the *system* opcode:

```
instr 1  
; csound5 lets you make a string with line returns inside double brackets  
  system {{ ps  
    date  
    cd ~/Desktop  
    pwd  
    ls -l  
    whois csounds.com  
  }}  
endin
```

And the *python* opcodes, among others:

```
pyruni {{
import random

pool = [(1 + i/10.0) ** 1.2 for i in range(100)]

def get_number_from_pool(n, p):
    if random.random() < p:
        i = int(random.random() * len(pool))
        pool[i] = n
    return random.choice(pool)
}}
```

String Manipulation Opcodes

These opcodes perform operations on string variables (note: most of the opcodes run at init time only, and have a version with a "k" suffix that runs at both init and performance time; exceptions to this rule include puts and strget):

- *strcpy* and *strcpyk* - Assigns to a string variable.
- *strcat* and *strcatk* - Concatenates strings, and stores the result in a variable.
- *strcmp* and *strcmpk* - Compares strings.
- *strget* - Assigns to a string variable, from strset table at the specified index, or string score p-field.
- *strlen* and *strlenk* - Returns the length of a string.
- *sprintf* - printf-style formatted output conversion, storing the result in a string variable.
- *sprintfk* - printf-style formatted output conversion, storing the result in a string variable at k-rate.
- *puts* - Prints a string constant or variable.
- *strindex* and *strindexk* - Returns the first occurrence of a string in another string.
- *strrindex* and *strrindexk* - Returns the last occurrence of a string in another string.
- *strsub* and *strsubk* - Returns a substring of the input string.

String Conversion Opcodes

These opcodes convert string variables (note: most of the opcodes run at init time only, and have a version with a "k" suffix that runs at both init and performance time; exceptions to this rule include puts and strget):

- *strtod* and *strtodk* - Converts string value to a floating point value at i-rate.
- *strtol* and *strtolk* - Converts string value to signed integer at i-rate.
- *strchar* and *strchark* - Returns the ASCII code of a character in a string.
- *strlower* and *strlowerk* - Converts a string to lower case.
- *strupper* and *strupperk* - Converts a string to upper case.

Vectorial Opcodes

The vectorial opcode family is designed to allow sections of f-tables to be treated as vectors for diverse operations on them.

Tables of vectors operators

The following Vectorial opcodes support read/write access to arrays of vectors (or arrays of arrays):

- *vtablei*
- *vtablelk*
- *vtablek*
- *vtablea*
- *vtablewi*
- *vtablewk*
- *vtablewa*
- *vtabi*
- *vtabk*
- *vtaba*
- *vtabwi*
- *vtabwk*
- *vtabwa*

Operations Between a Vectorial and a Scalar Signal

These opcodes perform numeric operations between a vectorial control signal (hosted inside a function table), and a scalar signal. Result is a new vector that overrides old values of the table. There are k-rate and i-rate versions of the opcodes.

All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Operations Between a Vectorial and a Scalar Signal:

- *vadd*
- *vmult*
- *vpow*
- *vexp*
- *vadd_i*

- *vmult_i*
- *vpow_i*
- *vexp_i*

Operations Between two Vectorial Signals

These opcodes perform operations between two vectors, that is, each element of the first vector is processed with the corresponding element of the other vector. The result is a new vector that overrides the old values of the source vector.

Operations Between two Vectorial Signals:

- *vaddv*
- *vsbv*
- *vmultv*
- *vdivv*
- *vpowv*
- *vexpv*
- *vcopy*
- *vmap*
- *vaddv_i*
- *vsbv_i*
- *vmultv_i*
- *vdivv_i*
- *vpowv_i*
- *vexpv_i*
- *vcopy_i*

All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *vcella*, *adsynt*, *adsynt2*, etc.

Vectorial Envelope Generators

The opcodes to generate vectors containing envelopes are *vlinseg* and *vexpseg*.

These opcodes are similar to *linseg* and *expseg*, but operate with vectorial signals instead of with scalar signals.

Output is a vector hosted by an f-table (that must be previously allocated), while each break-point of the envelope is actually a vector of values. All break-points must contain the same number of elements (*ielements*).

These operators are designed to be used together with other opcodes that operate with vectorial signals such as *vcella*, *adsynt*, *adsynt2*, etc.

Limiting and wrapping of vectorial control signals

The opcodes to perform limiting and wrapping of elements within a vector are:

- *vlimit*
- *vwrap*
- *vmirror*

These opcodes are similar to *limit*, *wrap* and *mirror*, but operate on a vector instead of a scalar signal. The old values of the vector contained in an f-table are over-written if they are out of min/max interval. If you want to keep the original values of the input vector, use the *vcopy* opcode to copy it in another table.

All these opcodes work at k-rate.

All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *vcella*, *adsynt*, *adsynt2* etc.

Vectorial Control-rate Delay Paths

Vectorial Control-rate Delay Paths:

- *vdelayk*
- *vport*
- *vecdelay*

Vectorial Random Signal Generators

These opcodes generate vectors of random numbers to be stored in tables. They generate a sort of 'vectorial band-limited noise'. All these opcodes work at k-rate.

Vectorial random signal generators: *vrandh* and *vrandi*.

Cellular automata vectors can be generated using: *vcella*.

Zak Patch System

The zak opcodes are used to create a system for i-rate, k-rate or a-rate patching. The zak system can be thought of as a global array of variables. These opcodes are useful for performing flexible patching or routing from one instrument to another. The system is similar to a patching matrix on a mixing console or to a modulation matrix on a synthesizer. It is also useful whenever an array of variables is required.

The zak system is initialized by the *zakinit* opcode, which is usually placed just after the other global initializations: *sr*, *kr*, *ksmps*, *nchnls*. The *zakinit* opcode defines two areas of memory, one area for i- and k-rate patching, and the other area for a-rate patching. The *zakinit* opcode may only be called once. Once the zak space is initialized, other zak opcodes can be used to read from, and write to the zak memory space, as well as perform various other tasks.

Zak channels count from 0, so if you define 1 channel, the only valid channel is channel 0.

Opcodes for the zak patch system are:

- Audio Rate: *zacl*, *zakinit*, *zamod*, *zar*, *zarg*, *zaw* and *zawm*.
- Control Rate: *zkcl*, *zkmod*, *zkr*, *zkw*, and *zkwm*.
- At initialization: *zir*, *ziw* and *ziwm*

Plugin Hosting

Csound currently hosts external plugins using *dssi4cs* (for LADSPA plugins) on Linux and *vst4cs* (for VST plugins) on Windows and Mac OS X.

DSSI and LADSPA for Csound

dssi4cs enables the use of DSSI and LADSPA plugin effects and synthesizers within Csound on Linux. The following opcodes are available:

- *dssiinit* - Loads a plugin.
- *dssiactivate* - Activates or deactivates a plugin if it has this facility
- *dssilist* - Lists all available plugins found in the LADSPA_PATH and DSSI_PATH global variables.
- *dssiaudio* - Process audio using a Plugin.
- *dssictrls* - Send control information to a plugin's control port.

See the entry for *dssiinit* for a usage example.



Note

Currently only LADSPA plugins are supported, but DSSI support is planned.

VST for Csound

vst4cs enables the use of VST plugin effects and synthesizers within Csound. The following opcodes are available:

- *vstinit* - Loads a plugin.
- *vstaudio*, *vstaudiog* - Returns a plugin's output.
- *vstmidiout* - Sends MIDI data to a plugin.
- *vstparamset*, *vstparamget* - Sends and receives automation data to and from the plugin.
- *vstnote* - Sends a MIDI note with definite duration.
- *vstinfo* - Outputs the Parameter and Program names for a plugin.
- *vstbankload* - Loads an `.fxb` Bank.
- *vstprogset* - Sets a Program in an `.fxb` Bank.
- *vstedit* - Opens the GUI editor for the plugin, when available.

Credits

By: Andres Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's vst~ object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

OSC and Network

OSC

OSC enables interaction between different audio processes, and in particular between Csound and other synthesis engines. The following opcodes are available:

- *OSCinit* - Start an OSC listener thread.
- *OSClisten* - Receive OSC messages.
- *OSCsend* - Send an OSC message.

Credits

By: John ffitich with the liblo library as inspiration and support.

Network

The following opcodes can stream or receive audio through UDP:

- *sockrecv*
- *socksend*

Remote Opcodes

The Remote opcodes enable transmission of score or MIDI events through a network, so remote instances (or a different local instance) can process them. The following opcodes are available:

- *insglobal* - Used to implement a remote orchestra.
- *insremot* - Used to implement a remote orchestra.
- *midiglobal* - Used to implement a remote MIDI orchestra.
- *midiremot* - Used to implement a remote MIDI orchestra.
- *remoteport* - Defines the port for use with the remote system.

Mixer Opcodes

The Mixer family of opcodes provides a global mixer for Csound. The Mixer opcodes include *MixerSend* for sending (that is, mixing in) an arate signal from any instrument to a channel of a mixer buss, *MixerReceive* for receiving an arate signal from a channel of any mixer buss in any instrument, *MixerSetLevel* for controlling (at krte) the level of the signal sent from a particular send to a particular buss, *MixerGetLevel* for reading (at krte) the level for sending a signal from a particular send to a particular buss, and *MixerClear* for resetting the busses to zero before the next kperiod of a performance.

Python Opcodes

Introduction

Using the Python opcode family, you can interact with a Python interpreter embedded in Csound in five ways:

1. Initialize the Python interpreter (the *pyinit* opcodes),
2. Run a statement (the *pyrun* opcodes),
3. Execute a script (the *pyexec* opcodes),
4. Invoke a callable and pass arguments (the *pycall* opcodes),
5. Evaluate an expression (the *pyeval* opcodes), or
6. Change the value of a Python object, possibly creating a new Python object (the *pyassign* opcodes);

and you can do any of these things:

1. At i-time or at k-time,
2. In the global Python namespace, or in a namespace specific to an individual instance of a Csound instrument (local or "l" context),
3. And can you can retrieve from 0 to 8 return values from callables that accept N parameters.

...this means that there are many Python-related opcodes. But all of these opcodes share the same *py* prefix, and have a regular naming scheme:

```
"py" + [optional context prefix] + [action name] + [optional x-time suffix]
```

Orchestra Syntax

Blocks of Python code, and indeed entire scripts, can be embedded in Csound orchestras using the `{{` and `}}` directives to enclose the script, as follows:

```
sr=44100
kr=4410
ksmps=10
nchnls=1
pyinit

giSinusoid ftgen 0, 0, 8192, 10, 1

pyruni {{
import random

pool = [(1 + i/10.0) ** 1.2 for i in range(100)]

def get_number_from_pool(n, p):
    if random.random() < p:
        i = int(random.random() * len(pool))
        pool[i] = n
    return random.choice(pool)
}}

instr 1
    k1 oscil 1, 3, giSinusoid
```

```
        k2 pycall1 "get_number_from_pool", k1 + 2, p4  
        printk 0.01, k2  
endin
```

Credits

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Image processing opcodes

Here is a list of opcodes that read/write image files:

- *imagecreate*
- *imagesize*
- *imagegetpixel*
- *imagesetpixel*
- *imagesave*
- *imageload*
- *imagefree*

Miscellaneous opcodes

Here is a list of opcodes that don't fall in any category:

- *system* - Call an external program via the system call.

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Orchestra Opcodes and Operators

!=

!= — Determines if one value is not equal to another.

Description

Determines if one value is not equal to another.

Syntax

```
(a != b ? v1 : v2)
```

where *a*, *b*, *v1* and *v2* may be expressions, but *a*, *b* not audio-rate.

Performance

In the above conditional, *a* and *b* are first compared. If the indicated relation is true (*a* not equal to *b*), then the conditional expression has the value of *v1*; if the relation is false, the expression has the value of *v2*.

NB.: If *v1* or *v2* are expressions, these will be evaluated before the conditional is determined.

In terms of binding strength, all conditional operators (i.e. the relational operators (<, etc.), and ?, and :) are weaker than the arithmetic and logical operators (+, -, *, /, && and //).

These are *operators* not *opcodes*. Therefore, they can be used within orchestra statements, but do not form complete statements themselves.

Examples

Here is an example of the != operator. It uses the file *notequal.csd* [examples/notequal.csd].

Example 7. Example of the != operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o notequal.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1.
instr 1
; Get the 4th p-field from the score.
k1 = p4

; Is it not equal to 3? (1 = true, 0 = false)
k2 = (p4 != 3 ? 1 : 0)

; Print the values of k1 and k2.
```

```
    printks "k1 = %f, k2 = %f\\n", 1, k1, k2
  endin

</CsInstruments>
<CsScore>

; Call Instrument #1 with a p4 = 2.
i 1 0 0.5 2
; Call Instrument #1 with a p4 = 3.
i 1 1 0.5 3
; Call Instrument #1 with a p4 = 4.
i 1 2 0.5 4
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1 = 2.000000, k2 = 1.000000
k1 = 3.000000, k2 = 0.000000
k1 = 4.000000, k2 = 1.000000
```

See Also

`==`, `>=`, `>`, `<=`, `<`

Credits

Example written by Kevin Conder.

#define

#define — Defines a macro.

Description

Macros are textual replacements which are made in the orchestra as it is being read. The macro system in Csound is a very simple one, and uses the characters # and \$ to define and call macros. This can save typing, and can lead to a coherent structure and consistent style. This is similar to, but independent of, the *macro system in the score language*.

#define NAME -- defines a simple macro. The name of the macro must begin with a letter and can consist of any combination of letters and numbers. Case is significant. This form is limiting, in that the variable names are fixed. More flexibility can be obtained by using a macro with arguments, described below.

#define NAME(a' b' c') -- defines a macro with arguments. This can be used in more complex situations. The name of the macro must begin with a letter and can consist of any combination of letters and numbers. Within the replacement text, the arguments can be substituted by the form: \$A. In fact, the implementation defines the arguments as simple macros. There may be up to 5 arguments, and the names may be any choice of letters. Remember that case is significant in macro names.

Syntax

```
#define NAME # replacement text #
```

```
#define NAME(a' b' c') # replacement text #
```

Initialization

replacement text # -- The replacement text is any character string (not containing a #) and can extend over multiple lines. The replacement text is enclosed within the # characters, which ensure that additional characters are not inadvertently captured.

Performance

Some care is needed with textual replacement macros, as they can sometimes do strange things. They take no notice of any meaning, so spaces are significant. This is why, unlike the C programming language, the definition has the replacement text surrounded by # characters. Used carefully, this simple macro system is a powerful concept, but it can be abused.

Examples

Here is a simple example of the defining a macro. It uses the file *define.csd* [examples/define.csd].

Example 8. Simple example of the define macro.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
```



```
; -o define.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Define the macros.
#define VOLUME #5000#
#define FREQ #440#
#define TABLE #1#

; Instrument #1
instr 1
; Use the macros.
; This will be expanded to "a1 oscil 5000, 440, 1".
a1 oscil $VOLUME, $FREQ, $TABLE

; Send it to the output.
out a1
endin

</CsInstruments>
<CsScore>

; Define Table #1 with an ordinary sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
Macro definition for VOLUME
Macro definition for CPS
Macro definition for TABLE
```

Here is an example of the defining a macro with arguments. It uses the file *define_args.csd* [examples/define_args.csd].

Example 9. Example of the define macro with arguments.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o define_args.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Define the oscillator macro.
#define OSCMACRO(VOLUME'FREQ'TABLE) #oscil $VOLUME, $FREQ, $TABLE#

; Instrument #1
instr 1
; Use the oscillator macro.
; This will be expanded to "a1 oscil 5000, 440, 1".
a1 $OSCMACRO(5000'440'1)

; Send it to the output.
out a1
endin
```

```
</CsInstruments>
<CsScore>

; Define Table #1 with an ordinary sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

Macro definition for OSCMACRO

Predefined Math Constant Macros

New in Csound 5.04 are predefined Math Constant Macros. The values defined are those found in the C header `math.h`, and are automatically defined when Csound starts and available for use in orchestras.

Macro	Value
\$M_E	2.7182818284590452354
\$M_LOG2E	1.4426950408889634074
\$M_LOG10E	0.43429448190325182765
\$M_LN2	0.69314718055994530942
\$M_LN10	2.30258509299404568402
\$M_PI	3.14159265358979323846
\$M_PI_2	1.57079632679489661923
\$M_PI_4	0.78539816339744830962
\$M_1_PI	0.31830988618379067154
\$M_2_PI	0.63661977236758134308
\$M_2_SQRTPI	1.12837916709551257390
\$M_SQRT2	1.41421356237309504880
\$M_SQRT1_2	0.70710678118654752440

See Also

\$NAME, #undef

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
April 1998

Examples written by Kevin Conder.

New in Csound version 3.48

#include

#include — Includes an external file for processing.

Description

Includes an external file for processing.

Syntax

```
#include "filename"
```

Performance

It is sometimes convenient to have the orchestra arranged in a number of files, for example with each instrument in a separate file. This style is supported by the *#include* facility which is part of the macro system. A line containing the text

```
#include "filename"
```

where the character " can be replaced by any suitable character. For most uses the double quote symbol will probably be the most convenient. The file name can include a full path.

This takes input from the named file until it ends, when input reverts to the previous input. *Note:* Csound versions prior to 4.19 had a limit of 20 on the depth of included files and macros.

Another suggested use of *#include* would be to define a set of macros which are part of the composer's style.

An extreme form would be to have each instrument defines as a macro, with the instrument number as a parameter. Then an entire orchestra could be constructed from a number of *#include* statements followed by macro calls.

```
#include "clarinet"  
#include "flute"  
#include "bassoon"  
$CLARINET(1)  
$FLUTE(2)  
$BASSOON(3)
```

It must be stressed that these changes are at the textual level and so take no cognizance of any meaning.

Examples

Here is an example of the include opcode. It uses the file *include.csd* [examples/include.csd], and *table1.inc* [examples/table1.inc].

Example 10. Example of the include opcode.

```
/* table1.inc */  
; Table #1, a sine wave.  
f 1 0 16384 10 1  
/* table1.inc */
```

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o include.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a basic oscillator.
instr 1
  kamp = 10000
  kcps = 440
  ifn = 1

  al oscil kamp, kcps, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Include the file for Table #1.
#include "table1.inc"

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
April 1998

Example written by Kevin Conder.

New in Csound version 3.48

#undef

#undef — Un-defines a macro.

Description

Macros are textual replacements which are made in the orchestra as it is being read. The macro system in Csound is a very simple one, and uses the characters # and \$ to define and call macros. This can save typing, and can lead to a coherent structure and consistent style. This is similar to, but independent of, the *macro system in the score language*.

#undef NAME -- undefines a macro name. If a macro is no longer required, it can be undefined with *#undef NAME*.

Syntax

#undef NAME

Performance

Some care is needed with textual replacement macros, as they can sometimes do strange things. They take no notice of any meaning, so spaces are significant. This is why, unlike the C programming language, the definition has the replacement text surrounded by # characters. Used carefully, this simple macro system is a powerful concept, but it can be abused.

See Also

#define, *\$NAME*

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
April 1998

New in Csound version 3.48

#ifdef

#ifdef — Conditional reading of code.

Description

If a macro is defined then *#ifdef* can incorporate text into an orchestra upto the next *#end*. This is similar to, but independent of, the *macro system in the score language*.

Syntax

```
#ifdef NAME  
  
....  
  
#else  
  
....  
  
#end
```

Performance

Note that the *#ifdef* can be nested, like in the C preprocessor language.

Examples

Here is a simple example of the conditional.

Example 11. Simple example of the #ifdef form.

```
#define debug  
  instr 1  
#ifdef debug  
  print "calling oscil"  
#end  
  a1 oscil 32000,440,1  
  out a1  
  endin
```

See Also

\$NAME, *#define*, *#ifndef*.

Credits

Author: John fitch
University of Bath/Codemist Ltd.
Bath, UK
April 2005

New in Csound5 (and 4.23f13)

#ifndef

`#ifndef` — Conditional reading of code.

Description

If the specified macro is not defined then *#ifndef* can incorporate text into an orchestra upto the next *#end*. This is similar to, but independent of, the *macro system in the score language*.

Syntax

```
#ifndef NAME  
  
....  
  
#else  
  
....  
  
#end
```

Performance

Note that the *#ifndef* can be nested, like in the C preprocessor language.

See Also

\$NAME, *#define*, *#ifdef*.

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
April 2005

New in Csound5 (and 4.23f13)

\$NAME

\$NAME — Calls a defined macro.

Description

Macros are textual replacements which are made in the orchestra as it is being read. The macro system in Csound is a very simple one, and uses the characters # and \$ to define and call macros. This can save typing, and can lead to a coherent structure and consistent style. This is similar to, but independent of, the *macro system in the score language*.

\$NAME -- calls a defined macro. To use a macro, the name is used following a \$ character. The name is terminated by the first character which is neither a letter nor a number. If it is necessary for the name not to terminate with a space, a period, which will be ignored, can be used to terminate the name. The string, *\$NAME.*, is replaced by the replacement text from the definition. The replacement text can also include macro calls.

Syntax

\$NAME

Initialization

replacement text # -- The replacement text is any character string (not containing a #) and can extend over multiple lines. The replacement text is enclosed within the # characters, which ensure that additional characters are not inadvertently captured.

Performance

Some care is needed with textual replacement macros, as they can sometimes do strange things. They take no notice of any meaning, so spaces are significant. This is why, unlike the C programming language, the definition has the replacement text surrounded by # characters. Used carefully, this simple macro system is a powerful concept, but it can be abused.

Examples

Here is an example of the calling a macro. It uses the file *define.csd* [examples/define.csd].

Example 12. An example of the calling a macro.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o define.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```



```
; Define the macros.
#define VOLUME #5000#
#define FREQ #440#
#define TABLE #1#

; Instrument #1
instr 1
; Use the macros.
; This will be expanded to "a1 oscil 5000, 440, 1".
a1 oscil $VOLUME, $FREQ, $TABLE

; Send it to the output.
out a1
endin

</CsInstruments>
<CsScore>

; Define Table #1 with an ordinary sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
Macro definition for VOLUME
Macro definition for CPS
Macro definition for TABLE
```

Here is an example of the calling a macro with arguments. It uses the file *define_args.csd* [examples/define_args.csd].

Example 13. An example of the calling a macro with arguments.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out Audio in No messages
-odac -iadc -d ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o define_args.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Define the oscillator macro.
#define OSCMACRO(VOLUME'FREQ'TABLE) #oscil $VOLUME, $FREQ, $TABLE#

; Instrument #1
instr 1
; Use the oscillator macro.
; This will be expanded to "a1 oscil 5000, 440, 1".
a1 $OSCMACRO(5000'440'1)

; Send it to the output.
out a1
endin

</CsInstruments>
<CsScore>

; Define Table #1 with an ordinary sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
```

e

```
</CsScore>  
</CsoundSynthesizer>
```

Its output should include a line like this:

```
Macro definition for OSCMACRO
```

See Also

#define, #undef

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
April, 1998

Examples written by Kevin Conder.

New in Csound version 3.48

%

% — Modulus operator.

Description

Arithmetic operators perform operations of change-sign (negate), don't-change-sign, logical AND logical OR, add, subtract, multiply and divide. Note that a value or an expression may fall between two of these operators, either of which could take it as its left or right argument, as in

$a + b * c$.

In such cases three rules apply:

1. $*$ and $/$ bind to their neighbors more strongly than $+$ and $-$. Thus the above expression is taken as

$a + (b * c)$

with $*$ taking b and c and then $+$ taking a and $b * c$.

2. $+$ and $-$ bind more strongly than $\&\&$, which in turn is stronger than $\|$:

$a \&\& b - c \| d$

is taken as

$(a \&\& (b - c)) \| d$

3. When both operators bind equally strongly, the operations are done left to right:

$a - b - c$

is taken as

$(a - b) - c$

Parentheses may be used as above to force particular groupings.

The operator $\%$ returns the value of a reduced by b , so that the result, in absolute value, is less than the absolute value of b , by repeated subtraction. This is the same as modulus function in integers. New in Csound version 3.50.

Syntax

`a % b` (no rate restriction)

where the arguments a and b may be further expressions.

Examples

Here is an example of the $\%$ operator. It uses the file *modulus.csd* [examples/modulus.csd].

Example 14. Example of the $\%$ operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o modulus.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  il = 5 % 3
  print il
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  il = 2.000
```

See Also

-, +, &&, //, *, /, ^

Credits

Example written by Kevin Conder.

&&

&& — Logical AND operator.

Description

Arithmetic operators perform operations of change-sign (negate), don't-change-sign, logical AND logical OR, add, subtract, multiply and divide. Note that a value or an expression may fall between two of these operators, either of which could take it as its left or right argument, as in

$a + b * c$.

In such cases three rules apply:

1. $*$ and $/$ bind to their neighbors more strongly than $+$ and $-$. Thus the above expression is taken as

$a + (b * c)$

with $*$ taking b and c and then $+$ taking a and $b * c$.

2. $+$ and $-$ bind more strongly than $\&\&$, which in turn is stronger than $\|$:

$a \&\& b - c \| d$

is taken as

$(a \&\& (b - c)) \| d$

3. When both operators bind equally strongly, the operations are done left to right:

$a - b - c$

is taken as

$(a - b) - c$

Parentheses may be used as above to force particular groupings.

Syntax

$a \&\& b$ (logical AND; not audio-rate)

where the arguments a and b may be further expressions.

See Also

$-$, $+$, $\|$, $*$, $/$, $^$, $\%$

>

> — Determines if one value is greater than another.

Description

Determines if one value is greater than another.

Syntax

```
(a > b ? v1 : v2)
```

where a , b , $v1$ and $v2$ may be expressions, but a , b not audio-rate.

Performance

In the above conditional, a and b are first compared. If the indicated relation is true (a greater than b), then the conditional expression has the value of $v1$; if the relation is false, the expression has the value of $v2$.

NB.: If $v1$ or $v2$ are expressions, these will be evaluated before the conditional is determined.

In terms of binding strength, all conditional operators (i.e. the relational operators ($<$, etc.), and $?$, and $:$) are weaker than the arithmetic and logical operators ($+$, $-$, $*$, $/$, $\&\&$ and $//$).

These are *operators* not *opcodes*. Therefore, they can be used within orchestra statements, but do not form complete statements themselves.

Examples

Here is an example of the $>$ operator. It uses the file *greaterthan.csd* [examples/greaterthan.csd].

Example 15. Example of the $>$ operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o greaterthan.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1.
instr 1
; Get the 4th p-field from the score.
k1 = p4

; Is it greater than 3? (1 = true, 0 = false)
k2 = (p4 > 3 ? 1 : 0)

; Print the values of k1 and k2.
```

```
    printks "k1 = %f, k2 = %f\\n", 1, k1, k2
  endin

</CsInstruments>
<CsScore>

; Call Instrument #1 with a p4 = 2.
i 1 0 0.5 2
; Call Instrument #1 with a p4 = 3.
i 1 1 0.5 3
; Call Instrument #1 with a p4 = 4.
i 1 2 0.5 4
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1 = 2.000000, k2 = 0.000000
k1 = 3.000000, k2 = 0.000000
k1 = 4.000000, k2 = 1.000000
```

See Also

`==`, `>=`, `<=`, `<`, `!=`

Credits

Example written by Kevin Conder.

>=

>= — Determines if one value is greater than or equal to another.

Description

Determines if one value is greater than or equal to another.

Syntax

```
(a >= b ? v1 : v2)
```

where *a*, *b*, *v1* and *v2* may be expressions, but *a*, *b* not audio-rate.

Performance

In the above conditional, *a* and *b* are first compared. If the indicated relation is true (*a* greater than or equal to *b*), then the conditional expression has the value of *v1*; if the relation is false, the expression has the value of *v2*.

NB.: If *v1* or *v2* are expressions, these will be evaluated before the conditional is determined.

In terms of binding strength, all conditional operators (i.e. the relational operators (<, etc.), and ?, and :) are weaker than the arithmetic and logical operators (+, -, *, /, && and //).

These are *operators* not *opcodes*. Therefore, they can be used within orchestra statements, but do not form complete statements themselves.

Examples

Here is an example of the >= operator. It uses the file *greaterequal.csd* [examples/greaterequal.csd].

Example 16. Example of the >= operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o greaterequal.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1.
instr 1
; Get the 4th p-field from the score.
k1 = p4

; Is it greater than or equal to 3? (1 = true, 0 = false)
k2 = (p4 >= 3 ? 1 : 0)

; Print the values of k1 and k2.
```



```
    printks "k1 = %f, k2 = %f\\n", 1, k1, k2
endin

</CsInstruments>
<CsScore>

; Call Instrument #1 with a p4 = 2.
i 1 0 0.5 2
; Call Instrument #1 with a p4 = 3.
i 1 1 0.5 3
; Call Instrument #1 with a p4 = 4.
i 1 2 0.5 4
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1 = 2.000000, k2 = 0.000000
k1 = 3.000000, k2 = 1.000000
k1 = 4.000000, k2 = 1.000000
```

See Also

`==`, `>`, `<=`, `<`, `!=`

Credits

Example written by Kevin Conder.

<

< — Determines if one value is less than another.

Description

Determines if one value is less than another.

Syntax

```
(a < b ? v1 : v2)
```

where a , b , $v1$ and $v2$ may be expressions, but a , b not audio-rate.

Performance

In the above conditional, a and b are first compared. If the indicated relation is true (a less than b), then the conditional expression has the value of $v1$; if the relation is false, the expression has the value of $v2$.

NB.: If $v1$ or $v2$ are expressions, these will be evaluated before the conditional is determined.

In terms of binding strength, all conditional operators (i.e. the relational operators (<, etc.), and ?, and :) are weaker than the arithmetic and logical operators (+, -, *, /, && and //).

These are *operators* not *opcodes*. Therefore, they can be used within orchestra statements, but do not form complete statements themselves.

Examples

Here is an example of the < operator. It uses the file *lessthan.csd* [examples/lessthan.csd].

Example 17. Example of the < operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o lessthan.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1.
instr 1
; Get the 4th p-field from the score.
k1 = p4

; Is it less than 3? (1 = true, 0 = false)
k2 = (p4 < 3 ? 1 : 0)

; Print the values of k1 and k2.
```

```
    printks "k1 = %f, k2 = %f\\n", 1, k1, k2
  endin

</CsInstruments>
<CsScore>

; Call Instrument #1 with a p4 = 2.
i 1 0 0.5 2
; Call Instrument #1 with a p4 = 3.
i 1 1 0.5 3
; Call Instrument #1 with a p4 = 4.
i 1 2 0.5 4
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1 = 2.000000, k2 = 1.000000
k1 = 3.000000, k2 = 0.000000
k1 = 4.000000, k2 = 0.000000
```

See Also

`==`, `>=`, `>`, `<=`, `!=`

Credits

Example written by Kevin Conder.

<=

<= — Determines if one value is less than or equal to another.

Description

Determines if one value is less than or equal to another.

Syntax

```
(a <= b ? v1 : v2)
```

where *a*, *b*, *v1* and *v2* may be expressions, but *a*, *b* not audio-rate.

Performance

In the above conditional, *a* and *b* are first compared. If the indicated relation is true (*a* less than or equal to *b*), then the conditional expression has the value of *v1*; if the relation is false, the expression has the value of *v2*.

NB.: If *v1* or *v2* are expressions, these will be evaluated before the conditional is determined.

In terms of binding strength, all conditional operators (i.e. the relational operators (<, etc.), and ?, and :) are weaker than the arithmetic and logical operators (+, -, *, /, && and //).

These are *operators* not *opcodes*. Therefore, they can be used within orchestra statements, but do not form complete statements themselves.

Examples

Here is an example of the <= operator. It uses the file *lessequal.csd* [examples/lessequal.csd].

Example 18. Example of the <= operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o lessequal.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1.
instr 1
; Get the 4th p-field from the score.
k1 = p4

; Is it less than or equal to 3? (1 = true, 0 = false)
k2 = (p4 <= 3 ? 1 : 0)

; Print the values of k1 and k2.
```

```
    printks "k1 = %f, k2 = %f\\n", 1, k1, k2
  endin

</CsInstruments>
<CsScore>

; Call Instrument #1 with a p4 = 2.
i 1 0 0.5 2
; Call Instrument #1 with a p4 = 3.
i 1 1 0.5 3
; Call Instrument #1 with a p4 = 4.
i 1 2 0.5 4
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1 = 2.000000, k2 = 1.000000
k1 = 3.000000, k2 = 1.000000
k1 = 4.000000, k2 = 0.000000
```

See Also

`==`, `>=`, `>`, `<`, `!=`

Credits

Example written by Kevin Conder.

* — Multiplication operator.

Description

Arithmetic operators perform operations of change-sign (negate), don't-change-sign, logical AND logical OR, add, subtract, multiply and divide. Note that a value or an expression may fall between two of these operators, either of which could take it as its left or right argument, as in

$a + b * c$.

In such cases three rules apply:

1. * and / bind to their neighbors more strongly than + and #. Thus the above expression is taken as

$a + (b * c)$

with * taking b and c and then + taking a and b * c.

2. + and # bind more strongly than &&, which in turn is stronger than ||:

$a \&\& b - c \parallel d$

is taken as

$(a \&\& (b - c)) \parallel d$

3. When both operators bind equally strongly, the operations are done left to right:

$a - b - c$

is taken as

$(a - b) - c$

Parentheses may be used as above to force particular groupings.

Syntax

$a * b$ (no rate restriction)

where the arguments a and b may be further expressions.

Examples

Here is an example of the `*` operator. It uses the file *multiplies.csd* [examples/multiplies.csd].

Example 19. Example of the `*` operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o multiplies.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  il = 24 * 8
  print il
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  il = 192.000
```

See Also

`-`, `+`, `&&`, `//`, `/`, `^`, `%`

Credits

Example written by Kevin Conder.

+

+ — Addition operator

Description

Arithmetic operators perform operations of change-sign (negate), don't-change-sign, logical AND logical OR, add, subtract, multiply and divide. Note that a value or an expression may fall between two of these operators, either of which could take it as its left or right argument, as in

$a + b * c.$

In such cases three rules apply:

1. $*$ and $/$ bind to their neighbors more strongly than $+$ and $\#$. Thus the above expression is taken as

$a + (b * c)$

with $*$ taking b and c and then $+$ taking a and $b * c$.

2. $+$ and $\#$ bind more strongly than $\&\&$, which in turn is stronger than $\|$:

$a \&\& b - c \| d$

is taken as

$(a \&\& (b - c)) \| d$

3. When both operators bind equally strongly, the operations are done left to right:

$a - b - c$

is taken as

$(a - b) - c$

Parentheses may be used as above to force particular groupings.

Syntax

`+a` (no rate restriction)

`a + b` (no rate restriction)

where the arguments *a* and *b* may be further expressions.

Examples

Here is an example of the + operator. It uses the file *adds.csd* [examples/adds.csd].

Example 20. Example of the + operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o adds.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = 24 + 8
  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  i1 = 32.000
```

See Also

-, &&, //, *, /, ^, %

Credits

Example written by Kevin Conder.

■

- — Subtraction operator.

Description

Arithmetic operators perform operations of change-sign (negate), don't-change-sign, logical AND logical OR, add, subtract, multiply and divide. Note that a value or an expression may fall between two of these operators, either of which could take it as its left or right argument, as in

$a + b * c$.

In such cases three rules apply:

1. $*$ and $/$ bind to their neighbors more strongly than $+$ and $\#$. Thus the above expression is taken as

$a + (b * c)$

with $*$ taking b and c and then $+$ taking a and $b * c$.

2. $+$ and $\#$ bind more strongly than $\&\&$, which in turn is stronger than $\|$:

$a \&\& b - c \| d$

is taken as

$(a \&\& (b - c)) \| d$

3. When both operators bind equally strongly, the operations are done left to right:

$a - b - c$

is taken as

$(a - b) - c$

Parentheses may be used as above to force particular groupings.

Syntax

`#a` (no rate restriction)

`a # b` (no rate restriction)

where the arguments *a* and *b* may be further expressions.

Examples

Here is an example of the - operator. It uses the file *subtracts.csd* [examples/subtracts.csd].

Example 21. Example of the - operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o subtracts.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = 24 - 8
  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  i1 = 16.000
```

See Also

+, &&, //, *, /, ^, %

Credits

Example written by Kevin Conder.

/

/ — Division operator.

Description

Arithmetic operators perform operations of change-sign (negate), don't-change-sign, logical AND logical OR, add, subtract, multiply and divide. Note that a value or an expression may fall between two of these operators, either of which could take it as its left or right argument, as in

$a + b * c$.

In such cases three rules apply:

1. $*$ and $/$ bind to their neighbors more strongly than $+$ and $-$. Thus the above expression is taken as

$a + (b * c)$

with $*$ taking b and c and then $+$ taking a and $b * c$.

2. $+$ and $-$ bind more strongly than $\&\&$, which in turn is stronger than \parallel :

$a \&\& b - c \parallel d$

is taken as

$(a \&\& (b - c)) \parallel d$

3. When both operators bind equally strongly, the operations are done left to right:

$a - b - c$

is taken as

$(a - b) - c$

Parentheses may be used as above to force particular groupings.

Syntax

a / b (no rate restriction)

where the arguments a and b may be further expressions.

Examples

Here is an example of the / operator. It uses the file *divides.csd* [examples/divides.csd].

Example 22. Example of the / operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o divides.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  il = 24 / 8
  print il
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  il = 3.000
```

See Also

-, +, &&, //, *, ^, %

Credits

Example written by Kevin Conder.

=

= — Performs a simple assignment.

Syntax

```
ares = xarg
```

```
ires = iarg
```

```
kres = karg
```

Description

Performs a simple assignment.

Initialization

= (simple assignment) - Put the value of the expression *iarg* (*karg*, *xarg*) into the named result. This provides a means of saving an evaluated result for later use.

Examples

Here is an example of the assign opcode. It uses the file *assign.csd* [examples/assign.csd].

Example 23. Example of the assign opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o assign.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Assign a value to the variable i1.
i1 = 1234

; Print the value of the i1 variable.
print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 1234.000
```

See Also

divz, init, tival

Credits

Example written by Kevin Conder.

==

== — Compares two values for equality.

Description

Compares two values for equality.

Syntax

```
(a == b ? v1 : v2)
```

where a , b , $v1$ and $v2$ may be expressions, but a , b not audio-rate.

Performance

In the above conditional, a and b are first compared. If the indicated relation is true (a is equal to b), then the conditional expression has the value of $v1$; if the relation is false, the expression has the value of $v2$. (For convenience, a sole "=" will function as "==".)

NB.: If $v1$ or $v2$ are expressions, these will be evaluated before the conditional is determined.

In terms of binding strength, all conditional operators (i.e. the relational operators (<, etc.), and ?, and :) are weaker than the arithmetic and logical operators (+, -, *, /, && and //).

These are *operators* not *opcodes*. Therefore, they can be used within orchestra statements, but do not form complete statements themselves.

Examples

Here is an example of the == operator. It uses the file *equals.csd* [examples/equals.csd].

Example 24. Example of the == operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o equal.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1.
instr 1
; Get the 4th p-field from the score.
k1 = p4

; Is it equal to 3? (1 = true, 0 = false)
k2 = (p4 == 3 ? 1 : 0)

; Print the values of k1 and k2.
```



```
    printks "k1 = %f, k2 = %f\\n", 1, k1, k2
  endin

</CsInstruments>
<CsScore>

; Call Instrument #1 with a p4 = 2.
i 1 0 0.5 2
; Call Instrument #1 with a p4 = 3.
i 1 1 0.5 3
; Call Instrument #1 with a p4 = 4.
i 1 2 0.5 4
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1 = 2.000000, k2 = 0.000000
k1 = 3.000000, k2 = 1.000000
k1 = 4.000000, k2 = 0.000000
```

See Also

`>=`, `>`, `<=`, `<`, `!=`

Credits

Example written by Kevin Conder.

^

^ — “Power of” operator.

Description

Arithmetic operators perform operations of change-sign (negate), don't-change-sign, logical AND logical OR, add, subtract, multiply and divide. Note that a value or an expression may fall between two of these operators, either of which could take it as its left or right argument, as in

$a + b * c$.

In such cases three rules apply:

1. $*$ and $/$ bind to their neighbors more strongly than $+$ and $-$. Thus the above expression is taken as

$a + (b * c)$

with $*$ taking b and c and then $+$ taking a and $b * c$.

2. $+$ and $-$ bind more strongly than $\&\&$, which in turn is stronger than $\|$:

$a \&\& b - c \| d$

is taken as

$(a \&\& (b - c)) \| d$

3. When both operators bind equally strongly, the operations are done left to right:

$a - b - c$

is taken as

$(a - b) - c$

Parentheses may be used as above to force particular groupings.

The operator $^$ raises a to the b power. b may not be audio-rate. Use with caution as precedence may not work correctly. See *pow*. (New in Csound version 3.493.)

Syntax

$a \wedge b$ (b not audio-rate)

where the arguments *a* and *b* may be further expressions.

Examples

Here is an example of the ^ operator. It uses the file *raises.csd* [examples/raises.csd].

Example 25. Example of the ^ operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o raises.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = 2 ^ 12
  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 4096.000
```

See Also

-, +, &&, //, *, /, %

Credits

Example written by Kevin Conder.

||

|| — Logical OR operator.

Description

Arithmetic operators perform operations of change-sign (negate), don't-change-sign, logical AND logical OR, add, subtract, multiply and divide. Note that a value or an expression may fall between two of these operators, either of which could take it as its left or right argument, as in

$a + b * c$.

In such cases three rules apply:

1. $*$ and $/$ bind to their neighbors more strongly than $+$ and $-$. Thus the above expression is taken as

$a + (b * c)$

with $*$ taking b and c and then $+$ taking a and $b * c$.

2. $+$ and $-$ bind more strongly than $\&\&$, which in turn is stronger than $||$:

$a \&\& b - c || d$

is taken as

$(a \&\& (b - c)) || d$

3. When both operators bind equally strongly, the operations are done left to right:

$a - b - c$

is taken as

$(a - b) - c$

Parentheses may be used as above to force particular groupings.

Syntax

$a || b$ (logical OR; not audio-rate)

where the arguments a and b may be further expressions.

See Also

`-`, `+`, `&&`, `*`, `/`, `^`, `%`

Odbfs

Odbfs — Sets the value of 0 decibels using full scale amplitude.

Description

Sets the value of 0 decibels using full scale amplitude.

Syntax

```
Odbfs = iarg
```

```
Odbfs
```

Initialization

iarg -- the value of 0 decibels using full scale amplitude.

Performance

The default is 32767, so all existing orcs *should* work.

Amplitude values in Csound are always relative to a "Odbfs" value representing the peak available amplitude before clipping. In the original Csound, this value was always 32767, corresponding to the bipolar range of a 16bit soundfile or 16bit AD/DA codec. This remains the *default* peak amplitude for Csound, for backward compatibility. The *Odbfs* value enables Csound to produce appropriately scaled values to whatever output format is being used, whether 16bit integer, 24bit integer, 32bit floats, or even 32bit integers.

ODBFS can be defined in the header, to set the amplitude reference Csound will use, but it can also be used as a variable inside instruments like this:

```
ipeak = Odbfs
```

```
asig oscil Odbfs, freq, 1  
out  asig * 0.3 * Odbfs
```

The purpose of the *Odbfs* opcode is for people to start to code Odbfs-relatively (and use the *ampdbfs()* opcodes a lot more!), rather than use explicit sample values. Using Odbfs=1 is in accordance to industry practice, as ranges from -1 to 1 are used in most commercial plugin formats and in most other synthesis systems like Pure Data.

Floats written to a file, when *Odbfs* = 1, will in effect go through no range translation at all. So the numbers in the file are exactly what the orc says they are.

For more details on amplitude values in Csound, see the section *Amplitude values in Csound*

Examples

Here is an example of the Odbfs opcode. It uses the file *Odbfs.csd* [examples/Odbfs.csd].

Example 26. Example of the 0dbfs opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o 0dbfs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Set the 0dbfs to the 16-bit maximum.
0dbfs = 32767

; Instrument #1.
instr 1
; Linearly increase the amplitude value "kamp" from
; 0 to 1 over the duration defined by p3.
kamp line 0, p3, 1

; Generate a basic tone using our amplitude value.
a1 oscil kamp, 440, 1

; Multiply the basic tone (with its amplitude between
; 0 and 1) by the full-scale 0dbfs value.
out a1 * 0dbfs
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the 0dbfs opcode. It uses the file *0dbfs-1.csd* [examples/0dbfs-1.csd]. This example has exactly the same output as the previous example, but output samples should now be normalized between -1 and 1.

Example 27. Example of the 0dbfs opcode with maximum amplitude of 1.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o 0dbfs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
```

```
nchnls = 1

; Set the 0dbfs to 1.
0dbfs = 1

; Instrument #1.
instr 1
; Linearly increase the amplitude value "kamp" from
; -90 to p4 (in dBfs) over the duration defined by p3.
kamp line -90, p3, p4
print ampdbfs(p4)
; Generate a basic tone using our amplitude value.
a1 oscil ampdbfs(kamp), 440, 1

; Since 0dbfs = 1 we don't need to multiply the output
out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for three seconds.
i 1 0 3 -6
e

</CsScore>
</CsoundSynthesizer>
```

See also

ampdbfs()

Credits

Author: Richard Dobson
May 2002

New in version 4.10

Example written by Kevin Conder.

Example written by Kevin Conder.

New in version 4.20



<< — Bitshift left operator.

Description

The bitshift operators shift the bits to the left or to the right the number of bits given.

The priority of these operators is less binding than the arithmetic ones, but more binding than the comparisons.

Parentheses may be used as above to force particular groupings.

Syntax

```
a << b  (bitshift left)
```

where the arguments *a* and *b* may be further expressions.

Examples

Here is an example of the bitshift left operator. It uses the file *bitshift.csd* [examples/bitshift.csd].

Example 28. Example of the bitshift left operator.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
;-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
-o bitshift.wav -W --nosound ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 128
nchnls = 2

instr 1 ;bit shift right
ival = p4>>p5
printf_i "%i>>%i = %i\n", 1, p4, p5, ival
endin

instr 2 ;bit shift left
ival = p4<<p5
printf_i "%i<<%i = %i\n", 1, p4, p5, ival
endin

</CsInstruments>
<CsScore>
i 1 0 0.1 2 1
i 1 + . 3 1
i 1 + . 7 2
i 1 + . 16 1
i 1 + . 16 2
i 1 + . 16 3

i 2 5 0.1 1 1
i 2 + . 1 2
i 2 + . 1 3
i 2 + . 1 4
i 2 + . 2 1
```

```
i 2 + . 2 2
i 2 + . 2 3
i 2 + . 3 2
e
</CsScore>
</CsoundSynthesizer>
```

The example above will produce the following output:

```
2>>1 = 1
B 0.000 .. 0.100 T 0.100 TT 0.100 M: 0.0 0.0
3>>1 = 1
B 0.100 .. 0.200 T 0.200 TT 0.200 M: 0.0 0.0
7>>2 = 1
B 0.200 .. 0.300 T 0.300 TT 0.300 M: 0.0 0.0
16>>1 = 8
B 0.300 .. 0.400 T 0.400 TT 0.400 M: 0.0 0.0
16>>2 = 4
B 0.400 .. 0.500 T 0.500 TT 0.500 M: 0.0 0.0
16>>3 = 2
B 0.500 .. 5.000 T 5.000 TT 5.000 M: 0.0 0.0
new alloc for instr 2:
1<<1 = 2
B 5.000 .. 5.100 T 5.100 TT 5.100 M: 0.0 0.0
1<<2 = 4
B 5.100 .. 5.200 T 5.200 TT 5.200 M: 0.0 0.0
1<<3 = 8
B 5.200 .. 5.300 T 5.300 TT 5.300 M: 0.0 0.0
1<<4 = 16
B 5.300 .. 5.400 T 5.400 TT 5.400 M: 0.0 0.0
2<<1 = 4
B 5.400 .. 5.500 T 5.500 TT 5.500 M: 0.0 0.0
2<<2 = 8
B 5.500 .. 5.600 T 5.600 TT 5.600 M: 0.0 0.0
2<<3 = 16
B 5.600 .. 5.700 T 5.700 TT 5.700 M: 0.0 0.0
3<<2 = 12
```

See Also

>>, &, / #

>>

>> — Bitshift right operator.

Description

The bitshift operators shift the bits to the left or to the right the number of bits given.

The priority of these operators is less binding than the arithmetic ones, but more binding than the comparisons.

Parentheses may be used as above to force particular groupings.

Syntax

```
a >> b (bitshift left)
```

where the arguments *a* and *b* may be further expressions.

Examples

See the entry for the << operator for an example.

See Also

<<, &, / #

&

& — Bitwise AND operator.

Description

The bitwise operators perform operations of bitwise AND, bitwise OR, bitwise NOT and bitwise non-equivalence.

The priority of these operators is less binding than the arithmetic ones, but more binding than the comparisons.

Parentheses may be used as above to force particular groupings.

Syntax

`a & b` (bitwise AND)

where the arguments *a* and *b* may be further expressions. They are converted to the nearest integer to machine precision and then the operation is performed.

See Also

`/`, `#`, `⊖`

|

| — Bitwise OR operator.

Description

The bitwise operators perform operations of bitwise AND, bitwise OR, bitwise NOT and bitwise non-equivalence.

The priority of these operators is less binding than the arithmetic ones, but more binding than the comparisons.

Parentheses may be used as above to force particular groupings.

Syntax

`a | b` (bitwise OR)

where the arguments *a* and *b* may be further expressions. They are converted to the nearest integer to machine precision and then the operation is performed.

See Also

`&`, `#`, `¬`

¬

¬ — Bitwise NOT operator.

Description

The bitwise operators perform operations of bitwise AND, bitwise OR, bitwise NOT and bitwise non-equivalence.

The priority of these operators is less binding than the arithmetic ones, but more binding than the comparisons.

Parentheses may be used as above to force particular groupings.

Syntax

`~ a` (bitwise NOT)

where the argument *a* may be a further expression. It is converted to the nearest integer to machine precision and then the operation is performed.

See Also

`&, / #`

#

— Bitwise NON EQUIVALENCE operator.

Description

The bitwise operators perform operations of bitwise AND, bitwise OR, bitwise NOT and bitwise non-equivalence.

The priority of these operators is less binding than the arithmetic ones, but more binding than the comparisons.

Parentheses may be used as above to force particular groupings.

Syntax

`a # b` (bitwise NON EQUIVALENCE)

where the arguments *a* and *b* may be further expressions. They are converted to the nearest integer to machine precision and then the operation is performed.

See Also

`&`, `/`, `¬`

a

a — Converts a k-rate parameter to an a-rate value with interpolation.

Description

Converts a k-rate parameter to an a-rate value with interpolation.

Syntax

a(x) (control-rate args only)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

Examples

Here is an example of the a opcode. It uses the file *opa.csd* [examples/opa.csd].

Example 29. Example of the a opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o a.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create a sine wave at k-rate.
kwave oscil 20000, 440, 1

; Convert the k-rate sine wave to the audio-rate.
awave = a(kwave)

; Output the audio-rate version of sine wave.
out awave
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```


See Also

i, k

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

New in version 4.21

abetarand

abetarand — Deprecated.

Description

Deprecated as of version 3.49. Use the *betarand* opcode instead.

abexprnd

abexprnd — Deprecated.

Description

Deprecated as of version 3.49. Use the *bexprnd* opcode instead.

abs

abs — Returns an absolute value.

Description

Returns the absolute value of x .

Syntax

abs(x) (no rate restriction)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

Examples

Here is an example of the abs opcode. It uses the file *abs.csd* [examples/abs.csd].

Example 30. Example of the abs opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o abs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = -6
  i2 = abs(i1)

  print i2
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  i2 = 6.000
```

See Also

exp, frac, int, log, log10, i, sqrt

Credits

Example written by Kevin Conder.

acauchy

acauchy — Deprecated.

Description

Deprecated as of version 3.49. Use the *cauchy* opcode instead.

active

active — Returns the number of active instances of an instrument.

Description

Returns the number of active instances of an instrument.

Syntax

```
ir active insnum
```

```
kres active kinsnum
```

Initialization

insnum -- number of the instrument to be reported

Performance

kinsnum -- number of the instrument to be reported

active returns the number of active instances of instrument number *insnum*/*kinsnum*. As of Csound4.17 the output is updated at k-rate (if input arg is k-rate), to allow running count of instr instances.

Examples

Here is a simple example of the active opcode. It uses the file *active.csd* [examples/active.csd].

Example 31. Simple example of the active opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o active.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a noisy waveform.
instr 1
; Generate a really noisy waveform.
anoisy rand 44100
; Turn down its amplitude.
aoutput gain anoisy, 2500
; Send it to the output.
out aoutput
endin
```

```
; Instrument #2 - counts active instruments.
instr 2
; Count the active instances of Instrument #1.
icount active 1
; Print the number of active instances.
print icount
endin

</CsInstruments>
<CsScore>

; Start the first instance of Instrument #1 at 0:00 seconds.
i 1 0.0 3.0

; Start the second instance of Instrument #1 at 0:015 seconds.
i 1 1.5 1.5

; Play Instrument #2 at 0:01 seconds, when we have only
; one active instance of Instrument #1.
i 2 1.0 0.1

; Play Instrument #2 at 0:02 seconds, when we have
; two active instances of Instrument #1.
i 2 2.0 0.1
e

</CsScore>
</CsSoundSynthesizer>
```

Its output should include lines like this:

```
instr 2: icount = 1.000
instr 2: icount = 2.000
```

Here is a more advanced example of the active opcode. It displays the results of the active opcode at k-rate instead of i-rate. It uses the file *active_k.csd* [examples/active_k.csd].

Example 32. Example of the active opcode at k-rate.

```
<CsSoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out Audio in
-odac -iadc ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o active_k.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a noisy waveform.
instr 1
; Generate a really noisy waveform.
anoisy rand 44100
; Turn down its amplitude.
aoutput gain anoisy, 2500
; Send it to the output.
out aoutput
endin

; Instrument #2 - counts active instruments at k-rate.
instr 2
; Count the active instances of Instrument #1.
kcount active 1
; Print the number of active instances.
printk2 kcount
endin

</CsInstruments>
<CsScore>
```



```
; Start the first instance of Instrument #1 at 0:00 seconds.
i 1 0.0 3.0

; Start the second instance of Instrument #1 at 0:015 seconds.
i 1 1.5 1.5

; Play Instrument #2 at 0:01 seconds, when we have only
; one active instance of Instrument #1.
i 2 1.0 0.1

; Play Instrument #2 at 0:02 seconds, when we have
; two active instances of Instrument #1.
i 2 2.0 0.1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
i2      1.00000
i2      2.00000
```

Here is another example of the active opcode, using the number of instances to calculate gain. It uses the file *active_scale.csd* [examples/active_scale.csd].

Example 33. Example of the active opcode at k-rate.

```
<CsoundSynthesizer>
<CsOptions>

</CsOptions>
<CsInstruments>

sr= 44100
ksmps = 64
0dbfs = 1

;by Victor Lazzarini 2008

instr 1
kscal active 1
kamp port 1/kscal, 0.01
asig oscili kamp, p4, 1
kenv linseg 0, 0.1,1,p3-0.2,1,0.1, 0

      out asig*kenv
endin

</CsInstruments>
<CsScore>
f1 0 16384 10 1

i1 0 10 440
i1 1 3 220
i1 2 5 350
i1 4 3 700
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
July, 1999

Examples written by Kevin Conder.

New in Csound version 3.57

adsr

`adsr` — Calculates the classical ADSR envelope using linear segments.

Description

Calculates the classical ADSR envelope using linear segments.

Syntax

```
ares adsr iatt, idec, islev, irel [, idel]
```

```
kres adsr iatt, idec, islev, irel [, idel]
```

Initialization

iatt -- duration of attack phase

idec -- duration of decay

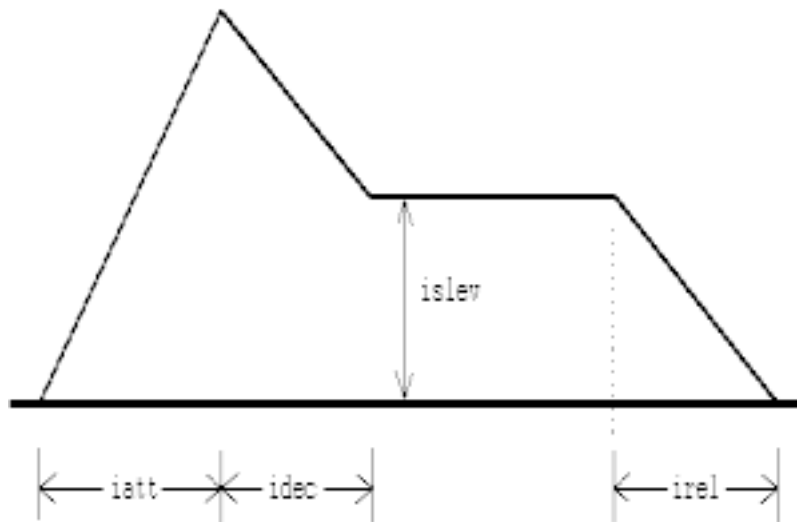
islev -- level for sustain phase

irel -- duration of release phase

idel -- period of zero before the envelope starts

Performance

The envelope is the range 0 to 1 and may need to be scaled further. The envelope may be described as:



Picture of an ADSR envelope.

The length of the sustain is calculated from the length of the note. This means *adsr* is not suitable for use with MIDI events. The opcode *madsr* uses the *linsegr* mechanism, and so can be used in MIDI applications.

adsr is new in Csound version 3.49.

Examples

Here is an example of the `adsr` opcode. It uses the file `adsr.csd` [examples/adsr.csd].

Example 34. Example of the `adsr` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o adsr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a simple instrument.
instr 1
; Set the amplitude.
kamp init 20000
; Get the frequency from the fourth p-field.
kcps = cpspch(p4)

al vco kamp, kcps, 1
out al
endin

; Instrument #2 - instrument with an ADSR envelope.
instr 2
iatt = 0.05
idec = 0.5
islev = 0.08
irel = 0.008

; Create an amplitude envelope.
kenv adsr iatt, idec, islev, irel
kamp = kenv * 20000

; Get the frequency from the fourth p-field.
kcps = cpspch(p4)

al vco kamp, kcps, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Set the tempo to 120 beats per minute.
t 0 120

; Play a melody with Instrument #1.
; p4 = frequency in pitch-class notation.
i 1 0 1 8.04
i 1 1 1 8.04
i 1 2 1 8.05
i 1 3 1 8.07
i 1 4 1 8.07
i 1 5 1 8.05
i 1 6 1 8.04
i 1 7 1 8.02
i 1 8 1 8.00
i 1 9 1 8.00
i 1 10 1 8.02
i 1 11 1 8.04
i 1 12 2 8.04
i 1 14 2 8.02
```

```
; Repeat the melody with Instrument #2.  
; p4 = frequency in pitch-class notation.  
i 2 16 1 8.04  
i 2 17 1 8.04  
i 2 18 1 8.05  
i 2 19 1 8.07  
i 2 20 1 8.07  
i 2 21 1 8.05  
i 2 22 1 8.04  
i 2 23 1 8.02  
i 2 24 1 8.00  
i 2 25 1 8.00  
i 2 26 1 8.02  
i 2 27 1 8.04  
i 2 28 2 8.04  
i 2 30 2 8.02  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

madsr, mxadsr, xadsr

Credits

Author: John ffitch

New in version 3.49

Example written by Kevin Conder.

adsyn

adsyn — Output is an additive set of individually controlled sinusoids, using an oscillator bank.

Description

Output is an additive set of individually controlled sinusoids, using an oscillator bank.

Syntax

```
ares adsyn kamod, kfmod, ksmod, ifilcod
```

Initialization

ifilcod -- integer or character-string denoting a control-file derived from analysis of an audio signal. An integer denotes the suffix of a file *adsyn.m* or *pvoc.m*; a character-string (in double quotes) gives a filename, optionally a full pathname. If not fullpath, the file is sought first in the current directory, then in the one given by the environment variable *SADIR* (if defined). *adsyn* control contains break-point amplitude- and frequency-envelope values organized for oscillator resynthesis, while *pvoc* control contains similar data organized for fft resynthesis. Memory usage depends on the size of the files involved, which are read and held entirely in memory during computation but are shared by multiple calls (see also *lpread*).

Performance

kamod -- amplitude factor of the contributing partials.

kfmod -- frequency factor of the contributing partials. It is a control-rate transposition factor: a value of 1 incurs no transposition, 1.5 transposes up a perfect fifth, and .5 down an octave.

ksmod -- speed factor of the contributing partials.

adsyn synthesizes complex time-varying timbres through the method of additive synthesis. Any number of sinusoids, each individually controlled in frequency and amplitude, can be summed by high-speed arithmetic to produce a high-fidelity result.

Component sinusoids are described by a control file describing amplitude and frequency tracks in millisecond breakpoint fashion. Tracks are defined by sequences of 16-bit binary integers:

```
-1, time, amp, time, amp,...  
-2, time, freq, time, freq,...
```

such as from hetrodyne filter analysis of an audio file. (For details see *hetro*.) The instantaneous amplitude and frequency values are used by an internal fixed-point oscillator that adds each active partial into an accumulated output signal. While there is a practical limit (limit removed in version 3.47) on the number of contributing partials, there is no restriction on their behavior over time. Any sound that can be described in terms of the behavior of sinusoids can be synthesized by *adsyn* alone.

Sound described by an *adsyn* control file can also be modified during re-synthesis. The signals *kamod*, *kfmod*, *ksmod* will modify the amplitude, frequency, and speed of contributing partials. These are multiplying factors, with *kfmod* modifying the frequency and *ksmod* modifying the *speed* with which the millisecond breakpoint line-segments are traversed. Thus .7, 1.5, and 2 will give rise to a softer sound, a perfect fifth higher, but only half as long. The values 1,1,1 will leave the sound unmodified. Each of these inputs can be a control signal.

Examples

Here is an example of the `adsyn` opcode. It uses the file `adsyn.csd` [examples/adsyn.csd], and `kickroll.het` [examples/kickroll.het]. The file “kickroll.het” was created by using the *hetro* utility with the audio file `kickroll.wav` [examples/kickroll.wav].

Example 35. Example of the `adsyn` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o adsyn.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; If the modulation amounts are set to 1, adsyn
; will not perform any special modulation.
kamod init 1
kfmod init 1
ksmod init 1

; Re-synthesizes the file "kickroll.het".
a1 adsyn kamod, kfmod, ksmod, "kickroll.het"

out a1 * 32768
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Example written by Kevin Conder.

adsynt

adsynt — Performs additive synthesis with an arbitrary number of partials, not necessarily harmonic.

Description

Performs additive synthesis with an arbitrary number of partials, not necessarily harmonic.

Syntax

```
ares adsynt kamp, kcps, iwfn, ifreqfn, iampfn, icnt [, iphs]
```

Initialization

iwfn -- table containing a waveform, usually a sine. Table values are not interpolated for performance reasons, so larger tables provide better quality.

ifreqfn -- table containing frequency values for each partial. *ifreqfn* may contain beginning frequency values for each partial, but is usually used for generating parameters at runtime with *tablew*. Frequencies must be relative to *kcps*. Size must be at least *icnt*.

iampfn -- table containing amplitude values for each partial. *iampfn* may contain beginning amplitude values for each partial, but is usually used for generating parameters at runtime with *tablew*. Amplitudes must be relative to *kamp*. Size must be at least *icnt*.

icnt -- number of partials to be generated

iphs -- initial phase of each oscillator, if *iphs* = -1, initialization is skipped. If *iphs* > 1, all phases will be initialized with a random value.

Performance

kamp -- amplitude of note

kcps -- base frequency of note. Partial frequencies will be relative to *kcps*.

Frequency and amplitude of each partial is given in the two tables provided. The purpose of this opcode is to have an instrument generate synthesis parameters at k-rate and write them to global parameter tables with the *tablew* opcode.

Examples

Here is an example of the adsynt opcode. It uses the file *adsynt.csd* [examples/adsynt.csd]. These two instruments perform additive synthesis. The output of each sounds like a Tibetan bowl. The first one is static, as parameters are only generated at init-time. In the second one, parameters are continuously changed.

Example 36. Example of the adsynt opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>
```



```
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o adsynt.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Generate a sinewave table.
giwave ftgen 1, 0, 1024, 10, 1
; Generate two empty tables for adsynt.
gifrqs ftgen 2, 0, 32, 7, 0, 32, 0
; A table for frequency and amp parameters.
giamps ftgen 3, 0, 32, 7, 0, 32, 0

; Generates parameters at init time
instr 1
; Generate 10 voices.
icnt = 10
; Init loop index.
index = 0

; Loop only executed at init time.
loop:
; Define non-harmonic partials.
ifreq pow index + 1, 1.5
; Define amplitudes.
iamp = 1 / (index+1)
; Write to tables.
tableiw ifreq, index, gifrqs
; Used by adsynt.
tableiw iamp, index, giamps

index = index + 1
; Do loop/
if (index < icnt) igoto loop

asig adsynt 5000, 150, giwave, gifrqs, giamps, icnt
out asig
endin

; Generates parameters every k-cycle.
instr 2
; Generate 10 voices.
icnt = 10
; Reset loop index.
kindex = 0

; Loop executed every k-cycle.
loop:
; Generate lfo for frequencies.
kspeed pow kindex + 1, 1.6
; Individual phase for each voice.
kphas phasorbnk kspeed * 0.7, kindex, icnt
klfo table kphas, giwave, 1
; Arbitrary parameter twiddling...
kdepth pow 1.4, kindex
kfreq pow kindex + 1, 1.5
kfreq = kfreq + klfo*0.006*kdepth

; Write freqs to table for adsynt.
tablew kfreq, kindex, gifrqs

; Generate lfo for amplitudes.
kspeed pow kindex + 1, 0.8
; Individual phase for each voice.
kphas phasorbnk kspeed*0.13, kindex, icnt, 2
klfo table kphas, giwave, 1
; Arbitrary parameter twiddling...
kamp pow 1 / (kindex + 1), 0.4
kamp = kamp * (0.3+0.35*(klfo+1))

; Write amps to table for adsynt.
tablew kamp, kindex, giamps

kindex = kindex + 1
; Do loop.
if (kindex < icnt) kgoto loop

asig adsynt 5000, 150, giwave, gifrqs, giamps, icnt
out asig
endin
```

```
</CsInstruments>
<CsScore>

; Play Instrument #1 for 2.5 seconds.
i 1 0 2.5
; Play Instrument #2 for 2.5 seconds.
i 2 3 2.5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Peter Neubäcker
Munich, Germany
August, 1999

New in Csound version 3.58

adsynt2

adsynt2 — Performs additive synthesis with an arbitrary number of partials -not necessarily harmonic- with interpolation.

Description

Performs additive synthesis with an arbitrary number of partials, not necessarily harmonic. (see *adsynt* for detailed manual)

Syntax

```
ar adsynt2 kamp, kcps, iwfn, ifreqfn, iampfn, icnt [, iphs]
```

Initialization

iwfn -- table containing a waveform, usually a sine. Table values are not interpolated for performance reasons, so larger tables provide better quality.

ifreqfn -- table containing frequency values for each partial. *ifreqfn* may contain beginning frequency values for each partial, but is usually used for generating parameters at runtime with *tablew*. Frequencies must be relative to *kcps*. Size must be at least *icnt*.

iampfn -- table containing amplitude values for each partial. *iampfn* may contain beginning amplitude values for each partial, but is usually used for generating parameters at runtime with *tablew*. Amplitudes must be relative to *kamp*. Size must be at least *icnt*.

icnt -- number of partials to be generated

iphs -- initial phase of each oscillator, if *iphs* = -1, initialization is skipped. If *iphs* > 1, all phases will be initialized with a random value.

Performance

kamp -- amplitude of note

kcps -- base frequency of note. Partial frequencies will be relative to *kcps*.

Frequency and amplitude of each partial is given in the two tables provided. The purpose of this opcode is to have an instrument generate synthesis parameters at k-rate and write them to global parameter tables with the *tablew* opcode.

adsynt2 is identical to *adsynt* (by Peter Neubäcker), except it provides linear interpolation for amplitude envelopes of each partial. It is a bit slower than *adsynt*, but interpolation highly improves sound quality in fast amplitude envelope transients when $kr < sr$ (i.e. when $ksmps > 1$). No interpolation is provided for pitch envelopes, since in this case sound quality degradation is not so evident even with high values of *ksmps*. It is not recommended when $kr = sr$, in this case *adsynt* is better (since it is faster).

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

aexprand

aexprand — Deprecated.

Description

Deprecated as of version 3.49. Use the *exprand* opcode instead.

aftouch

aftouch — Get the current after-touch value for this channel.

Description

Get the current after-touch value for this channel.

Syntax

```
kaft aftouch [imin] [, imax]
```

Initialization

imin (optional, default=0) -- minimum limit on values obtained.

imax (optional, default=127) -- maximum limit on values obtained.

Performance

Get the current after-touch value for this channel.

Examples

Here is an example of the aftouch opcode. It uses the file *aftouch.csd* [examples/aftouch.csd].

Example 37. Example of the aftouch opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d          -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o aftouch.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  k1 aftouch

  printk2 k1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 12 seconds.
i 1 0 12
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

ampmidi, cpsmidi, cpsmidib, midictrl, notnum, octmidi, octmidib, pchbend, pchmidi, pchmidib, veloc

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

agauss

agauss — Deprecated.

Description

Deprecated as of version 3.49. Use the *gauss* opcode instead.

agogobel

agogobel — Deprecated.

Description

New in version 3.47

Deprecated as of version 3.52. Use the *gogobel* opcode instead.

alinrand

alinrand — Deprecated.

Description

Deprecated as of version 3.49. Use the *linrand* opcode instead.

alpass

alpass — Reverberates an input signal with a flat frequency response.

Description

Reverberates an input signal with a flat frequency response.

Syntax

```
ares alpass asig, krvt, ilpt [, iskip] [, insmps]
```

Initialization

ilpt -- loop time in seconds, which determines the “echo density” of the reverberation. This in turn characterizes the “color” of the filter whose frequency response curve will contain $ilpt * sr/2$ peaks spaced evenly between 0 and $sr/2$ (the Nyquist frequency). Loop time can be as large as available memory will permit. The space required for an n second loop is $4n*sr$ bytes. The delay space is allocated and returned as in *delay*.

iskip (optional, default=0) -- initial disposition of delay-loop data space (cf. *reson*). The default value is 0.

insmps (optional, default=0) -- delay amount, as a number of samples.

Performance

krvt -- the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude).

This filter reiterates the input with an echo density determined by loop time *ilpt*. The attenuation rate is independent and is determined by *krvt*, the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude). Output will begin to appear immediately.

Examples

Here is an example of the alpass opcode. It uses the file *alpass.csd* [examples/alpass.csd].

Example 38. Example of the alpass opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o alpass.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
```

```
nchnls = 1

; Initialize the audio mixer.
gamix init 0

; Instrument #1.
instr 1
  ; Generate a source signal.
  a1 oscili 30000, cpspch(p4), 1
  ; Output the direct sound.
  out a1

  ; Add the source signal to the audio mixer.
  gamix = gamix + a1
endin

; Instrument #99 (highest instr number executed last)
instr 99
  krvt = 1.5
  ilpt = 0.1

  ; Filter the mixed signal.
  a99 alpass gamix, krvt, ilpt
  ; Output the result.
  out a99

  ; Empty the mixer for the next pass.
  gamix = 0
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 128 10 1

; p4 = frequency (in a pitch-class)
; Play Instrument #1 for a tenth of a second, p4=7.00
i 1 0 0.1 7.00
; Play Instrument #1 for a tenth of a second, p4=7.02
i 1 1 0.1 7.02
; Play Instrument #1 for a tenth of a second, p4=7.04
i 1 2 0.1 7.04
; Play Instrument #1 for a tenth of a second, p4=7.06
i 1 3 0.1 7.06

; Make sure the filter remains active.
i 99 0 5
e

</CsScore>
</CsoundSynthesizer>
```

See Also

comb, *reverb*, *valpass*, *vcomb*

Credits

Author: William “Pete” Moss (*vcomb* and *valpass*)
University of Texas at Austin
Austin, Texas USA
January 2002

Example written by Kevin Conder.

ampdb

ampdb — Returns the amplitude equivalent of the decibel value x.

Description

Returns the amplitude equivalent of the decibel value x. Thus:

- 60 dB = 1000
- 66 dB = 1995.262
- 72 dB = 3891.07
- 78 dB = 7943.279
- 84 dB = 15848.926
- 90 dB = 31622.764

Syntax

ampdb(x) (no rate restriction)

Examples

Here is an example of the ampdb opcode. It uses the file *ampdb.csd* [examples/ampdb.csd].

Example 39. Example of the ampdb opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o ampdb.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  idb = 90
  iamp = ampdb(idb)

  print iamp
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
```

```
i 1 0 1  
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  iamp = 31622.764
```

See Also

ampdbfs, db, dbamp, dbfsamp

Credits

Example written by Kevin Conder.

ampdbfs

ampdbfs — Returns the amplitude equivalent (in 16-bit signed integer scale) of the full scale decibel (dB FS) value x .

Description

Returns the amplitude equivalent of the full scale decibel (dB FS) value x . The logarithmic full scale decibel values will be converted to linear 16-bit signed integer values from #32,768 to +32,767.

Syntax

ampdbfs(x) (no rate restriction)

Examples

Here is an example of the **ampdbfs** opcode. It uses the file *ampdbfs.csd* [examples/ampdbfs.csd].

Example 40. Example of the **ampdbfs** opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ampdbfs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  idb = -1
  iamp = ampdbfs(idb)

  print iamp
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  iamp = 29203.621
```

See Also

ampdb, dbamp, dbfsamp, 0dbfs

Credits

Example written by Kevin Conder.

New is Csound version 4.10.

ampmidi

ampmidi — Get the velocity of the current MIDI event.

Description

Get the velocity of the current MIDI event.

Syntax

```
iamp ampmidi iscal [, ifn]
```

Initialization

iscal -- i-time scaling factor

ifn (optional, default=0) -- function table number of a normalized translation table, by which the incoming value is first interpreted. The default value is 0, denoting no translation.

Performance

Get the velocity of the current MIDI event, optionally pass it through a normalized translation table, and return an amplitude value in the range 0 - *iscal*.

Examples

Here is an example of the ampmidi opcode. It uses the file *ampmidi.csd* [examples/ampmidi.csd].

Example 41. Example of the ampmidi opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d          -M0    ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o ampmidi.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Scale the amplitude between 0 and 1.
; This example expects MIDI note inputs on channel 1
il ampmidi 1

print il
endin

</CsInstruments>
<CsScore>

;Dummy f-table to give time for real-time MIDI events
```



```
f 0 8000  
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

aftouch, cpsmidi, cpsmidib, midictrl, notnum, octmidi, octmidib, pchbend, pchmidi, pchmidib, veloc

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

apcauchy

apcauchy — Deprecated.

Description

Deprecated as of version 3.49. Use the *pcauchy* opcode instead.

apoisson

apoisson — Deprecated.

Description

Deprecated as of version 3.49. Use the *poisson* opcode instead.

apow

apow — Deprecated.

Description

Deprecated as of version 3.48. Use the *pow* opcode instead.

areson

areson — A notch filter whose transfer functions are the complements of the reson opcode.

Description

A notch filter whose transfer functions are the complements of the reson opcode.

Syntax

```
ares areson asig, kcf, kbw [, iscl] [, iskip]
```

Initialization

iscl (optional, default=0) -- coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than *kcf* are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. (This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise.) A zero value signifies no scaling of the signal, leaving that to some later adjustment (see *balance*). The default value is 0.

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

ares -- the output signal at audio rate.

asig -- the input signal at audio rate.

kcf -- the center frequency of the filter, or frequency position of the peak response.

kbw -- bandwidth of the filter (the Hz difference between the upper and lower half-power points).

areson is a filter whose transfer functions is the complement of *reson*. Thus *areson* is a notch filter whose transfer functions represents the “filtered out” aspects of their complements. However, power scaling is not normalized in *areson* but remains the true complement of the corresponding unit. Thus an audio signal, filtered by parallel matching *reson* and *areson* units, would under addition simply reconstruct the original spectrum.

This property is particularly useful for controlled mixing of different sources (see *lpreson*). Complex response curves such as those with multiple peaks can be obtained by using a bank of suitable filters in series. (The resultant response is the product of the component responses.) In such cases, the combined attenuation may result in a serious loss of signal power, but this can be regained by the use of *balance*.

Examples

Here is an example of the *areson* opcode. It uses the file *areson.csd* [examples/areson.csd].

Example 42. Example of the areson opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using com-

mand line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o areson.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1 - an unfiltered noise waveform.
instr 1
; Generate a white noise signal.
asig rand 20000

out asig
endin

; Instrument #2 - a filtered noise waveform.
instr 2
; Generate a white noise signal.
asig rand 20000

; Filter it using the areson opcode.
kcf init 1000
kbw init 100
afilt areson asig, kcf, kbw

; Clip the filtered signal's amplitude to 85 dB.
al clip afilt, 2, ampdb(85)
out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

aresonk, atone, atonek, port, portk, reson, resonk, tone, tonek

aresonk

aresonk — A notch filter whose transfer functions are the complements of the reson opcode.

Description

A notch filter whose transfer functions are the complements of the reson opcode.

Syntax

```
kres aresonk ksig, kcf, kbw [, iscl] [, iskip]
```

Initialization

iscl (optional, default=0) -- coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than kcf are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. (This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise.) A zero value signifies no scaling of the signal, leaving that to some later adjustment (see *balance*). The default value is 0.

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

kres -- the output signal at control-rate.

ksig -- the input signal at control-rate.

kcf -- the center frequency of the filter, or frequency position of the peak response.

kbw -- bandwidth of the filter (the Hz difference between the upper and lower half-power points).

aresonk is a filter whose transfer functions is the complement of *resonk*. Thus *aresonk* is a notch filter whose transfer functions represents the “filtered out” aspects of their complements. However, power scaling is not normalized in *aresonk* but remains the true complement of the corresponding unit.

See Also

areson, *atone*, *atonek*, *port*, *portk*, *reson*, *resonk*, *tone*, *tonek*

atone

atone — A hi-pass filter whose transfer functions are the complements of the *tone* opcode.

Description

A hi-pass filter whose transfer functions are the complements of the *tone* opcode.

Syntax

```
ares atone asig, khp [, iskip]
```

Initialization

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

ares -- the output signal at audio rate.

asig -- the input signal at audio rate.

khp -- the response curve's half-power point, in Hertz. Half power is defined as peak power / root 2.

atone is a filter whose transfer functions is the complement of *tone*. *atone* is thus a form of high-pass filter whose transfer functions represent the “filtered out” aspects of their complements. However, power scaling is not normalized in *atone* but remains the true complement of the corresponding unit. Thus an audio signal, filtered by parallel matching *tone* and *atone* units, would under addition simply reconstruct the original spectrum.

This property is particularly useful for controlled mixing of different sources (see *lpreson*). Complex response curves such as those with multiple peaks can be obtained by using a bank of suitable filters in series. (The resultant response is the product of the component responses.) In such cases, the combined attenuation may result in a serious loss of signal power, but this can be regained by the use of *balance*.

Examples

Here is an example of the *atone* opcode. It uses the file *atone.csd* [examples/atone.csd].

Example 43. Example of the *atone* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o atone.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```



```
; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1 - an unfiltered noise waveform.
instr 1
  ; Generate a white noise signal.
  asig rand 20000

  out asig
endin

; Instrument #2 - a filtered noise waveform.
instr 2
  ; Generate a white noise signal.
  asig rand 20000

  ; Filter it using the atone opcode.
  khp init 2000
  afilt atone asig, khp

  ; Clip the filtered signal's amplitude to 85 dB.
  al clip afilt, 2, ampdb(85)
  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

areson, aresonk, atonek, port, portk, reson, resonk, tone, tonek

atonek

atonek — A hi-pass filter whose transfer functions are the complements of the *tonek* opcode.

Description

A hi-pass filter whose transfer functions are the complements of the *tonek* opcode.

Syntax

```
kres atonek ksig, khp [, iskip]
```

Initialization

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

kres -- the output signal at control-rate.

ksig -- the input signal at control-rate.

khp -- the response curve's half-power point, in Hertz. Half power is defined as peak power / root 2.

atonek is a filter whose transfer functions is the complement of *tonek*. *atonek* is thus a form of high-pass filter whose transfer functions represent the “filtered out” aspects of their complements. However, power scaling is not normalized in *atonek* but remains the true complement of the corresponding unit.

See Also

areson, *aresonk*, *atone*, *port*, *portk*, *reson*, *resonk*, *tone*, *tonek*

Credits

Author: Robin Whittle
Australia
May 1997

atonex

atonex — Emulates a stack of filters using the atone opcode.

Description

atonex is equivalent to a filter consisting of more layers of *atone* with the same arguments, serially connected. Using a stack of a larger number of filters allows a sharper cutoff. They are faster than using a larger number instances in a Csound orchestra of the old opcodes, because only one initialization and k- cycle are needed at time and the audio loop falls entirely inside the cache memory of processor.

Syntax

```
ares atonex asig, khp [, inumlayer] [, iskip]
```

Initialization

inumlayer (optional) -- number of elements in the filter stack. Default value is 4.

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- input signal

khp -- the response curve's half-power point. Half power is defined as peak power / root 2.

See Also

resonx, *tonex*

Credits

Author: Gabriel Maldonado (adapted by John ffitch)
Italy

New in Csound version 3.49

atrirand

atrirand — Deprecated.

Description

Deprecated as of version 3.49. Use the *trirand* opcode instead.

ATSadd

ATSadd — uses the data from an ATS analysis file to perform additive synthesis.

Description

ATSadd reads from an ATS analysis file and uses the data to perform additive synthesis using an internal array of interpolating oscillators.

Syntax

```
ar ATSadd ktimepnt, kfmod, iatsfile, ifn, ipartials[, ipartialoffset, \
    ipartialincr, igatefn]
```

Initialization

iatsfile – the ATS number (n in ats.n) or the name in quotes of the analysis file made using *ATS* [<http://www-ccrma.stanford.edu/~juan/ATS.html>].

ifn – table number of a stored function containing a sine wave for *ATSadd* and a cosine for *ATSaddnz* (see examples below for more info)

ipartials – number of partials that will be used in the resynthesis (the noise has a maximum of 25 bands)

ipartialoffset (optional) – is the first partial used (defaults to 0).

ipartialincr (optional) – sets an increment by which these synthesis opcodes counts up from *ipartialoffset* for ibins components in the re-synthesis (defaults to 1).

igatefn (optional) – is the number of a stored function which will be applied to the amplitudes of the analysis bins before resynthesis takes place. If *igatefn* is greater than 0 the amplitudes of each bin will be scaled by *igatefn* through a simple mapping process. First, the amplitudes of all of the bins in all of the frames in the entire analysis file are compared to determine the maximum amplitude value. This value is then used create normalized amplitudes as indices into the stored function *igatefn*. The maximum amplitude will map to the last point in the function. An amplitude of 0 will map to the first point in the function. Values between 0 and 1 will map accordingly to points along the function table. See the examples below.

Performance

ktimepnt – The time pointer in seconds used to index the ATS file. Used for *ATSadd* exactly the same as for *pvoc*.

ATSadd and *ATSaddnz* are based on *pvadd* by Richard Karpen and use files created by Juan Pampin's *ATS (Analysis - Transformation - Synthesis)* [<http://www-ccrma.stanford.edu/~juan/ATS.html>].

kfmod – A control-rate transposition factor: a value of 1 incurs no transposition, 1.5 transposes up a perfect fifth, and .5 down an octave. Used for *ATSadd* exactly the same as for *pvoc*.

ATSadd reads from an ATS analysis file and uses the data to perform additive synthesis using an internal array of interpolating oscillators. The user supplies the wave table (usually one period of a sine wave), and can choose which analysis partials will be used in the re-synthesis.

Examples

```
ktime line 0, p3, 2.5
asig atsadd ktime, 1, "clarinet.ats", 1, 20, 2
```

In the example above, *ipartials* is 20 and *ipartialoffset* is 2. This will synthesize the 3rd thru 22nd partials in the "clarinet.ats" analysis file. *kmod* is 1 so there will be no pitch transformation. Since the *ktimepnt* envelope moves from 0 to 2.5 over the duration of the note, the analysis file will be read from 0 to 2.5 seconds of the original duration of the analysis over the duration of the csound note, this way we can change the duration independent of the pitch.

```
ktime line 0, p3, 2.5  
asig atsadd ktime, 1.0125, "clarinet.ats", 1, 20, 0, 2
```

In the above example we synthesize 20 partials as in example 1 except this time we're using a *ipartialoffset* of 0 and *ipartialincr* of 2, which means that we'll start from the first partial and synthesize 20 partials total, skipping every other one (ie. partial 1, 3, 5,...). We've also increased the pitch of the result (*kfmod* is set to 1.0125).

See also

ATSread, *ATSreadnz*, *ATSinfo*, *ATSbufread*, *ATScross*, *ATSinterpread*, *ATSpartialtap*, *ATSaddnz*, *ATSinnoi*

Credits

Author: Alex Norman
Seattle, Washington
2004

ATSaddnz

ATSaddnz — uses the data from an ATS analysis file to perform noise resynthesis.

Description

ATSaddnz reads from an ATS analysis file and uses the data to perform additive synthesis using a modified randi function.

Syntax

```
ar ATSaddnz ktimepnt, iatsfile, ibands[, ibandoffset, ibandincr]
```

Initialization

iatsfile – the ATS number (n in ats.n) or the name in quotes of the analysis file made using *ATS* [<http://www-ccrma.stanford.edu/~juan/ATS.html>].

ibands – number of noise bands that will be used in the resynthesis (the noise has a maximum of 25 bands)

ibandoffset (optional) – is the first noise band used (defaults to 0).

ibandincr (optional) – sets an increment by which these synthesis opcodes counts up from *ibandoffset* for ibins components in the re-synthesis (defaults to 1).

Performance

ktimepnt – The time pointer in seconds used to index the ATS file. Used for *ATSaddnz* exactly the same as for *pvoc* and *ATSadd*.

ATSaddnz and *ATSadd* are based on *pvadd* by Richard Karpen and use files created by Juan Pampin's *ATS (Analysis - Transformation - Synthesis)* [<http://www-ccrma.stanford.edu/~juan/ATS.html>].

ATSaddnz also reads from an ATS file but it resynthesizes the noise from noise energy data contained in the ATS file. It uses a modified randi function to create band limited noise and modulates that with a cosine wave, to synthesize a user specified selection of frequency bands. Modulating the noise is required to put the band limited noise in the correct place in the frequency spectrum.

Examples

```
ktime line 0, p3, 2.5
asig atsaddnz ktime, "clarinet.ats", 25
```

In the example above we're synthesizing all 25 noise bands from the data contained in the ATS analysis file called "clarinet.ats".

```
ktime line 2.5, p3, 0
asig atsaddnz ktime, 1, "clarinet.ats", 1, 24
```

Here we synthesize only the 25th noise band (*ibandoffset* of 24 and *ibands* of 1). Also our time pointer is going from 2.5 to 0 over the duration of the note so we're reading backwards from 2.5 seconds in the analysis file.

See also

ATSread, ATSreadnz, ATInfo, ATSbufread, ATScross, ATSinterpread, ATSpartialtap, ATSaddnz, ATSinnoi

Credits

Author: Alex Norman
Seattle, Washington
2004

ATSbufread

ATSbufread — reads data from and ATS data file and stores it in an internal data table of frequency, amplitude pairs.

Description

ATSbufread reads data from and ATS data file and stores it in an internal data table of frequency, amplitude pairs.

Syntax

```
ATSbufread ktimepnt, kfmod, iatsfile, ipartials[, ipartialoffset, \
            ipartialincr]
```

Initialization

iatsfile – the ATS number (n in ats.n) or the name in quotes of the analysis file made using *ATS* [<http://www-ccrma.stanford.edu/~juan/ATS.html>].

ipartials – number of partials that will be used in the resynthesis (the noise has a maximum of 25 bands)

ipartialoffset (optional) – is the first partial used (defaults to 0).

ipartialincr (optional) – sets an increment by which these synthesis opcodes counts up from *ipartialoffset* for ibins components in the re-synthesis (defaults to 1).

Performance

ktimepnt – The time pointer in seconds used to index the ATS file. Used for *ATSbufread* exactly the same as for *pvoc*.

kfmod – an input for performing pitch transposition or frequency modulation on all of the synthesized partials, if no fm or pitch change is desired then use a 1 for this value.

ATSbufread is based on *pvbufread* by Richard Karpen. *ATScross*, *ATSinterpread* and *ATSpartialtap* are all dependent on *ATSbufread* just as *pvcross* and *pvinterp* are on *pvbufread*. *ATSbufread* reads data from and ATS data file and stores it in an internal data table of frequency, amplitude pairs. The data stored by an *ATSbufread* can only be accessed by other unit generators, and therefore, due to the architecture of Csound, an *ATSbufread* must come before (but not necessarily directly) any dependent unit generator. Besides the fact that *ATSbufread* doesn't output any data directly, it works almost exactly as *ATSadd*. The ugen uses a time pointer (*ktimepnt*) to index the data in time, *ipartials*, *ipartialoffset* and *ipartialincr* to select which partials to store in the table and *kfmod* to scale partials in frequency.

Examples

See the examples for *ATScross*, *ATSinterpread* and *ATSpartialtap*

See also

ATSread, *ATSreadnz*, *ATSinfo*, *ATSinnoi*, *ATScross*, *ATSinterpread*, *ATSpartialtap*, *ATSadd*, *ATSaddnz*

Credits

Author: Alex Norman
Seattle, Washington
2004

ATScross

ATScross — perform cross synthesis from ATS analysis files.

Description

ATScross uses data from an ATS analysis file and data from an *ATSbufread* to perform cross synthesis.

Syntax

```
ar ATScross ktimepnt, kfmod, iatsfile, ifn, kmylev, kbuflev, ipartials \
    [, ipartialoffset, ipartialincr]
```

Initialization

iatsfile – integer or character-string denoting a control-file derived from ATS analysis of an audio signal. An integer denotes the suffix of a file ATS.m; a character-string (in double quotes) gives a filename, optionally a full pathname. If not full-path, the file is sought first in the current directory, then in the one given by the environment variable SADIR (if defined).

ifn – table number of a stored function containing a sine wave.

ipartials – number of partials that will be used in the resynthesis

ipartialoffset (optional) – is the first partial used (defaults to 0).

ipartialincr (optional) – sets an increment by which these synthesis opcodes counts up from *ipartialoffset* for ibins components in the re-synthesis (defaults to 1).

Performance

ktimepnt – The time pointer in seconds used to index the ATS file. Used for *ATScross* exactly the same as for *pvoc*.

kfmod – an input for performing pitch transposition or frequency modulation on all of the synthesized partials, if no fm or pitch change is desired then use a 1 for this value.

kmylev - scales the *ATScross* component of the frequency spectrum applied to the partials from the ATS file indicated by the *atscross* opcode. The frequency spectrum information comes from the *atscross* ATS file. A value of 1 (and 0 for *kbuflev*) gives the same results as *ATSadd*.

kbuflev - scales the *ATSbufread* component of the frequency spectrum applied to the partials from the ATS file indicated by the *ATScross* opcode. The frequency spectrum information comes from the *ATSbufread* ATS file. A value of 1 (and 0 for *kmylev*) results in partials that have frequency information from the ATS file given by the *ATScross*, but amplitudes imposed by data from the ATS file given by *ATSbufread*.

ATScross uses data from an ATS analysis file (indicated by *iatsfile*) and data from an *ATSbufread* to perform cross synthesis. *ATScross* uses *ktimepnt*, *kfmod*, *ipartials*, *ipartialoffset* and *ipartialincr* just like *ATSadd*. *ATScross* synthesizes a sine-wave for each partial selected by the user and uses the frequency of that partial (after scaling in frequency by *kfmod*) to index the table created by *ATSbufread*. Interpolation is used to get in-between values. *ATScross* uses the sum of the amplitude data from its ATS file (scaled by *kmylev*) and the amplitude data gained from an *ATSbufread* (scaled by *kbuflev*) to scale the amplitude of each partial it synthesizes. Setting *kmylev* to one and *kbuflev* to zero will make *ATScross* act exactly like *ATSadd*. Setting *kmylev* to zero and *kbuflev* to one will produce a sound that has all the partials selected by the *ATScross* ugen, but with amplitudes taken from an *ATSbufread*. The time pointers of the *ATSbufread* and *ATScross* do not need to be the same.

Examples

```
ktime line 0, p3, 2.4
ktime2 line 0, p3, .5
kline expseg 0.001, .9, 1, p3-.9, 1
kline2 expseg .001, p3, 1
atsbufread ktime2, 1, "crt.ats", 20
aout atscross ktime, 1, "cl.ats", 1, kline, .001* (1 - kline2), 42
```

This example performs cross synthesis using two ATS files, "crt.ats" and "cl.ats". The result of this will be a sound that starts out with the shape (in frequency) of crt.ats, and ends with the shape of cl.ats. All the sine-wave frequencies come from cl.ats. The *kbuflev* value is scaled because the energy produced by applying crt.ats's frequency spectrum to cl.ats's partials is very large. Notice also that the time pointers of the atsbufread (crt.ats) and atscross (cl.ats) need not have the same value, this way you can read through the two ATS files at different rates.

See also

ATSread, ATSreadnz, ATSinfo, ATSSinnoi, ATSbufread, ATSinterpread, ATSpartialtap, ATSadd, AT-Saddnz

Credits

Author: Alex Norman
Seattle, Washington
2004

ATSinfo

ATSinfo — reads data out of the header of an ATS file.

Description

atsinfo reads data out of the header of an ATS file.

Syntax

```
idata ATSinfo iatsfile, ilocation
```

Initialization

iatsfile – the ATS number (n in ats.n) or the name in quotes of the analysis file made using *ATS* [<http://www-ccrma.stanford.edu/~juan/ATS.html>].

ilocation – indicates which location in the header file to return. The data in the header gives information about the data contained in the rest of the ATS file. The possible values for *ilocation* are given in the following list:

- 0 - Sample rate (Hz)
- 1 - Frame Size (samples)
- 2 - Window Size (samples)
- 3 - Number of Partial
- 4 - Number of Frames
- 5 - Maximum Amplitude
- 6 - Maximum Frequency (Hz)
- 7 - Duration (seconds)
- 8 - ATS file Type

Performance

Macros can really improve the legibility of your csound code, I've provided my Macro Definitions below:

```
#define ATS_SAMP_RATE #0#  
#define ATS_FRAME_SZ #1#  
#define ATS_WIN_SZ #2#  
#define ATS_N_PARTIALS #3#  
#define ATS_N_FRAMES #4#  
#define ATS_AMP_MAX #5#  
#define ATS_FREQ_MAX #6#  
#define ATS_DUR #7#  
#define ATS_TYPE #8#
```

ATSinfo can be useful for writing generic instruments that will work with many ATS files, even if they have different lengths and different numbers of partials etc. Example 2 is a simple application of this.

Examples

1.

```
imax_freq atsinfo "cl.ats", $ATS_FREQ_MAX
```

In the example above we get the maximum frequency value from the ATS file "cl.ats" and store it in `imax_freq`. We use at Csound Macro (defined above) `$ATS_FREQ_MAX`, which is equivalent to the number 6.

2.

```
i_npartials atsinfo p4, $ATS_N_PARTIALS  
i_dur      atsinfo p4, $ATS_DUR  
ktimepnt line 0, p3, i_dur  
aout      atsadd ktimepnt, 1, p4, 1, i_npartials
```

In the example above we use *ATSinfo* to retrieve the duration and number of partials in the ATS file indicated by `p4`. With this info we synthesize the partials using `atsadd`. Since the duration and number of partials are not "hard-coded" we can use this code with any ats file.

See also

ATSread, *ATSreadnz*, *ATSbufread*, *ATScross*, *ATSinterpread*, *ATSpartialtap*, *ATSadd*, *ATSaddnz*, *ATSinnoi*

Credits

Author: Alex Norman
Seattle, Washington
2004

ATSinterpread

ATSinterpread — allows a user to determine the frequency envelope of any *ATSbufread*.

Description

ATSinterpread allows a user to determine the frequency envelope of any *ATSbufread*.

Syntax

```
kamp ATSinterpread kfreq
```

Performance

kfreq - a frequency value (given in Hertz) used by *ATSinterpread* as in index into the table produced by an *ATSbufread*.

ATSinterpread takes a frequency value (*kfreq* in Hz). This frequency is used to index the data of an *ATSbufread*. The return value is an amplitude gained from the *ATSbufread* after interpolation. *ATSinterpread* allows a user to determine the frequency envelope of any *ATSbufread*. This data could be useful for a number of reasons, one might be performing cross synthesis of data from an ATS file and non ATS data.

Examples

```
ktime line 0, p3, 2.4
atsbufread ktime, 1, "cl.ats", 42
kamp atsinterpread p4
aosc oscili kamp, p4, 1
```

This example shows how to use *ATSinterpread*. Here a frequency is given by the score (*p4*) and this frequency is given to an *ATSinterpread* (with a corresponding *ATSbufread*). The *ATSinterpread* uses this frequency to output a corresponding amplitude value, based on the *atsfile* given by the *ATSbufread* (*cl.ats* in this case). We then use that amplitude to scale a sine-wave that is synthesized with the same frequency (*p4*). You could extend this to include multiple sine-waves. This way you could synthesize any reasonable frequency (within the low and high frequencies of the indicated ATS file), and maintain the shape (in frequency) of the indicated *atsfile* (given by the *ATSbufread*).

See also

ATSread, *ATSreadnz*, *ATSinfo*, *ATSsinnoi*, *ATSbufread*, *ATScross*, *ATSpartialtap*, *ATSadd*, *ATSaddnz*

Credits

Author: Alex Norman
Seattle, Washington
2004

ATSread

ATSread — reads data from an ATS file.

Description

ATSread returns the amplitude (*kamp*) and frequency (*kfreq*) information of a user specified partial contained in the ATS analysis file at the time indicated by the time pointer *ktimepnt*.

Syntax

```
kfreq, kamp ATSread ktimepnt, iatsfile, ipartial
```

Initialization

iatsfile – the ATS number (n in ats.n) or the name in quotes of the analysis file made using *ATS* [<http://www-ccrma.stanford.edu/~juan/ATS.html>].

ipartial – the number of the analysis partial to return the frequency in Hz and amplitude.

Performance

kfreq, *kamp* - outputs of the *ATSread* unit. These values represent the frequency and amplitude of a specific partial selected by the user using *ipartial*. The partials' informations are derived from an ATS analysis. *ATSread* linearly interpolates the frequency and amplitude between frames in the ATS analysis file at k-rate. The output is dependent on the data in the analysis file and the pointer *ktimepnt*.

ktimepnt – The time pointer in seconds used to index the ATS file. Used for *ATSread* exactly the same as for *pvoc* and *ATSadd*.

Examples

```
ktime line 0, p3, 2.5  
kfreq, kamp atsread ktime, "clarinet.ats", 2  
aout oscili 1000000 * kamp, kfreq, 1
```

Here we're using *ATSread* to get the 2nd partial's frequency and amplitude data out of the 'clarinet.ats' ATS analysis file. We're using that data to drive an oscillator, but we could use it for anything else that can take a k-rate input, like the bandwidth and resonance of a filter etc.

See also

ATSreadnz, *ATSinfo*, *ATSbufread*, *ATScross*, *ATSinterpread*, *ATSpartialtap*, *ATSadd*, *ATSaddnz*, *ATSsinnoi*

Credits

Author: Alex Norman
Seattle, Washington
2004

Description

ATSreadnz returns the energy (*kenergy*) of a user specified noise band (1-25 bands) at the time indicated by the time pointer *ktimepnt*.

kenenergy **ATSreadnz** ktimepnt, iatsfile, iband

ibands – the number of the noise band to return the energy data.

An ATS analysis differs from a pvanal in that ATS tracks the partials and computes the noise energy of the sound being analyzed. For more info about ATS analysis read Juan Pampin's description on the the *ATS* [<http://www-ccrma.stanford.edu/~juan/ATS.html>] web-page.

Function table 2 used in the oscillator is a cosine, which is needed to shift the band limited noise into the correct place in the frequency spectrum. The randi function creates a band of noise centered about 0 Hz that has a bandwidth of about 110 Hz; multiplying it by a cosine will shift it to be centered at 455 Hz, which is the center frequency of the 5th critical noise band. This is only an example, for synthesizing the noise you'd be better off just using *ATSaddnz* unless you want to use

your own noise synthesis algorithm. Maybe you could use the noise energy for something else like applying a small amount of jitter to specific partials or for controlling something totally unrelated to the source sound?

See also

ATSread, *ATSinfo*, *ATSbufread*, *ATScross*, *ATSinterpread*, *ATSpartialtap*, *ATSadd*, *ATSaddnz*, *ATSSinnoi*

Credits

Author: Alex Norman
Seattle, Washington
2004

ATSpartialtap

ATSpartialtap — returns a frequency, amplitude pair from an *ATSbufread* opcode.

Description

ATSpartialtap takes a partial number and returns a frequency, amplitude pair. The frequency and amplitude data comes from an *atsbufread* *ATSbufread* opcode.

Syntax

```
kfrq, kamp ATSpartialtap ipartialnum
```

Initialization

ipartialnum - indicates the partial that the *ATSpartialtap* opcode should read from an *ATSbufread*.

Performance

kfrq - returns the frequency value for the requested partial.

kamp - returns the amplitude value for the requested partial.

ATSpartialtap takes a partial number and returns a frequency, amplitude pair. The frequency and amplitude data comes from an *ATSbufread* opcode. This is more restricted version of *ATSread*, since each *ATSread* opcode has its own independent time pointer, and *ATSpartialtap* is restricted to the data given by an *ATSbufread*. Its simplicity is its attractive feature.

Examples

```
ktime line 0, p3, 2.4
atsbufread ktime, 1, "crt.ats", 20
kfreq1, kamp1 atspartialtap 1
kfreq2, kamp2 atspartialtap 10
kfreq3, kamp3 atspartialtap 20
```

This example here uses an *ATSpartialtap*, and an *ATSbufread* to read partials 1, 10 and 20 from 'crt.ats'. These amplitudes and frequencies could be used to re-synthesize those partials, or something all together different.

See also

ATSread, *ATSreadnz*, *ATSinfo*, *ATSinnoi*, *ATSbufread*, *ATScross*, *ATSinterpread*, *ATSadd*, *ATSaddnz*

Credits

Author: Alex Norman
Seattle, Washington
2004

ATSSinnoi

ATSSinnoi — uses the data from an ATS analysis file to perform resynthesis.

Description

ATSSinnoi reads data from an ATS data file and uses the information to synthesize sines and noise together.

Syntax

```
ar ATSSinnoi ktimepnt, ksinlev, knzlev, kfmod, iatsfile, ipartials \  
    [, ipartialoffset, ipartialincr]
```

Initialization

iatsfile – the ATS number (n in ats.n) or the name in quotes of the analysis file made using *ATS* [<http://www-ccrma.stanford.edu/~juan/ATS.html>].

ipartials – number of partials that will be used in the resynthesis (the noise has a maximum of 25 bands)

ipartialoffset (optional) – is the first partial used (defaults to 0).

ipartialincr (optional) – sets an increment by which these synthesis opcodes counts up from *ipartialoffset* for ibins components in the re-synthesis (defaults to 1).

Performance

ktimepnt – The time pointer in seconds used to index the ATS file. Used for *ATSSinnoi* exactly the same as for *pvoc*.

ksinlev - controls the level of the sines in the *ATSSinnoi* ugen. A value of 1 gives full volume sine-waves.

knzlev - controls the level of the noise components in the *ATSSinnoi* ugen. A value of 1 gives full volume noise.

kfmod – an input for performing pitch transposition or frequency modulation on all of the synthesized partials, if no fm or pitch change is desired then use a 1 for this value.

ATSSinnoi reads data from an ATS data file and uses the information to synthesize sines and noise together. The noise energy for each band is distributed equally among each partial that falls in that band. Each partial is then synthesized, along with that partial's noise component. Each noise component is then modulated by the corresponding partial to be put in the correct place in the frequency spectrum. The level of the noise and the partials are individually controllable. See the *ATS* [<http://www-ccrma.stanford.edu/~juan/ATS.html>] webpage for more info about the sinnoi synthesis. An ATS analysis differs from a pvanal in that ATS tracks the partials and computes the noise energy of the sound being analyzed. For more info about ATS analysis read Juan Pampin's description on the the *ATS* [<http://www-ccrma.stanford.edu/~juan/ATS.html>] web-page.

Examples

```
ktime line 0, p3, 2.5  
asig atssinnoi ktime, 1, 1, 1, "clarinet.ats", 42
```

Here we synthesize both the noise and the sinewaves (all 42 partials) contained in "clarinet.ats" together. The relative volumes of the noise and the partials are unaltered (each set to 1).

```
ktime line 0, p3, 2.5
knzfade expon 0.001, p3, 2.5
asig atssinnoi ktime, 1, knzfade, 1, "clarinet.ats", 42
```

This example here is like example 5 except that we use an envelope to control *knzlev* (the noise level). The result of this will be a clarinet sound that has its noise component fade in over the duration of the note.

See also

ATSread, ATSreadnz, ATSinfo, ATShufread, ATScross, ATSinterpread, ATSpartialtap, ATSadd, AT-Saddnz

Credits

Author: Alex Norman
Seattle, Washington
2004

aunirand

aunirand — Deprecated.

Description

Deprecated as of version 3.49. Use the *unirand* opcode instead.

aweibull

aweibull — Deprecated.

Description

Deprecated as of version 3.49. Use the *weibull* opcode instead.

babo

babo — A physical model reverberator.

Description

babo stands for *ball-within-the-box*. It is a physical model reverberator based on the paper by Davide Rocchesso "The Ball within the Box: a sound-processing metaphor", Computer Music Journal, Vol 19, N.4, pp.45-47, Winter 1995.

The resonator geometry can be defined, along with some response characteristics, the position of the listener within the resonator, and the position of the sound source.

Syntax

```
a1, a2 babo asig, ksrcx, ksrcy, ksrcz, irx, iry, irz [, idiff] [, ifno]
```

Initialization

irx, iry, irz -- the coordinates of the geometry of the resonator (length of the edges in meters)

idiff -- is the coefficient of diffusion at the walls, which regulates the amount of diffusion (0-1, where 0 = no diffusion, 1 = maximum diffusion - default: 1)

ifno -- expert values function: a function number that holds all the additional parameters of the resonator. This is typically a GEN2--type function used in non-rescaling mode. They are as follows:

- *decay* -- main decay of the resonator (default: 0.99)
- *hydecay* -- high frequency decay of the resonator (default: 0.1)
- *rcvx, rcvy, rcvz* -- the coordinates of the position of the receiver (the listener) (in meters; 0,0,0 is the resonator center)
- *rdistance* -- the distance in meters between the two pickups (your ears, for example - default: 0.3)
- *direct* -- the attenuation of the direct signal (0-1, default: 0.5)
- *early_diff* -- the attenuation coefficient of the early reflections (0-1, default: 0.8)

Performance

asig -- the input signal

ksrcx, ksrcy, ksrcz -- the virtual coordinates of the source of sound (the input signal). These are allowed to move at k-rate and provide all the necessary variations in terms of response of the resonator.

Examples

Here is a simple example of the babo opcode. It uses the file *babo.csd* [examples/babo.csd], and *beats.wav* [examples/beats.wav].

Example 44. A simple example of the babo opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o babo.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Nicola Bernardini */
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; minimal babo instrument
;
instr 1
    ix      = p4 ; x position of source
    iy      = p5 ; y position of source
    iz      = p6 ; z position of source
    ixsize  = p7 ; width of the resonator
    iysize  = p8 ; depth of the resonator
    izsize  = p9 ; height of the resonator

    ainput soundin "beats.wav"

    al,ar    babo      ainput*0.7, ix, iy, iz, ixsize, iysize, izsize

    outs     al,ar

endin

</CsInstruments>
<CsScore>

/* Written by Nicola Bernardini */
; simple babo usage:
;
;p4      : x position of source
;p5      : y position of source
;p6      : z position of source
;p7      : width of the resonator
;p8      : depth of the resonator
;p9      : height of the resonator
;
i 1 0 10 6 4 3      14.39 11.86 10
;          ^^^^^^      ^^^^^^^^^^^^^
;          |||/|||/      ++++++::: optimal room dims according to
;          |||/|||/      Milner and Bernard JASA 85(2), 1989
;          |||/|||/      ++++++::: source position
e

</CsScore>
</CsoundSynthesizer>
```

Here is an advanced example of the babo opcode. It uses the file *babo_expert.csd* [examples/babo_expert.csd], and *beats.wav* [examples/beats.wav].

Example 45. An advanced example of the babo opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o babo_expert.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```

```

/* Written by Nicola Bernardini */
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; full blown babo instrument with movement
;
instr 2
ixstart = p4 ; start x position of source (left-right)
ixend = p7 ; end x position of source
iystart = p5 ; start y position of source (front-back)
iyend = p8 ; end y position of source
izstart = p6 ; start z position of source (up-down)
izend = p9 ; end z position of source
ixsize = p10 ; width of the resonator
iysize = p11 ; depth of the resonator
izsize = p12 ; height of the resonator
idiff = p13 ; diffusion coefficient
iexpert = p14 ; power user values stored in this function

ainput soundin "beats.wav"
ksource_x line ixstart, p3, ixend
ksource_y line iystart, p3, iyend
ksource_z line izstart, p3, izend

al,ar babo ainput*0.7, ksource_x, ksource_y, ksource_z, ixsize, iysize, izsize, idiff, iexpert

outs al,ar

endin

</CsInstruments>
<CsScore>

/* Written by Nicola Bernardini */
; full blown instrument
;p4 : start x position of source (left-right)
;p5 : end x position of source
;p6 : start y position of source (front-back)
;p7 : end y position of source
;p8 : start z position of source (up-down)
;p9 : end z position of source
;p10 : width of the resonator
;p11 : depth of the resonator
;p12 : height of the resonator
;p13 : diffusion coefficient
;p14 : power user values stored in this function

;
; decay hidecay rx ry rz rdistance direct early_diff
f1 0 8 -2 0.95 0.95 0 0 0 0.3 0.5 0.8 ; brighter
f2 0 8 -2 0.95 0.5 0 0 0 0.3 0.5 0.8 ; default (to be set as)
f3 0 8 -2 0.95 0.01 0 0 0 0.3 0.5 0.8 ; darker
f4 0 8 -2 0.95 0.7 0 0 0 0.3 0.1 0.4 ; to hear the effect of diffusion
f5 0 8 -2 0.9 0.5 0 0 0 0.3 2.0 0.98 ; to hear the movement
f6 0 8 -2 0.99 0.1 0 0 0 0.3 0.5 0.8 ; default vals
;
; ----- gen. number: negative to avoid rescaling

i2 0 10 6 4 3 6 4 3 14.39 11.86 10 1 6 ; defaults
i2 + 4 6 4 3 6 4 3 14.39 11.86 10 1 1 ; hear brightness 1
i2 + 4 6 4 3 -6 -4 3 14.39 11.86 10 1 2 ; hear brightness 2
i2 + 4 6 4 3 -6 -4 3 14.39 11.86 10 1 3 ; hear brightness 3
i2 + 3 .6 .4 .3 -.6 -.4 .3 1.439 1.186 1.0 0.0 4 ; hear diffusion 1
i2 + 3 .6 .4 .3 -.6 -.4 .3 1.439 1.186 1.0 1.0 4 ; hear diffusion 2
i2 + 4 12 4 3 -12 -4 -3 24.39 21.86 20 1 5 ; hear movement
;
i2 + 4 6 4 3 6 4 3 14.39 11.86 10 1 1 ; hear brightness 1
i2 + 4 6 4 3 -6 -4 3 14.39 11.86 10 1 2 ; hear brightness 2
i2 + 4 6 4 3 -6 -4 3 14.39 11.86 10 1 3 ; hear brightness 3
i2 + 3 .6 .4 .3 -.6 -.4 .3 1.439 1.186 1.0 0.0 4 ; hear diffusion 1
i2 + 3 .6 .4 .3 -.6 -.4 .3 1.439 1.186 1.0 1.0 4 ; hear diffusion 2
i2 + 4 12 4 3 -12 -4 -3 24.39 21.86 20 1 5 ; hear movement
;
; //////////////////////////////////////// | --: expert values function
; //////////////////////////////////////// +--: diffusion
; //////////////////////////////////////// -----: optimal room dims according to Milner and Bernard JA
;
; -----: source position start and end
e

</CsScore>
</CsoundSynthesizer>

```

Credits

Author: Paolo Filippi
Padova, Italy
1999

Nicola Bernardini
Rome, Italy
2000

New in Csound version 4.09

balance

balance — Adjust one audio signal according to the values of another.

Description

The rms power of *asig* can be interrogated, set, or adjusted to match that of a comparator signal.

Syntax

```
ares balance asig, acomp [, ihp] [, iskip]
```

Initialization

ihp (optional) -- half-power point (in Hz) of a special internal low-pass filter. The default value is 10.

iskip (optional, default=0) -- initial disposition of internal data space (see *reson*). The default value is 0.

Performance

asig -- input audio signal

acomp -- the comparator signal

balance outputs a version of *asig*, amplitude-modified so that its rms power is equal to that of a comparator signal *acomp*. Thus a signal that has suffered loss of power (eg., in passing through a filter bank) can be restored by matching it with, for instance, its own source. It should be noted that *gain* and *balance* provide amplitude modification only - output signals are not altered in any other respect.

Examples

Here is an example of the *balance* opcode. It uses the file *balance.csd* [examples/balance.csd].

Example 46. Example of the *balance* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o balance.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a band-limited pulse train.
```

```
asrc buzz 30000, 440, sr/440, 1

; Send the source signal through 2 filters.
a1 reson asrc, 1000, 100
a2 reson a1, 3000, 500

; Balance the filtered signal with the source.
afin balance a2, asrc

out afin
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

gain, rms

bamboo

bamboo — Semi-physical model of a bamboo sound.

Description

bamboo is a semi-physical model of a bamboo sound. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares bamboo kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] \  
      [, ifreq1] [, ifreq2]
```

Initialization

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 1.25.

idamp (optional) -- the damping factor, as part of this equation:

$\text{damping_amount} = 0.9999 + (\text{idamp} * 0.002)$

The default *damping_amount* is 0.9999 which means that the default value of *idamp* is 0. The maximum *damping_amount* is 1.0 (no damping). This means the maximum value for *idamp* is 0.05.

The recommended range for *idamp* is usually below 75% of the maximum value.

imaxshake (optional, default=0) -- amount of energy to add back into the system. The value should be in range 0 to 1.

ifreq (optional) -- the main resonant frequency. The default value is 2800.

ifreq1 (optional) -- the first resonant frequency. The default value is 2240.

ifreq2 (optional) -- the second resonant frequency. The default value is 3360.

Performance

kamp -- Amplitude of output. Note: As these instruments are stochastic, this is only an approximation.

Examples

Here is an example of the bamboo opcode. It uses the file *bamboo.csd* [examples/bamboo.csd].

Example 47. Example of the bamboo opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>
```

```
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-o dac      -i adc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o bamboo.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 01 ;example of bamboo
a1 bamboo p4, 0.01
  out a1
endin

</CsInstruments>
<CsScore>

i1 0 1 20000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

dripwater, guiro, sleighbells, tambourine

Credits

Author: Perry Cook, part of the PhISEM (Physically Informed Stochastic Event Modeling)
Adapted by John fitch
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

barmodel

barmodel — Creates a tone similar to a struck metal bar.

Description

Audio output is a tone similar to a struck metal bar, using a physical model developed from solving the partial differential equation. There are controls over the boundary conditions as well as the bar characteristics.

Syntax

```
ares barmodel kbcL, kbcR, iK, ib, kscan, iT30, ipos, ivel, iwid
```

Initialization

iK -- dimensionless stiffness parameter. If this parameter is negative then the initialisation is skipped and the previous state of the bar is continued.

ib -- high-frequency loss parameter (keep this small).

iT30 -- 30 db decay time in seconds.

ipos -- position along the bar that the strike occurs.

ivel -- normalized strike velocity.

iwid -- spatial width of strike.

Performance

A note is played on a metallic bar, with the arguments as below.

kbcL -- Boundary condition at left end of bar (1 is clamped, 2 pivoting and 3 free).

kbcR -- Boundary condition at right end of bar (1 is clamped, 2 pivoting and 3 free).

kscan -- Speed of scanning the output location.

Note that changing the boundary conditions during playing may lead to glitches and is made available as an experiment. The use of a non-zero *kscan* can give apparent re-introduction of sound due to modulation.

Examples

Here is an example of the **barmodel** opcode. It uses the file *barmodel.csd* [examples/barmodel.csd].

Example 48. Example of the barmodel opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
```



```
-odac          -iadc      -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o barmodel.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr          =          44100
kr          =          4410
ksmps      =          10
nchnls     =          1

; Instrument #1.
instr 1
  aq          barmodel      1, 1, p4, 0.001, 0.23, 5, p5, p6, p7
                    out          aq
endin

</CsInstruments>
<CsScore>

i1 0.0 0.5  3 0.2 500  0.05
i1 0.5 0.5 -3 0.3 1000 0.05
i1 1.0 0.5 -3 0.4 1000 0.1
i1 1.5 4.0 -3 0.5 800  0.05
e
/* barmodel */

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Stefan Bilbao
University of Edinburgh, UK
Author: John fitch
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 5.01

bbcutm

bbcutm — Generates breakbeat-style cut-ups of a mono audio stream.

Description

The BreakBeat Cutter automatically generates cut-ups of a source audio stream in the style of drum and bass/jungle breakbeat manipulations. There are two versions, for mono (*bbcutm*) or stereo (*bbcuts*) sources. Whilst originally based on breakbeat cutting, the opcode can be applied to any type of source audio.

The prototypical cut sequence favoured over one bar with eighth note subdivisions would be

3+ 3R + 2

where we take a 3 unit block from the source's start, repeat it, then 2 units from the 7th and 8th eighth notes of the source.

We talk of rendering phrases (a sequence of cuts before reaching a new phrase at the beginning of a bar) and units (as subdivision th notes).

The opcode comes most alive when multiple synchronised versions are used simultaneously.

Syntax

```
a1 bbcutm asource, ibps, isubdiv, ibarlength, iphrasebars, inumrepeats \  
    [, istutterspeed] [, istutterchance] [, ienvchoice ]
```

Initialization

ibps -- Tempo to cut at, in beats per second.

isubdiv -- Subdivisions unit, for a bar. So 8 is eighth notes (of a 4/4 bar).

ibarlength -- How many beats per bar. Set to 4 for default 4/4 bar behaviour.

iphrasebars -- The output cuts are generated in phrases, each phrase is up to iphrasebars long

inumrepeats -- In normal use the algorithm would allow up to one additional repeat of a given cut at a time. This parameter allows that to be changed. Value 1 is normal- up to one extra repeat. 0 would avoid repeating, and you would always get back the original source except for enveloping and stuttering.

istutterspeed -- (optional, default=1) The stutter can be an integer multiple of the subdivision speed. For instance, if subdiv is 8 (quavers) and stutterspeed is 2, then the stutter is in semiquavers (sixteenth notes= subdiv 16). The default is 1.

istutterchance -- (optional, default=0) The tail of a phrase has this chance of becoming a single repeating one unit cell stutter (0.0 to 1.0). The default is 0.

ienvchoice -- (optional, default=1) choose 1 for on (exponential envelope for cut grains) or 0 for off. Off will cause clicking, but may give good noisy results, especially for percussive sources. The default is 1, on.

Performance

asource -- The audio signal to be cut up. This version runs in real-time without knowledge of future audio.

Examples

Here is a simple example of the *bbcutm* opcode. It uses the file *bbcutm.csd* [examples/bbcutm.csd], and *beats.wav* [examples/beats.wav].

Example 49. A simple example of the *bbcutm* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o bbcutm.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - Play an audio file normally.
instr 1
  asource soundin "beats.wav"
  out asource
endin

; Instrument #2 - Cut-up an audio file.
instr 2
  asource soundin "beats.wav"

  ibps = 4
  isubdiv = 8
  ibarlength = 4
  iphrasebars = 1
  inumrepeats = 2

  al bbcutm asource, ibps, isubdiv, ibarlength, iphrasebars, inumrepeats

  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 3 2
e

</CsScore>
</CsoundSynthesizer>
```

Here are some more advanced examples...

Example 50. First steps- mono and stereo versions

```
<CsoundSynthesizer>
<CsInstruments>
```

```
sr      =      44100
kr      =      4410
ksmps   =      10
nchnls  =      2

instr 1
  asource diskin "break7.wav",1,0,1 ; a source breakbeat sample, wraparound lest it stop!

  ; cuts in eighth notes per 4/4 bar, up to 4 bar phrases, up to 1
  ; repeat in total (standard use) rare stuttering at 16 note speed,
  ; no enveloping
  asig bbcutm asource, 2.6937, 8, 4, 4, 1, 2, 0.1, 0

  outs      asig,asig
endin

instr 2 ;stereo version
  asource1,asource2 diskin "break7stereo.wav", 1, 0, 1 ; a source breakbeat sample, wraparound

  ; cuts in eighth notes per 4/4 bar, up to 4 bar phrases, up to 1
  ; repeat in total (standard use) rare stuttering at 16 note speed,
  ; no enveloping
  asig1,asig2 bbcuts asource1, asource2, 2.6937, 8, 4, 4, 1, 2, 0.1, 0

  outs      asig1,asig2
endin

</CsInstruments>
<CsScore>
i1 0 10
i2 11 10
e
</CsScore>
</CsoundSynthesizer>
```

Example 51. Multiple simultaneous synchronised breaks

```
<CsoundSynthesizer>
<CsInstruments>
sr      =      44100
kr      =      4410
ksmps   =      10
nchnls  =      2

instr 1
  ibps    = 2.6937
  iplaybackspeed = ibps/p5
  asource diskin p4, iplaybackspeed, 0, 1

  asig bbcutm asource, 2.6937, p6, 4, 4, p7, 2, 0.1, 1

  out      asig
endin

</CsInstruments>
<CsScore>

; source      bps cut repeats
i1 0 10 "break1.wav" 2.3 8 2 //2.3 is the source original tempo
i1 0 10 "break2.wav" 2.4 8 3
i1 0 10 "break3.wav" 2.5 16 4
e
</CsScore>
</CsoundSynthesizer>
```

Example 52. Cutting up any old audio- much more interesting noises than this should be possible!

```
<CsoundSynthesizer>
<CsInstruments>
sr      =      44100
```

```
kr          =          4410
ksmps      =          10
nchnls     =          2

instr 1
  asource oscil 20000, 70, 1
  ; ain, bps, subdiv, barlength, phrasebars, numrepeats,
  ; stutterspeed, stutterchance, envelopingon
  asig bbcutm asource, 2, 32, 1, 1, 2, 4, 0.6, 1
  outs asig
endin

</CsInstruments>
<CsScore>
f1 0 256 10 1
i1 0 10
e
</CsScore>
</CsoundSynthesizer>
```

Example 53. Constant stuttering- faked, not possible since can only stutter in last half bar could make extra stuttering option parameter

```
<CsoundSynthesizer>
<CsInstruments>
sr          =          44100
kr          =          4410
ksmps      =          10
nchnls     =          2

instr 1
  asource diskin "break7.wav", 1, 0, 1

  ;16th note cuts- but cut size 2 over half a beat.
  ;each half beat will either survive intact or be turned into
  ;the first sixteenth played twice in succession

  asig bbcutm asource, 2.6937, 2, 0.5, 1, 2, 2, 1.0, 0
  outs asig
endin

</CsInstruments>
<CsScore>
i1 0 30
e
</CsScore>
</CsoundSynthesizer>
```

See Also

bbcuts

Credits

Author: Nick Collins
London
August 2001

New in version 4.13

bbcuts

bbcuts — Generates breakbeat-style cut-ups of a stereo audio stream.

Description

The BreakBeat Cutter automatically generates cut-ups of a source audio stream in the style of drum and bass/jungle breakbeat manipulations. There are two versions, for mono (*bbcutm*) or stereo (*bbcuts*) sources. Whilst originally based on breakbeat cutting, the opcode can be applied to any type of source audio.

The prototypical cut sequence favoured over one bar with eighth note subdivisions would be

3+ 3R + 2

where we take a 3 unit block from the source's start, repeat it, then 2 units from the 7th and 8th eighth notes of the source.

We talk of rendering phrases (a sequence of cuts before reaching a new phrase at the beginning of a bar) and units (as subdivision th notes).

The opcode comes most alive when multiple synchronised versions are used simultaneously.

Syntax

```
a1,a2 bbcuts asource1, asource2, ibps, isubdiv, ibarlength, iphrasebars, \  
inumrepeats [, istutterspeed] [, istutterchance] [, ienvchoice]
```

Initialization

ibps -- Tempo to cut at, in beats per second.

isubdiv -- Subdivisions unit, for a bar. So 8 is eighth notes (of a 4/4 bar).

ibarlength -- How many beats per bar. Set to 4 for default 4/4 bar behaviour.

iphrasebars -- The output cuts are generated in phrases, each phrase is up to iphrasebars long

inumrepeats -- In normal use the algorithm would allow up to one additional repeat of a given cut at a time. This parameter allows that to be changed. Value 1 is normal- up to one extra repeat. 0 would avoid repeating, and you would always get back the original source except for enveloping and stuttering.

istutterspeed -- (optional, default=1) The stutter can be an integer multiple of the subdivision speed. For instance, if subdiv is 8 (quavers) and stutterspeed is 2, then the stutter is in semiquavers (sixteenth notes= subdiv 16). The default is 1.

istutterchance -- (optional, default=0) The tail of a phrase has this chance of becoming a single repeating one unit cell stutter (0.0 to 1.0). The default is 0.

ienvchoice -- (optional, default=1) choose 1 for on (exponential envelope for cut grains) or 0 for off. Off will cause clicking, but may give good noisy results, especially for percussive sources. The default is 1, on.

Performance

asource -- The audio signal to be cut up. This version runs in real-time without knowledge of future audio.

Examples

See the advanced examples for the *bbcutm* opcode.

See Also

bbcutm

Credits

Author: Nick Collins
London
August 2001

New in version 4.13

betarand

betarand — Beta distribution random number generator (positive values only).

Description

Beta distribution random number generator (positive values only). This is an x-class noise generator.

Syntax

ares **betarand** krange, kalpha, kbeta

ires **betarand** krange, kalpha, kbeta

kres **betarand** krange, kalpha, kbeta

Performance

krange -- range of the random numbers (0 - *krange*).

kalpha -- alpha value. If *kalpha* is smaller than one, smaller values favor values near 0.

kbeta -- beta value. If *kbeta* is smaller than one, smaller values favor values near *krange*.

If both *kalpha* and *kbeta* equal one we have uniform distribution. If both *kalpha* and *kbeta* are greater than one we have a sort of Gaussian distribution. Outputs only positive numbers.

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the betarand opcode. It uses the file *betarand.csd* [examples/betarand.csd].

Example 54. Example of the betarand opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o betarand.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
```



```
nchnls = 1

; Instrument #1.
instr 1
; Generate a number between 0 and 1 with a
; uniform distribution.
; krange = 1
; kalpha = 1
; kbeta = 1

i1 betarand 1, 1, 1

print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like:

```
instr 1: i1 = 24583.412
```

See Also

seed, bexprnd, cauchy, exprand, gauss, linrand, pcauchy, poisson, trirand, unirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Existed in 3.30

Example written by Kevin Conder.

bexprnd

bexprnd — Exponential distribution random number generator.

Description

Exponential distribution random number generator. This is an x-class noise generator.

Syntax

ares **bexprnd** krange

ires **bexprnd** krange

kres **bexprnd** krange

Performance

krange -- the range of the random numbers (*-krange* to *+krange*)

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the bexprnd opcode. It uses the file *bexprnd.csd* [examples/bexprnd.csd].

Example 55. Example of the bexprnd opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac        -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o bexprnd.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number between -1 and 1.
; krange = 1

il bexprnd 1
```

```
    print i1
  endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like:

```
instr 1:  i1 = 1.141
```

See Also

seed, betarand, cauchy, exprand, gauss, linrand, pcauchy, poisson, trirand, unirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder.

bformenc

bformenc — Deprecated. Codes a signal into the ambisonic B format.

Description

Codes a signal into the ambisonic B format. Note that this opcode is deprecated as it is inaccurate, and is replaced by the much better opcode *bformenc1* which replicates all the important features.

Syntax

```
aw, ax, ay, az bformenc asig, kalpha, kbeta, kord0, kord1
```

```
aw, ax, ay, az, ar, as, at, au, av bformenc asig, kalpha, kbeta, \  
kord0, kord1, kord2
```

```
aw, ax, ay, az, ar, as, at, au, av, ak, al, am, an, ao, ap, aq bformenc \  
asig, kalpha, kbeta, kord0, kord1, kord2, kord3
```

Performance

aw, ax, ay, ... -- output cells of the B format.

asig -- input signal.

kalpha — azimuth angle in degrees (clockwise).

kbeta -- altitude angle in degrees.

kord0 -- linear gain of the zero order B format.

kord1 -- linear gain of the first order B format.

kord2 -- linear gain of the second order B format.

kord3 -- linear gain of the third order B format.

Example

Here is an example of the bformenc opcode. It uses the file *bformenc.csd* [examples/bformenc.csd].

Example 56. Example of the bformenc opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  No messages  
;-odac      -iadc      -d      ;;RT audio I/O  
; For Non-realtime output leave only the line below:  
-o bformenc.wav -W ;; for file output any platform  
</CsOptions>  
<CsInstruments>  
sr = 44100  
kr = 4410  
ksmps = 10  
nchnls = 8
```

```
instr 1
  ; generate pink noise
  anoise pinkish 1000

  ; two full turns
  kalpha line 0, p3, 720
  kbeta = 0

  ; fade ambisonic order from 2nd to 0th during second turn
  kord0 = 1
  kord1 linseg 1, p3 / 2, 1, p3 / 2, 0
  kord2 linseg 1, p3 / 2, 1, p3 / 2, 0

  ; generate B format
  aw, ax, ay, az, ar, as, at, au, av bformenc anoise, kalpha, kbeta, kord0, kord1, kord2

  ; decode B format for 8 channel circle loudspeaker setup
  a1, a2, a3, a4, a5, a6, a7, a8 bformdec 4, aw, ax, ay, az, ar, as, at, au, av

  ; write audio out
  outo a1, a2, a3, a4, a5, a6, a7, a8
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 20 seconds.
i 1 0 20
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Samuel Groner
2005

New in version 5.07. Deprecated in 5.09.

bformenc1

bformenc1 — Codes a signal into the ambisonic B format.

Description

Codes a signal into the ambisonic B format

Syntax

```
aw, ax, ay, az bformenc1 asig, kalpha, kbeta
```

```
aw, ax, ay, az, ar, as, at, au, av bformenc1 asig, kalpha, kbeta
```

```
aw, ax, ay, az, ar, as, at, au, av, ak, al, am, an, ao, ap, aq bformenc1 \  
    asig, kalpha, kbeta
```

Performance

aw, ax, ay, ... -- output cells of the B format.

asig -- input signal.

kalpha — azimuth angle in degrees (anticlockwise).

kbeta -- altitude angle in degrees.

Example

Here is an example of the bformenc1 opcode. It uses the file *bformenc1.csd* [examples/bformenc1.csd].

Example 57. Example of the bformenc1 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  No messages  
;-odac      -iadc      -d      ;;;RT audio I/O  
; For Non-realtime ouput leave only the line below:  
-o bformenc.wav -W ;;; for file output any platform  
</CsOptions>  
<CsInstruments>  
sr = 44100  
kr = 4410  
ksmps = 10  
nchnls = 8  
  
instr 1  
    ; generate pink noise  
    anoise pinkish 1000  
  
    ; two full turns  
    kalpha line 0, p3, 720  
    kbeta = 0  
  
    ; fade ambisonic order from 2nd to 0th during second turn  
    kord0 = 1  
    kord1 linseg 1, p3 / 2, 1, p3 / 2, 0
```

```
kord2 linseg 1, p3 / 2, 1, p3 / 2, 0

; generate B format
aw, ax, ay, az, ar, as, at, au, av bformenc1 anoise, kalpha, kbeta, kord0, kord1, kord2

; decode B format for 8 channel circle loudspeaker setup
a1, a2, a3, a4, a5, a6, a7, a8 bformdec1 4, aw, ax, ay, az, ar, as, at, au, av

; write audio out
outo a1, a2, a3, a4, a5, a6, a7, a8

endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 20 seconds.
i 1 0 20
e

</CsScore>
</CsoundSynthesizer>
```

See Also

bformdec1

Credits

Author: Richard Furse, Bruce Wiggins and Fons Adriaensen, following code by Samuel Groner 2008

New in version 5.09

bformdec

bformdec — Deprecated. Decodes an ambisonic B format signal.

Description

Decodes an ambisonic B format signal into loudspeaker specific signals. Note that this opcode is deprecated as it is inaccurate, and is replaced by the much better opcode *bformdec1* which replicates all the important features.

Syntax

```
ao1, ao2 bformdec isetup, aw, ax, ay, az [, ar, as, at, au, av \
    [, abk, al, am, an, ao, ap, aq]]

ao1, ao2, ao3, ao4 bformdec isetup, aw, ax, ay, az [, ar, as, at, \
    au, av [, abk, al, am, an, ao, ap, aq]]

ao1, ao2, ao3, ao4, ao5 bformdec isetup, aw, ax, ay, az [, ar, as, \
    at, au, av [, abk, al, am, an, ao, ap, aq]]

ao1, ao2, ao3, ao4, ao5, ao6, ao7, ao8 bformdec isetup, aw, ax, ay, az \
    [, ar, as, at, au, av [, abk, al, am, an, ao, ap, aq]]]
```

Initialization

isetup — loudspeaker setup. There are five supported setups: 1 denotes stereo setup. There must be two output cells with loudspeaker positions assumed to be (330/0, 30/0).

2 denotes quad setup. There must be four output cells. Loudspeaker positions assumed to be (45°/0), (135°/0), (225/0), (315/0).

3 is a 5.1 surround setup. There must be five output cells. LFE channel is not supported. Loudspeaker positions assumed to be (330/0), (30/0), (0/0), (250/0), (110/0).

4 denotes eight loudspeaker circle setup. There must be eight output cells. Loudspeaker positions assumed to be (22.5/0), (67.5/0), (112.5/0), (157.5/0), (202.5/0), (247.5/0), (292.5/0), (337.5/0).

5 means an eight loudspeaker cubic setup. There must be eight output cells. Loudspeaker positions assumed to be (45/0), (45/30), (135/0), (135/30), (225/0), (225/30), (315/0), (315/30).

Performance

aw, ax, ay, ... -- input signal in the B format.

ao1 .. ao8 — loudspeaker specific output signals.

Example

Here is an example of the bformdec opcode. It uses the file *bformenc.csd* [examples/bformenc.csd].

Example 58. Example of the bformdec opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.


```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
;-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
-o bformenc.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr = 44100
kr = 4410
ksmps = 10
nchnls = 8

instr 1
; generate pink noise
anoise pinkish 1000

; two full turns
kalpha line 0, p3, 720
kbeta = 0

; fade ambisonic order from 2nd to 0th during second turn
kord0 = 1
kord1 linseg 1, p3 / 2, 1, p3 / 2, 0
kord2 linseg 1, p3 / 2, 1, p3 / 2, 0

; generate B format
aw, ax, ay, az, ar, as, at, au, av bformenc anoise, kalpha, kbeta, kord0, kord1, kord2

; decode B format for 8 channel circle loudspeaker setup
a1, a2, a3, a4, a5, a6, a7, a8 bformdec 4, aw, ax, ay, az, ar, as, at, au, av

; write audio out
outo a1, a2, a3, a4, a5, a6, a7, a8
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 20 seconds.
i 1 0 20
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Samuel Groner
2005

New in version 5.07. Deprecated in 5.09

bformdec1

bformdec1 — Decodes an ambisonic B format signal

Description

Decodes an ambisonic B format signal into loudspeaker specific signals.

Syntax

```
ao1, ao2 bformdec1 isetup, aw, ax, ay, az [, ar, as, at, au, av \  
[, abk, al, am, an, ao, ap, aq]]
```

```
ao1, ao2, ao3, ao4 bformdec1 isetup, aw, ax, ay, az [, ar, as, at, \  
au, av [, abk, al, am, an, ao, ap, aq]]
```

```
ao1, ao2, ao3, ao4, ao5 bformdec1 isetup, aw, ax, ay, az [, ar, as, \  
at, au, av [, abk, al, am, an, ao, ap, aq]]
```

```
ao1, ao2, ao3, ao4, ao5, ao6, ao7, ao8 bformdec1 isetup, aw, ax, ay, az \  
[, ar, as, at, au, av [, abk, al, am, an, ao, ap, aq]]]
```

Initialization

Note that horizontal angles are measured anticlockwise in this description.

isetup — loudspeaker setup. There are five supported setups:

- 1. Stereo - L(90), R(-90); this is an M+S style stereo decode.
- 2. Quad - FL(45), BL(135), BR(-135), FR(-45). This is a first-order 'in-phase' decode.
- 3. 5.0 - L(30),R(-30),C(0),BL(110),BR(-110). Note that many people do not actually use the angles above for their speaker arrays and a good decode for DVD etc can be achieved using the Quad configuration to feed L, R, BL and BR (leaving C silent).
- 4.

Octagon

-

FFL(22.5),FLL(67.5),BLL(112.5),BBL(157.5),BBR(-157.5),BRR(-112.5),FRR(-67.5),FFR(-22.5
) . This is a first-, second- or third-order 'in-phase' decode, depending on the number of input
channels.
- 5.

Cube

-

FLD(45,-35.26),FLU(45,35.26),BLD(135,-35.26),BLU(135,35.26),BRD(-135,-35.26),BRU(-135,
35.26),FRD(-45,-35.26),FRU(-45,35.26). This is a first-order 'in-phase' decode.

Performance

aw, ax, ay, ... -- input signal in the B format.

ao1 .. ao8 — loudspeaker specific output signals.

Example

Here is an example of the bformdec1 opcode. It uses the file *bformenc1.csd* [examples/bformenc1.csd].

Example 59. Example of the bformdec1 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
;-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
-o bformenc.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr = 44100
kr = 4410
ksmps = 10
nchnls = 8

instr 1
; generate pink noise
anoise pinkish 1000

; two full turns
kalpha line 0, p3, 720
kbeta = 0

; fade ambisonic order from 2nd to 0th during second turn
kord0 = 1
kord1 linseg 1, p3 / 2, 1, p3 / 2, 0
kord2 linseg 1, p3 / 2, 1, p3 / 2, 0

; generate B format
aw, ax, ay, az, ar, as, at, au, av bformenc1 anoise, kalpha, kbeta, kord0, kord1, kord2

; decode B format for 8 channel circle loudspeaker setup
a1, a2, a3, a4, a5, a6, a7, a8 bformdec1 4, aw, ax, ay, az, ar, as, at, au, av

; write audio out
outo a1, a2, a3, a4, a5, a6, a7, a8
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 20 seconds.
i 1 0 20
e

</CsScore>
</CsoundSynthesizer>
```

See Also

bformenc1

Credits

Author: Richard Furse, Bruce Wiggins and Fons Adriaensen, following code by Samuel Groner 2008

New in version 5.09

binit

binit — PVS tracks to amplitude+frequency conversion.

Description

The binit opcode takes an input containing a TRACKS pv streaming signal (as generated, for instance by partials) and converts it into a equal-bandwidth bin-frame containing amplitude and frequency pairs (PVS_AMP_FREQ), suitable for overlap-add resynthesis (such as performed by pvsynth) or further PVS streaming phase vocoder signal transformations. For each frequency bin, it will look for a suitable track signal to fill it; if not found, the bin will be empty (0 amplitude). If more than one track fits a certain bin, the one with highest amplitude will be chosen. This means that not all of the input signal is actually 'binned', the operation is lossy. However, in many situations this loss is not perceptually relevant.

Syntax

```
fsig binit fin, isize
```

Performance

fsig -- output pv stream in PVS_AMP_FREQ format

fin -- input pv stream in TRACKS format

isize -- FFT size of output (N).

Examples

Example 60. Example

```
ain inch 1 ; input signal
fs1,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fs1,fsi2,.003,1,3,500 ; partial tracking
fbins binit fst, 2048 ; convert it back to bins
aout pvsynth fbins ; overlap-add resynthesis
out aout
```

The example above shows partial tracking of an ifd-analysis signal, conversion to bin frames and overlap-add resynthesis.

Credits

Author: Victor Lazzarini;
February 2006

New in Csound5.01

biquad

biquad — A sweepable general purpose biquadratic digital filter.

Description

A sweepable general purpose biquadratic digital filter.

Syntax

```
ares biquad asig, kb0, kb1, kb2, ka0, ka1, ka2 [, iskip]
```

Initialization

iskip (optional, default=0) -- if non-zero, initialization will be skipped. Default value 0. (New in Csound version 3.50)

Performance

asig -- input signal

biquad is a general purpose biquadratic digital filter of the form:

$$a0*y(n) + a1*y[n-1] + a2*y[n-2] = b0*x[n] + b1*x[n-1] + b2*x[n-2]$$

This filter has the following frequency response:

$$H(Z) = \frac{B(Z)}{A(Z)} = \frac{b0 + b1*Z^{-1} + b2*Z^{-2}}{a0 + a1*Z^{-1} + a2*Z^{-2}}$$

This type of filter is often encountered in digital signal processing literature. It allows six user-defined k-rate coefficients.

Examples

Here is an example of the biquad opcode. It uses the file *biquad.csd* [examples/biquad.csd].

Example 61. Example of the biquad opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o biquad.wav -W ;; for file output any platform
</CsOptions>
```

```
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Instrument #1.
instr 1
; Get the values from the score.
idur = p3
iamp = p4
icps = cpspch(p5)
kfco = p6
krez = p7

; Calculate the biquadratic filter's coefficients
kfcon = 2*3.14159265*kfco/sr
kalpha = 1-2*krez*cos(kfcon)*cos(kfcon)+krez*krez*cos(2*kfcon)
kbeta = krez*krez*sin(2*kfcon)-2*krez*cos(kfcon)*sin(kfcon)
kgama = 1+cos(kfcon)
km1 = kalpha*kgama+kbeta*sin(kfcon)
km2 = kalpha*kgama-kbeta*sin(kfcon)
kden = sqrt(km1*km1+km2*km2)
kb0 = 1.5*(kalpha*kalpha+kbeta*kbeta)/kden
kb1 = kb0
kb2 = 0
ka0 = 1
ka1 = -2*krez*cos(kfcon)
ka2 = krez*krez

; Generate an input signal.
axn vco 1, icps, 1

; Filter the input signal.
ayn biquad axn, kb0, kb1, kb2, ka0, ka1, ka2
outs ayn*iamp/2, ayn*iamp/2
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Sta Dur Amp Pitch Fco Rez
i 1 0.0 1.0 20000 6.00 1000 .8
i 1 1.0 1.0 20000 6.03 2000 .95
e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the biquad opcode used for modal synthesis. It uses the file *biquad-2.csd* [examples/biquad-2.csd]. See the *Modal Frequency Ratios* appendix for other frequency ratios.

Example 62. Example of the biquad opcode for modal synthesis.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out Audio in
-odac -iadc ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o biquad-2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

/* modal synthesis using biquad filters as oscillators
Example by Scott Lindroth 2007 */

instr 1
```

```
ipi = 3.1415926
idenom = sr*0.5

ipulseSpd = p4
icps      = p5
ipan      = p6
iamp      = p7
iModes    = p8

apulse    mpulse iamp, 0

icps      = cspch( icps )

; filter gain

iamp1 = 600
iamp2 = 1000
iamp3 = 1000
iamp4 = 1000
iamp5 = 1000
iamp6 = 1000

; resonance

irpole1 = 0.99999
irpole2 = irpole12
irpole3 = irpole13
irpole4 = irpole14
irpole5 = irpole15
irpole6 = irpole16

; modal frequencies

if (iModes == 1) goto modes1
if (iModes == 2) goto modes2

modes1:
if1      = icps * 1           ;pot lid
if2      = icps * 6.27
if3      = icps * 3.2
if4      = icps * 9.92
if5      = icps * 14.15
if6      = icps * 6.23
goto nextPart

modes2:
if1      = icps * 1           ;uniform wood bar
if2      = icps * 2.572
if3      = icps * 4.644
if4      = icps * 6.984
if5      = icps * 9.723
if6      = icps * 12.0
goto nextPart

nextPart:

; convert frequency to radian frequency

itheta1 = (if1/idenom) * ipi
itheta2 = (if2/idenom) * ipi
itheta3 = (if3/idenom) * ipi
itheta4 = (if4/idenom) * ipi
itheta5 = (if5/idenom) * ipi
itheta6 = (if6/idenom) * ipi

; calculate coefficients

ib11 = -2 * irpole1 * cos(itheta1)
ib21 = irpole1 * irpole1
ib12 = -2 * irpole2 * cos(itheta2)
ib22 = irpole2 * irpole2
ib13 = -2 * irpole3 * cos(itheta3)
ib23 = irpole3 * irpole3
ib14 = -2 * irpole4 * cos(itheta4)
ib24 = irpole4 * irpole4
ib15 = -2 * irpole5 * cos(itheta5)
ib25 = irpole5 * irpole5
ib16 = -2 * irpole6 * cos(itheta6)
ib26 = irpole6 * irpole6

;printk 1, ib 11
;printk 1, ib 21

; also try setting the -1 coeff. to 0, but be sure to scale down the amplitude!

asin1    biquad apulse * iamp1, 1, 0, -1, 1, ib11, ib21
asin2    biquad apulse * iamp2, 1, 0, -1, 1, ib12, ib22
```

```
asin3      biquad  apulse * iamp3, 1, 0, -1, 1, ib13, ib23
asin4      biquad  apulse * iamp4, 1, 0, -1, 1, ib14, ib24
asin5      biquad  apulse * iamp5, 1, 0, -1, 1, ib15, ib25
asin6      biquad  apulse * iamp6, 1, 0, -1, 1, ib16, ib26

afin      =      (asin1 + asin2 + asin3 + asin4 + asin5 + asin6)

outs      afin * sqrt(p6), afin*sqrt(1-p6)

endin
</CsInstruments>
<CsScore>
;ins      st      dur      pulseSpd      pch      pan      amp      Modes
i1        0      12      0      7.089      0      0.7      2
i1        .      .      .      7.09      1      .      .
i1        .      .      .      7.091      0.5      .      .

i1        0      12      0      8.039      0      0.7      2
i1        0      12      0      8.04      1      0.7      2
i1        0      12      0      8.041      0.5      0.7      2

i1        9      .      .      7.089      0      .      2
i1        .      .      .      7.09      1      .      .
i1        .      .      .      7.091      0.5      .      .

i1        9      12      0      8.019      0      0.7      2
i1        9      12      0      8.02      1      0.7      2
i1        9      12      0      8.021      0.5      0.7      2
e
</CsScore>
</CsoundSynthesizer>
```

See Also

biquada, moogvcf, rezzy

Credits

Author: Hans Mikelson
October 1998

New in Csound version 3.49

biquada

biquada — A sweepable general purpose biquadratic digital filter with a-rate parameters.

Description

A sweepable general purpose biquadratic digital filter.

Syntax

```
ares biquada asig, ab0, ab1, ab2, aa0, aa1, aa2 [, iskip]
```

Initialization

iskip (optional, default=0) -- if non-zero, initialization will be skipped. Default value 0. (New in Csound version 3.50)

Performance

asig -- input signal

biquada is a general purpose biquadratic digital filter of the form:

$$a_0*y(n) + a_1*y[n-1] + a_2*y[n-2] = b_0*x[n] + b_1*x[n-1] + b_2*x[n-2]$$

This filter has the following frequency response:

$$H(Z) = \frac{B(Z)}{A(Z)} = \frac{b_0 + b_1*Z^{-1} + b_2*Z^{-2}}{a_0 + a_1*Z^{-1} + a_2*Z^{-2}}$$

This type of filter is often encountered in digital signal processing literature. It allows six user-defined a-rate coefficients.

See Also

biquad

Credits

Author: Hans Mikelson
October 1998

New in Csound version 3.49

birnd

birnd — Returns a random number in a bi-polar range.

Description

Returns a random number in a bi-polar range.

Syntax

birnd(x) (init- or control-rate only)

Where the argument within the parentheses may be an expression. These value converters sample a global random sequence, but do not reference *seed*. The result can be a term in a further expression.

Performance

Returns a random number in the bipolar range $-x$ to x . *rnd* and *birnd* obtain values from a global pseudo-random number generator, then scale them into the requested range. The single global generator will thus distribute its sequence to these units throughout the performance, in whatever order the requests arrive.

Examples

Here is an example of the birnd opcode. It uses the file *birnd.csd* [examples/birnd.csd].

Example 63. Example of the birnd opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o birnd.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number from -1 to 1.
i1 = birnd(1)
print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #1 for one second.
i 1 1 1
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  i1 = 0.947  
instr 1:  i1 = -0.721
```

See Also

rnd

Credits

Author: Barry L. Vercoe
MIT
Cambridge, Massachusetts
1997

Extended in 3.47 to x-rate by John ffitich.

Example written by Kevin Conder.

bqrez

bqrez — A second-order multi-mode filter.

Description

A second-order multi-mode filter.

Syntax

```
ares bqrez asig, xfco, xres [, imode] [, iskip]
```

Initialization

imode (optional, default=0) -- The mode of the filter. Choose from one of the following:

- 0 = low-pass (default)
- 1 = high-pass
- 2 = band-pass
- 3 = band-reject
- 4 = all-pass

iskip (optional, default=0) -- if non zero skip the initialisation of the filter. (New in Csound version 4.23f13 and 5.0)

Performance

ares -- output audio signal.

asig -- input audio signal.

xfco -- filter cut-off frequency in Hz. May be i-time, k-rate, a-rate.

xres -- amount of resonance. Values of 1 to 100 are typical. Resonance should be one or greater. A value of 100 gives a 20dB gain at the cutoff frequency. May be i-time, k-rate, a-rate.

All filter modes can be frequency modulated as well as the resonance can also be frequency modulated.

bqrez is a resonant low-pass filter created using the Laplace s-domain equations for low-pass, high-pass, and band-pass filters normalized to a frequency. The bi-linear transform was used which contains a frequency transform constant from s-domain to z-domain to exactly match the frequencies together. A lot of trigonometric identities were used to simplify the calculation. It is very stable across the working frequency range up to the Nyquist frequency.

Examples

Here is an example of the *bqrez* opcode. It uses the file *bqrez.csd* [examples/bqrez.csd].

Example 64. Example of the *bqrez* opcode borrowed from the “rezy” opcode

in Kevin Conder's manual.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o bgrez.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Matt Gerassimof from example by Kevin Conder */
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a nice sawtooth waveform.
asig vco 16000, 220, 1

; Vary the filter-cutoff frequency from .2 to 2 KHz.
kfco line 200, p3, 2000

; Set the resonance amount.
kres init 0.99

al bgrez asig, kfco, kres

out al
endin

</CsInstruments>
<CsScore>

/* Written by Matt Gerassimof from example by Kevin Conder */
; Table #1, a sine wave for the vco opcode.
f 1 0 16384 10 1

; Play Instrument #1 for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

biquad, moogvcf, rezy

Credits

Author: Matt Gerassimoff
New in version 4.32
Written in November 2002.

butbp

butbp — Same as the butterbp opcode.

Description

Same as the *butterbp* opcode.

Syntax

```
ares butbp asig, kfreq, kband [, iskip]
```

butbr

butbr — Same as the butterbr opcode.

Description

Same as the *butterbr* opcode.

Syntax

```
ares butbr asig, kfreq, kband [, iskip]
```

buthp

buthp — Same as the butterhp opcode.

Description

Same as the *butterhp* opcode.

Syntax

```
ares buthp asig, kfreq [, iskip]
```


butlp

butlp — Same as the butterlp opcode.

Description

Same as the *butterlp* opcode.

Syntax

```
ares butlp asig, kfreq [, iskip]
```

butterbp

butterbp — A band-pass Butterworth filter.

Description

Implementation of a second-order band-pass Butterworth filter. This opcode can also be written as *butbp*.

Syntax

```
ares butterbp asig, kfreq, kband [, iskip]
```

Initialization

iskip (optional, default=0) -- Skip initialization if present and non-zero.

Performance

These filters are Butterworth second-order IIR filters. They are slightly slower than the original filters in Csound, but they offer an almost flat passband and very good precision and stopband attenuation.

asig -- Input signal to be filtered.

kfreq -- Cutoff or center frequency for each of the filters.

kband -- Bandwidth of the bandpass and bandreject filters.

Examples

Here is an example of the butterbp opcode. It uses the file *butterbp.csd* [examples/butterbp.csd].

Example 65. Example of the butterbp opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o butterbp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1 - an unfiltered noise waveform.
instr 1
; White noise signal
asig rand 22050

out asig
endin
```

```
; Instrument #2 - a filtered noise waveform.
instr 2
; White noise signal
asig rand 22050

; Filter it, passing only 1950 to 2050 Hz.
abp butterbp asig, 2000, 100

out abp
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

butterbr, butterhp, butterlp

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Existed in 3.30

butterbr

butterbr — A band-reject Butterworth filter.

Description

Implementation of a second-order band-reject Butterworth filter. This opcode can also be written as *butbr*.

Syntax

```
ares butterbr asig, kfreq, kband [, iskip]
```

Initialization

iskip (optional, default=0) -- Skip initialization if present and non-zero.

Performance

These filters are Butterworth second-order IIR filters. They are slightly slower than the original filters in Csound, but they offer an almost flat passband and very good precision and stopband attenuation.

asig -- Input signal to be filtered.

kfreq -- Cutoff or center frequency for each of the filters.

kband -- Bandwidth of the bandpass and bandreject filters.

Examples

Here is an example of the butterbr opcode. It uses the file *butterbr.csd* [examples/butterbr.csd].

Example 66. Example of the butterbr opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc       -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o butterbr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1 - an unfiltered noise waveform.
instr 1
; White noise signal
asig rand 22050

out asig
endin
```

```
; Instrument #2 - a filtered noise waveform.
instr 2
; White noise signal
asig rand 22050

; Filter it, cutting 2000 to 6000 Hz.
abr butterbr asig, 4000, 2000

out abr
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

butterbp, butterhp, butterlp

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Existed in 3.30

butterhp

butterhp — A high-pass Butterworth filter.

Description

Implementation of second-order high-pass Butterworth filter. This opcode can also be written as *buthp*.

Syntax

```
ares butterhp asig, kfreq [, iskip]
```

Initialization

iskip (optional, default=0) -- Skip initialization if present and non-zero.

Performance

These filters are Butterworth second-order IIR filters. They are slightly slower than the original filters in Csound, but they offer an almost flat passband and very good precision and stopband attenuation.

asig -- Input signal to be filtered.

kfreq -- Cutoff or center frequency for each of the filters.

Examples

Here is an example of the butterhp opcode. It uses the file *butterhp.csd* [examples/butterhp.csd].

Example 67. Example of the butterhp opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  Silent
-odac out      -idac in  -d      ;;realtime output
; For Non-realtime ouput leave only the line below:
; -o butterhp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1 - an unfiltered noise waveform.
instr 1
; White noise signal
asig rand 22050

out asig
endin
```

```
; Instrument #2 - a filtered noise waveform.
instr 2
; White noise signal
asig rand 22050

; Filter it, passing frequencies above 250 Hz.
ahp butterhp asig, 250

out ahp
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

butterbp, butterbr, butterlp

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Existed in 3.30

butterlp

butterlp — A low-pass Butterworth filter.

Description

Implementation of a second-order low-pass Butterworth filter. This opcode can also be written as *butlp*.

Syntax

```
ares butterlp asig, kfreq [, iskip]
```

Initialization

iskip (optional, default=0) -- Skip initialization if present and non-zero.

Performance

These filters are Butterworth second-order IIR filters. They are slightly slower than the original filters in Csound, but they offer an almost flat passband and very good precision and stopband attenuation.

asig -- Input signal to be filtered.

kfreq -- Cutoff or center frequency for each of the filters.

Examples

Here is an example of the butterlp opcode. It uses the file *butterlp.csd* [examples/butterlp.csd].

Example 68. Example of the butterlp opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o butterlp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1 - an unfiltered noise waveform.
instr 1
; White noise signal
asig rand 22050

out asig
endin
```



```
; Instrument #2 - a filtered noise waveform.
instr 2
; White noise signal
asig rand 22050

; Filter it, cutting frequencies above 1 KHz.
alp butterlp asig, 1000

out alp
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

butterbp, butterbr, butterhp

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Existed in 3.30

button

button — Sense on-screen controls.

Description

Sense on-screen controls. Requires Winsound or TCL/TK.

Syntax

```
kres button knum
```

Performance

kres -- value of the button control. If the button has been pushed since the last k-period, then return 1, otherwise return 0.

knum -- the number of the button. If it does not exist, it is made on-screen at initialization.

See Also

checkbox

Credits

Author: John ffitch
University of Bath, Codemist. Ltd.
Bath, UK
September 2000

New in Csound version 4.08

buzz

buzz — Output is a set of harmonically related sine partials.

Description

Output is a set of harmonically related sine partials.

Syntax

```
ares buzz xamp, xcps, knh, ifn [, iphs]
```

Initialization

ifn -- table number of a stored function containing a sine wave. A large table of at least 8192 points is recommended.

iphs (optional, default=0) -- initial phase of the fundamental frequency, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is zero

Performance

xamp -- amplitude

xcps -- frequency in cycles per second

The *buzz* units generate an additive set of harmonically related cosine partials of fundamental frequency *xcps*, and whose amplitudes are scaled so their summation peak equals *xamp*. The selection and strength of partials is determined by the following control parameters:

knh -- total number of harmonics requested. New in Csound version 3.57, *knh* defaults to one. If *knh* is negative, the absolute value is used.

buzz and *gbuzz* are useful as complex sound sources in subtractive synthesis. *buzz* is a special case of the more general *gbuzz* in which $klh = kmul = 1$; it thus produces a set of *knh* equal-strength harmonic partials, beginning with the fundamental. (This is a band-limited pulse train; if the partials extend to the Nyquist, i.e. $knh = \text{int}(sr / 2 / \text{fundamental freq.})$, the result is a real pulse train of amplitude *xamp*.)

Although *knh* may be varied during performance, its internal value is necessarily integer and may cause “pops” due to discontinuities in the output. *buzz* can be amplitude- and/or frequency-modulated by either control or audio signals.

N.B. This unit has its analog in *GENII*, in which the same set of cosines can be stored in a function table for sampling by an oscillator. Although computationally more efficient, the stored pulse train has a fixed spectral content, not a time-varying one as above.

Examples

Here is an example of the *buzz* opcode. It uses the file *buzz.csd* [examples/buzz.csd].

Example 69. Example of the buzz opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using com-

mand line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o buzz.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 20000
  kcps = 440
  knh = 3
  ifn = 1

  al buzz kamp, kcps, knh, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

gbuzz

Credits

September 2003. Thanks to Kanata Motohashi for correcting the mentions of the *kmul* parameter.

Example written by Kevin Conder.

cabasa

cabasa — Semi-physical model of a cabasa sound.

Description

cabasa is a semi-physical model of a cabasa sound. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares cabasa iamp, idettack [, inum] [, idamp] [, imaxshake]
```

Initialization

iamp -- Amplitude of output. Note: As these instruments are stochastic, this is only a approximation.

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 512.

idamp (optional) -- the damping factor, as part of this equation:

$$\text{damping_amount} = 0.998 + (\text{idamp} * 0.002)$$

The default *damping_amount* is 0.997 which means that the default value of *idamp* is -0.5. The maximum *damping_amount* is 1.0 (no damping). This means the maximum value for *idamp* is 1.0.

The recommended range for *idamp* is usually below 75% of the maximum value.

imaxshake (optional) -- amount of energy to add back into the system. The value should be in range 0 to 1.

Examples

Here is an example of the cabasa opcode. It uses the file *cabasa.csd* [examples/cabasa.csd].

Example 70. Example of the cabasa opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cabasa.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

;orchestra -----

sr =          44100
kr =          4410
ksmps =       10
nchnls =      1
```

```
instr 01 ;an example of a cabasa
a1 cabasa p4, 0.01
out a1
endin

</CsInstruments>
<CsScore>

;score -----

i1 0 1 26000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

crunch, sandpaper, sekere, stix

Credits

Author: Perry Cook, part of the PhISEM (Physically Informed Stochastic Event Modeling)
Adapted by John ffitch
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

cauchy

cauchy — Cauchy distribution random number generator.

Description

Cauchy distribution random number generator. This is an x-class noise generator.

Syntax

```
ares cauchy kalpha
```

```
ires cauchy kalpha
```

```
kres cauchy kalpha
```

Performance

kalpha -- controls the spread from zero (big *kalpha* = big spread). Outputs both positive and negative numbers.

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the cauchy opcode. It uses the file *cauchy.csd* [examples/cauchy.csd].

Example 71. Example of the cauchy opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cauchy.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number, spread from 10.
; kalpha = 10

il cauchy 10
```

```
    print i1
  endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  i1 = -0.106
```

See Also

seed, betarand, bexprnd, exprand, gauss, linrand, pcauchy, poisson, trirand, unirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Existed in 3.30

Example written by Kevin Conder.

ceil

ceil — Returns the smallest integer not less than x

Description

Returns the smallest integer not less than x

Syntax

`ceil(x)` (init-, control-, or audio-rate arg allowed)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

See Also

abs, exp, int, log, log10, i, sqrt

Credits

Author: Istvan Varga
New in Csound 5
2005

cent

cent — Calculates a factor to raise/lower a frequency by a given amount of cents.

Description

Calculates a factor to raise/lower a frequency by a given amount of cents.

Syntax

`cent(x)`

This function works at a-rate, i-rate, and k-rate.

Initialization

x -- a value expressed in cents.

Performance

The value returned by the *cent* function is a factor. You can multiply a frequency by this factor to raise/lower it by the given amount of cents.

Examples

Here is an example of the cent opcode. It uses the file *cent.csd* [examples/cent.csd].

Example 72. Example of the cent opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cent.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; The root note is A above middle-C (440 Hz)
iroot = 440

; Raise the root note by 300 cents to C.
icents = 300

; Calculate the new note.
ifactor = cent(icents)
inew = iroot * ifactor

; Print out of all of the values.
print iroot
print ifactor
```

```
    print inew
  endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  iroot = 440.000
instr 1:  ifactor = 1.189
instr 1:  inew = 523.229
```

See Also

db, octave, semitone

Credits

Example written by Kevin Conder.

New in version 4.16

cggoto

cggoto — Conditionally transfer control on every pass.

Description

Transfer control to *label* on every pass. (Combination of *cigoto* and *ckgoto*)

Syntax

```
cggoto condition, label
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

Examples

Here is an example of the cggoto opcode. It uses the file *cggoto.csd* [examples/cggoto.csd].

Example 73. Example of the cggoto opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cggoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  il = p4

  ; If il is equal to one, play a high note.
  ; Otherwise play a low note.
  cggoto (il == 1), highnote

lownote:
  a1 oscil 10000, 220, 1
  goto playit

highnote:
  a1 oscil 10000, 440, 1
  goto playit

playit:
  out a1
endin

</CsInstruments>
<CsScore>

; Table #1: a simple sine wave.
f 1 0 32768 10 1

; Play lownote for one second.
```

```
i 1 0 1 1
; Play highnote for one second.
i 1 0 1 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

cigoto, ckgoto, cngoto, if, igoto, kgoto, tigoto, timeout

Credits

Added a note by Jim Aikin.

Example written by Kevin Conder.

chanctrl

chanctrl — Get the current value of a MIDI channel controller.

Description

Get the current value of a controller and optionally map it onto specified range.

Syntax

```
ival chanctrl ichnl, ictlno [, ilow] [, ihigh]
```

```
kval chanctrl ichnl, ictlno [, ilow] [, ihigh]
```

Initialization

ichnl -- the MIDI channel (1-16).

ictlno -- the MIDI controller number (0-127).

ilow, ihigh -- low and high ranges for mapping

Credits

Author: Mike Berry
Mills College
May, 1997

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

changed

changed — k-rate signal change detector.

Description

This opcode outputs a trigger signal that informs when any one of its k-rate arguments has changed. Useful with valuator widgets or MIDI controllers.

Syntax

```
ktrig changed kvar1 [, kvar2,..., kvarN]
```

Performance

ktrig - Outputs a value of 1 when any of the k-rate signals has changed, otherwise outputs 0.

kvar1 [, *kvar2*,..., *kvarN*] - k-rate variables to watch for changes.

Examples

Here is an example of the changed opcode. It uses the file *changed.csd* [examples/changed.csd].

Example 74. Example of the changed opcode.

```
<CsoundSynthesizer>
<CsOptions>
-odac -B441 -b441
</CsOptions>
<CsInstruments>

sr      =      44100
kr      =      100
ksmps   =      441
nchnls  =      2

      instr      1
ksig oscil 2,0.5,1
kint = int(ksig)
ktrig changed kint
printk 0.2, kint
printk2 ktrig
      endin

</CsInstruments>
<CsScore>
f 1 0 1024 10 1
i 1 0 20

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado.

Example written by Andrés Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

chani

chani — Reads data from the software bus

Description

Reads data from a channel of the inward software bus.

Syntax

```
kval chani kchan
```

```
aval chani kchan
```

Initialization

Performance

kchan -- a positive integer that indicates which channel of the software bus to read

Note that the inward and outward software busses are independent, and are not mixer busses. The last value remains until a new value is written. There is no imposed limit to the number of busses but they use memory so small numbers are to be preferred.

Example

The example shows the software bus being used as an asynchronous control signal to select a filter cutoff. It assumes that an external program that has access to the API is feeding the values

```
sr = 44100
ksmps = 100
nchnls = 1

instr 1
  kc  chani 1
  a1  oscil p4, p5, 100
  a2  lowpass2 a1, kc, 200
  out  a2
endin
```

Credits

Author: John fitch
2005

New in Csound5.00

chano

chano — Send data to the outwards software bus

Description

Send data to a channel of the outward software bus.

Syntax

```
chano kval, kchan
```

```
chano aval, kchan
```

Initialization

Performance

xval --- value to transmit

kchan -- a positive integer that indicates which channel of the software bus to write

Note that the inward and outward software busses are independent, and are not mixer buses. The last value remains until a new value is written. There is no imposed limit to the number of busses but they use memory so small numbers are to be preferred.

Example

The example shows the software bus being used as an asynchronous control signal to select a filter cutoff. It assumes that an external program that has access to the API is feeding the values

```
sr = 44100
ksmps = 100
nchnls = 1

instr 1
  al oscil      p4, p5, 100
  chano        1, al
endin
```

Credits

Author: John fitch
2005

New in Csound5.00

chebyshevpoly

chebyshevpoly — Efficiently evaluates the sum of Chebyshev polynomials of arbitrary order.

Description

The *chebyshevpoly* opcode calculates the value of a polynomial expression with a single a-rate input variable that is made up of a linear combination of the first N Chebyshev polynomials of the first kind. Each Chebyshev polynomial, $T_n(x)$, is weighted by a k-rate coefficient, kn , so that the opcode is calculating a sum of any number of terms in the form $kn * T_n(x)$. Thus, the *chebyshevpoly* opcode allows for the waveshaping of an audio signal with a *dynamic* transfer function that gives precise control over the harmonic content of the output.

Syntax

```
aout chebyshevpoly ain, k0 [, k1 [, k2 [...]]]
```

Performance

ain -- the input signal used as the independent variable of the Chebyshev polynomials ("x").

aout -- the output signal ("y").

k0, k1, k2, ... -- k-rate multipliers for each Chebyshev polynomial.

This opcode is very useful for dynamic waveshaping of an audio signal. Traditional waveshaping techniques utilize a lookup table for the transfer function -- usually a sum of Chebyshev polynomials. When a sine wave at full-scale amplitude is used as an index to read the table, the precise harmonic spectrum as defined by the weights of the Chebyshev polynomials is produced. A dynamic spectrum is achieved by varying the amplitude of the input sine wave, but this produces a non-linear change in the spectrum.

By directly calculating the Chebyshev polynomials, the *chebyshevpoly* opcode allows more control over the spectrum and the number of harmonic partials added to the input can be varied with time. The value of each kn coefficient directly controls the amplitude of the n th harmonic partial if the input *ain* is a sine wave with amplitude = 1.0. This makes *chebyshevpoly* an efficient additive synthesis engine for N partials that requires only one oscillator instead of N oscillators. The amplitude or waveform of the input signal can also be changed for different waveshaping effects.

If we consider the input parameter *ain* to be "x" and the output *aout* to be "y", then the *chebyshevpoly* opcode calculates the following equation:

$$y = k0 * T_0(x) + k1 * T_1(x) + k2 * T_2(x) + k3 * T_3(x) + \dots$$

where the $T_n(x)$ are defined by the recurrence relation

$$\begin{aligned} T_0(x) &= 1, \\ T_1(x) &= x, \\ T_n(x) &= 2x * T_{n-1}(x) - T_{n-2}(x) \end{aligned}$$

More information about Chebyshev polynomials can be found on Wikipedia at http://en.wikipedia.org/wiki/Chebyshev_polynomial [ht-

[tp://en.wikipedia.org/wiki/Chebyshev_polynomial\]](http://en.wikipedia.org/wiki/Chebyshev_polynomial)

See Also

polynomial, *mac maca*

Examples

Here is an example of the `chebyshevpoly` opcode. It uses the file *chebyshevpoly.csd* [examples/chebyshevpoly.csd].

Example 75. Example of the `chebyshevpoly` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o abs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 441

; time-varying mixture of first six harmonics
instr 1
    ; According to the GEN13 manual entry,
    ; the pattern + - - + + - - for the signs of
    ; the chebyshev coefficients has nice properties.

    ; these six lines control the relative powers of the harmonics
    k1      line      1.0, p3, 0.0
    k2      line      -0.5, p3, 0.0
    k3      line      -0.333, p3, -1.0
    k4      line      0.0, p3, 0.5
    k5      line      0.0, p3, 0.7
    k6      line      0.0, p3, -1.0

    ; play the sine wave at a frequency of 256 Hz
    ax      oscili     1.0, 256, 1

    ; waveshape it
    ay      chebyshevpoly ax, 0, k1, k2, k3, k4, k5, k6

    ; avoid clicks, scale final amplitude, and output
    adeclick linseg     0.0, 0.05, 1.0, p3 - 0.1, 1.0, 0.05, 0.0
    out      ay * adeclick * 10000
endin

</CsInstruments>
<CsScore>
f1 0 32768 10 1 ; a sine wave

i1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Anthony Kozar
January 2008

New in Csound version 5.08

checkbox

checkbox — Sense on-screen controls.

Description

Sense on-screen controls. Requires Winsound or TCL/TK.

Syntax

kres **checkbox** knum

Performance

kres -- value of the checkbox control. If the checkbox is set (pushed) then return 1, if not, return 0.

knun -- the number of the checkbox. If it does not exist, it is made on-screen at initialization.

Examples

Here is a simple example of the checkbox opcode. It uses the file *checkbox.csd* [examples/checkbox.csd].

Example 76. Simple example of the checkbox opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o checkbox.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

instr 1
; Get the value from the checkbox.
k1 checkbox 1

; If the checkbox is selected then k2=440, otherwise k2=880.
k2 = (k1 == 0 ? 440 : 880)

a1 oscil 10000, k2, 1
out a1
endin

</CsInstruments>
<CsScore>

; Just generate a nice, ordinary sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for ten seconds.
i 1 0 10
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

button

Credits

Author: John fitch
University of Bath, Codemist. Ltd.
Bath, UK
September, 2000

Example written by Kevin Conder.

New in Csound version 4.08

chn

chn — Declare a channel of the named software bus.

Description

Declare a channel of the named software bus, with setting optional parameters in the case of a control channel. If the channel does not exist yet, it is created, with an initial value of zero or empty string. Otherwise, the type (control, audio, or string) of the existing channel must match the declaration, or an init error occurs. The input/output mode of an existing channel is updated so that it becomes the bitwise OR of the previous and the newly specified value.

Syntax

```
chn_k Sname, imode[, itype, idflt, imin, imax]
```

```
chn_a Sname, imode
```

```
chn_s Sname, imode
```

Initialization

imode -- sum of at least one of 1 for input and 2 for output.

itype (optional, defaults to 0) -- channel subtype for control channels only. Possible values are:

- 0: default/unspecified (*idflt*, *imin*, and *imax* are ignored)
- 1: integer values only
- 2: linear scale
- 3: exponential scale

idflt (optional, defaults to 0) -- default value, for control channels with non-zero *itype* only. Must be greater than or equal to *imin*, and less than or equal to *imax*.

imin (optional, defaults to 0) -- minimum value, for control channels with non-zero *itype* only. Must be non-zero for exponential scale (*itype* = 3).

imax (optional, defaults to 0) -- maximum value, for control channels with non-zero *itype* only. Must be greater than *imin*. In the case of exponential scale, it should also match the sign of *imin*.

Notes

The channel parameters (*imode*, *itype*, *idflt*, *imin*, and *imax*) are only hints for the host application or external software accessing the bus through the API, and do not actually restrict reading from or writing to the channel in any way. Also, the initial value of a newly created control channel is zero, regardless of the setting of *idflt*.

For communication with external software, using `chnexport` may be preferred, as it allows direct access to orchestra variables exported as channels of the bus, eliminating the need for using `chnset` and `chnget` to send or receive data.

Performance

chn_k, **chn_a**, and **chn_S** declare a control, audio, or string channel, respectively.

Example

The example shows the software bus being used as an asynchronous control signal to select a filter cutoff. It assumes that an external program that has access to the API is feeding the values.

```
sr = 44100
ksmps = 100
nchnls = 1

chn_k "cutoff", 1, 3, 1000, 500, 2000

instr 1
  kc  chnget  "cutoff"
  a1  oscil   p4, p5, 100
  a2  lowpass2 a1, kc, 200
  out  a2
endin
```

Credits

Author: Istvan Varga
2005

chnclear

chnclear — Clears an audio output channel of the named software bus.

Description

Clears an audio channel of the named software bus to zero. Implies declaring the channel with `imode=2` (see also `chn_a`).

Syntax

`chnclear` *Sname*

Initialization

Sname -- a string that indicates which named channel of the software bus to write.

Credits

Author: Istvan Varga
2006

chnexport

chnexport — Export a global variable as a channel of the bus.

Description

Export a global variable as a channel of the bus; the channel should not already exist, otherwise an init error occurs. This opcode is normally called from the orchestra header, and allows the host application to read or write orchestra variables directly, without having to use `chnget` or `chnset` to copy data.

Syntax

```
gival chnexport Sname, imode[, itype, idflt, imin, imax]
```

```
gkval chnexport Sname, imode[, itype, idflt, imin, imax]
```

```
gaval chnexport Sname, imode
```

```
gSval chnexport Sname, imode
```

Initialization

imode -- sum of at least one of 1 for input and 2 for output.

itype (optional, defaults to 0) -- channel subtype for control channels only. Possible values are:

- 0: default/unspecified (*idflt*, *imin*, and *imax* are ignored)
- 1: integer values only
- 2: linear scale
- 3: exponential scale

idflt (optional, defaults to 0) -- default value, for control channels with non-zero *itype* only. Must be greater than or equal to *imin*, and less than or equal to *imax*.

imin (optional, defaults to 0) -- minimum value, for control channels with non-zero *itype* only. Must be non-zero for exponential scale (*itype* = 3).

imax (optional, defaults to 0) -- maximum value, for control channels with non-zero *itype* only. Must be greater than *imin*. In the case of exponential scale, it should also match the sign of *imin*.

Notes

The channel parameters (*imode*, *itype*, *idflt*, *imin*, and *imax*) are only hints for the host application or external software accessing the bus through the API, and do not actually restrict reading from or writing to the channel in any way.

While the global variable is used as output argument, `chnexport` does not actually change it, and always runs at i-time only. If the variable is not previously declared, it is created by `Csound` with an initial value of zero or empty string.

Performance

Example

The example shows the software bus being used as an asynchronous control signal to select a filter cutoff. It assumes that an external program that has access to the API is feeding the values.

```
sr = 44100
ksmps = 100
nchnls = 1

gkc init 1000 ; set default value
gkc chnexport "cutoff", 1, 3, i(gkc), 500, 2000

instr 1
  a1 oscil p4, p5, 100
  a2 lowpass2 a1, gkc, 200
  out a2
endin
```

Credits

Author: Istvan Varga
2005

chnget

chnget — Reads data from the software bus.

Description

Reads data from a channel of the inward named software bus. Implies declaring the channel with `imode=1` (see also `chn_k`, `chn_a`, and `chn_S`).

Syntax

`ival` **chnget** *Sname*

`kval` **chnget** *Sname*

`aval` **chnget** *Sname*

`Sval` **chnget** *Sname*

Initialization

Sname -- a string that identifies a channel of the named software bus to read.

Performance

ival -- the control value read at i-time.

kval -- the control value read at performance time.

aval -- the audio signal read at performance time.

Sval -- the string value read at i-time.

Example

The example shows the software bus being used as an asynchronous control signal to select a filter cutoff. It assumes that an external program that has access to the API is feeding the values.

```
sr = 44100
ksmps = 100
nchnls = 1

instr 1
  kc  chnget  "cutoff"
  a1  oscil   p4, p5, 100
  a2  lowpass2 a1, kc, 200
  out  out    a2
endin
```

Credits

Author: Istvan Varga
2005

chnmix

chnmix — Writes audio data to the named software bus, mixing to the previous output.

Description

Adds an audio signal to a channel of the named software bus. Implies declaring the channel with `imode=2` (see also `chn_a`).

Syntax

```
chnmix aval, Sname
```

Initialization

Sname -- a string that indicates which named channel of the software bus to write.

Performance

aval -- the audio signal to write at performance time.

Credits

Author: Istvan Varga
2006

chnparams

chnparams — Query parameters of a channel.

Description

Query parameters of a channel (if it does not exist, all returned values are zero).

Syntax

```
itype, imode, ictltype, idflt, imin, imax chnparams
```

Initialization

itype -- channel data type (1: control, 2: audio, 3: string)

imode -- sum of 1 for input and 2 for output

ictltype -- special parameter for control channel only; if not available, set to zero.

idflt -- special parameter for control channel only; if not available, set to zero.

imin -- special parameter for control channel only; if not available, set to zero.

imax -- special parameter for control channel only; if not available, set to zero.

Performance

Example

Credits

Author: Istvan Varga
2005

chnset

chnset — Writes data to the named software bus.

Description

Write to a channel of the named software bus. Implies declaring the channel with imode=2 (see also chn_k, chn_a, and chn_S).

Syntax

```
chnset ival, Sname
```

```
chnset kval, Sname
```

```
chnset aval, Sname
```

```
chnset Sval, Sname
```

Initialization

Sname -- a string that indicates which named channel of the software bus to write.

Performance

ival -- the control value to write at i-time.

kval -- the control value to write at performance time.

aval -- the audio signal to write at performance time.

Sval -- the string value to write at i-time.

Example

The example shows the software bus being used to write pitch information to a controlling program.

```
sr = 44100
ksmps = 100
nchnls = 1

instr 1
  al in
  kp,ka pitchamdf al
  chnset kp, "pitch"
endin
```

Credits

Author: Istvan Varga
2005

chuap

chuap — Simulates Chua's oscillator, an LRC oscillator with an active resistor, proved capable of bifurcation and chaotic attractors, with k-rate control of circuit elements.

Description

Simulates Chua's oscillator, an LRC oscillator with an active resistor, proved capable of bifurcation and chaotic attractors, with k-rate control of circuit elements.

Syntax

```
aI3, aV2, aV1 chuap kL, kR0, kC1, kG, kGa, kGb, kE, kC2, iI3, iV2, iV1, ktime_step
```

Initialization

iI3 -- Initial current at G

iV2 -- Initial voltage at C2

iV1 -- Initial voltage at C1

Performance

kL -- Inductor L

kR0 -- Resistor R0

kC1 -- Capacitor C1

kG -- Resistor G

kGa -- Resistor V (nonlinearity term)

kGb -- Resistor V (nonlinearity term)

kGb -- Resistor V (nonlinearity term)

ktime_step -- Delta time in the difference equation, can be used to more or less control pitch.

Chua's oscillator is a simple LRC oscillator with an active resistor. The oscillator can be driven into period bifurcation, and thus to chaos, because of the nonlinear response of the active resistor.

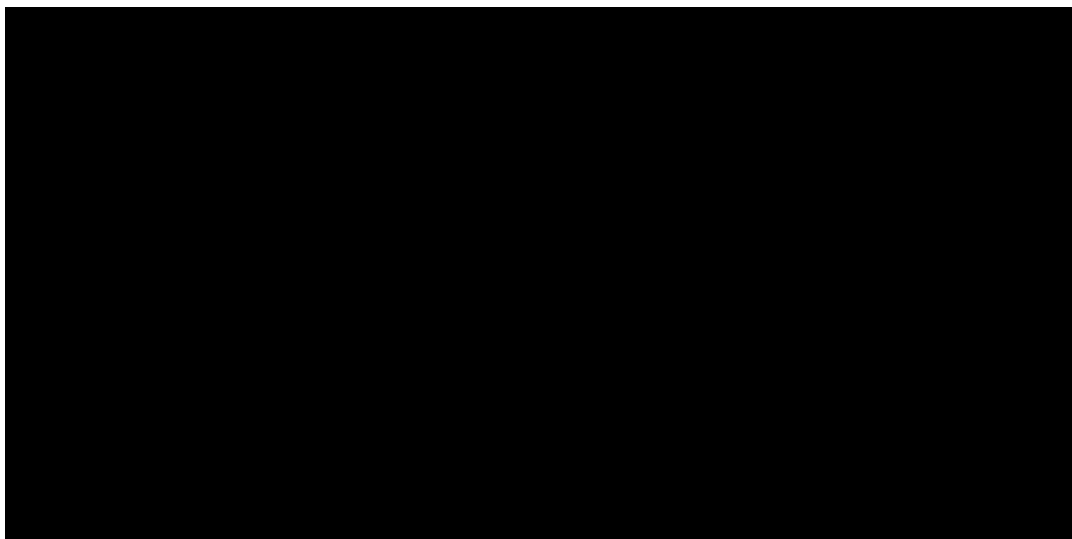


Diagram of Chua's Oscillator Circuit

The circuit is described by a set of three ordinary differential equations called Chua's equations:

$$\frac{dI_3}{dt} = -\frac{R_0}{L} I_3 - \frac{1}{L} V_2$$

$$\frac{dV_2}{dt} = -\frac{1}{C_2} I_3 - \frac{G}{C_2} (V_2 - V_1)$$

$$\frac{dV_1}{dt} = \frac{G}{C_1} (V_2 - V_1) - \frac{1}{C_1} f(V_1)$$

where $f()$ is a piecewise discontinuity simulating the active resistor:

$$f(V_1) = G_b V_1 + (G_a - G_b)(|V_1 + E| - |V_1 - E|)$$

A solution of these equations $(I_3, V_2, V_1)(t)$ starting from an initial state $(I_3, V_2, V_1)(0)$ is called a trajectory of Chua's oscillator. The Csound implementation is a difference equation simulation of Chua's oscillator with Runge-Kutta integration.



Warning

Be careful! Some sets of parameters will produce amplitude spikes or positive feedback that could damage your speakers.

Examples

Here is an example of the `chuap` opcode. It uses the file `chuap.csd` [examples/chuap.csd].

Example 77. Example of the `chuap` opcode.


```
<CsoundSynthesizer>
<CsOptions>
csound -RWfo chuas_oscillator.wav
</CsOptions>
<CsInstruments>
sr          =          44100
ksmps       =          100
nchnls      =          2
0dbfs       =          10000

gibuzztable  ftgen      1, 0, 16384, 10, 1

                                instr 1
                                ; sys_variables = system_vars(5:12); % L,R0,C2,G,Ga,Gb,E,C1 or p8:p15
                                ; integ_variables = [system_vars(14:16),system_vars(1:2)]; % x0,y0,z0,dataset_size,
istep_size   =          p5
iL           =          p8
iR0          =          p9
iC2          =          p10
iG           =          p11
iGa          =          p12
iGb          =          p13
iE           =          p14
iC1          =          p15
iI3          =          p17
iV2          =          p18
iV1          =          p19
iattack      =          0.02
isustain     =          p3
irelease     =          0.02
p3           =          iattack + isustain + irelease
iscale       =          1.0
adamping     =          linseg 0.0, iattack, iscale, isustain, iscale, irelease, 0.0
aguide       =          buzz 5000, 440, sr/440, gibuzztable
aI3, aV2, aV1 =          chuap iL, iR0, iC2, iG, iGa, iGb, iE, iC1, iI3, iV2, iV1, istep_size
asignal      =          aV2, aguide
                                balance
                                outs
                                endin
</CsInstruments>
<CsScore>
;          Adapted from ABC++ MATLAB example data.
i 1 0 20 1500 .1 -1 -1 -0.00707925 0.00001647 100 1 -.99955324 -1.00028375 1 -.00222159 204.8 -2
i 1 + 20 1500 .425 0 -1 1.3506168 0 -4.50746268737 -1 2.4924 .93 1 1 0 -22.28662665
i 1 + 20 1024 .05 -1 -1 0.00667 0.000651 10 -1 .856 1.1 1 .06 51.2 -20.200590133667 .172539
i 1 + 20 1024 0.05 -1 -1 0.00667 0.000651 10 -1 0.856 1.1 1 0.1 153.6 21.12496758 0.03001749 0.5158
</CsScore>
</CsoundSynthesizer>
```

Credits

Inventor of Chua's oscillator: *Leon O. Chua* [<http://www.eecs.berkeley.edu/~chua>]

Author of MATLAB simulation: James Patrick McEvoy *MATLAB Adventures in Bifurcations and Chaos* (ABC++) [ht-

[tp://www.mathworks.com/matlabcentral/fileexchange/loadFile.do?objectId=3541](http://www.mathworks.com/matlabcentral/fileexchange/loadFile.do?objectId=3541)]

Author of Csound port: Michael Gogins

New in Csound version 5.09

cigoto

cigoto — Conditionally transfer control during the i-time pass.

Description

During the i-time pass only, unconditionally transfer control to the statement labeled by *label*.

Syntax

```
cigoto condition, label
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

Examples

Here is an example of the cigoto opcode. It uses the file *cigoto.csd* [examples/cigoto.csd].

Example 78. Example of the cigoto opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Get the value of the 4th p-field from the score.
iparam = p4

; If iparam is 1 then play the high note.
; If not then play the low note.
cigoto (iparam ==1), highnote
      igoto lownote

highnote:
  ifreq = 880
  goto playit

lownote:
  ifreq = 440
  goto playit

playit:
; Print the values of iparam and ifreq.
print iparam
print ifreq

  a1 oscil 10000, ifreq, 1
  out a1
endin
```

```
</CsInstruments>
<CsScore>

; Table #1: a simple sine wave.
f 1 0 32768 10 1

; p4: 1 = high note, anything else = low note
; Play Instrument #1 for one second, a low note.
i 1 0 1 0
; Play a Instrument #1 for one second, a high note.
i 1 1 1 1
e

</CsScore>
</CsSoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  iparam = 0.000
instr 1:  ifreq = 440.000
instr 1:  iparam = 1.000
instr 1:  ifreq = 880.000
```

See Also

cggoto, ckgoto, cngoto, goto, if, kgoto, rigoto, tigoto, timeout

Credits

Added a note by Jim Aikin.

Example written by Kevin Conder.

ckgoto

ckgoto — Conditionally transfer control during the p-time passes.

Description

During the p-time passes only, unconditionally transfer control to the statement labeled by *label*.

Syntax

ckgoto condition, label

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

Examples

Here is an example of the ckgoto opcode. It uses the file *ckgoto.csd* [examples/ckgoto.csd].

Example 79. Example of the ckgoto opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ckgoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Change kval linearly from 0 to 2 over
; the period set by the third p-field.
kval line 0, p3, 2

; If kval is greater than or equal to 1 then play the high note.
; If not then play the low note.
ckgoto (kval >= 1), highnote
      kgoto lownote

highnote:
  kfreq = 880
  goto playit

lownote:
  kfreq = 440
  goto playit

playit:
; Print the values of kval and kfreq.
printks "kval = %f, kfreq = %f\\n", 1, kval, kfreq

  a1 oscil 10000, kfreq, 1
  out a1
endin
```

```
</CsInstruments>
<CsScore>

; Table: a simple sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
kval = 0.000000, kfreq = 440.000000
kval = 0.999732, kfreq = 440.000000
kval = 1.999639, kfreq = 880.000000
```

See Also

cggoto, cigoto, cngoto, goto, if, igoto, tigoto, timeout

Credits

Added a note by Jim Aikin.

Example written by Kevin Conder.

clear

clear — Zeroes a list of audio signals.

Description

clear zeroes a list of audio signals.

Syntax

```
clear avar1 [, avar2] [, avar3] [...]
```

Performance

avar1, *avar2*, *avar3*, ... -- signals to be zeroed

clear sets every sample of each of the given audio signals to zero when it is performed. This is equivalent to writing $avarN = 0$ in the orchestra for each of the specified variables. Typically, *clear* is used with global variables that combine multiple signals from different sources and change with each k-pass (performance loop) through all of the active instrument instances. After the final usage of such a variable and before the next k-pass, it is necessary to clear the variable so that it does not add the next cycle's signals to the previous result. *clear* is especially useful in combination with *vincr* (variable increment) and they are intended to be used together with file output opcodes such as *fout*.

Examples

See the *fout* opcode for an example.

See Also

vincr

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

clfilt

clfilt — Implements low-pass and high-pass filters of different styles.

Description

Implements the classical standard analog filter types: low-pass and high-pass. They are implemented with the four classical kinds of filters: Butterworth, Chebyshev Type I, Chebyshev Type II, and Elliptical. The number of poles may be any even number from 2 to 80.

Syntax

```
ares clfilt asig, kfreq, itype, inpol [, ikind] [, ipbr] [, isba] [, iskip]
```

Initialization

itype -- 0 for low-pass, 1 for high-pass.

inpol -- The number of poles in the filter. It must be an even number from 2 to 80.

ikind (optional) -- 0 for Butterworth, 1 for Chebyshev Type I, 2 for Chebyshev Type II, 3 for Elliptical. Defaults to 0 (Butterworth)

ipbr (optional) -- The pass-band ripple in dB. Must be greater than 0. It is ignored by Butterworth and Chebyshev Type II. The default is 1 dB.

isba (optional) -- The stop-band attenuation in dB. Must be less than 0. It is ignored by Butterworth and Chebyshev Type I. The default is -60 dB.

iskip (optional) -- 0 initializes all filter internal states to 0. 1 skips initialization. The default is 0.

Performance

asig -- The input audio signal.

kfreq -- The corner frequency for low-pass or high-pass.

Examples

Here is an example of the clfilt opcode as a low-pass filter. It uses the file *clfilt_lowpass.csd* [examples/clfilt_lowpass.csd].

Example 80. Example of the clfilt opcode as a low-pass filter.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o clfilt_lowpass.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
```

```
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1 - an unfiltered noise waveform.
instr 1
; White noise signal
asig rand 22050

out asig
endin

; Instrument #2 - a filtered noise waveform.
instr 2
; White noise signal
asig rand 22050

; Lowpass filter signal asig with a
; 10-pole Butterworth at 500 Hz.
al clfilt asig, 500, 0, 10

out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the `clfilt` opcode as a high-pass filter. It uses the file `clfilt_highpass.csd` [examples/clfilt_highpass.csd].

Example 81. Example of the `clfilt` opcode as a high-pass filter.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac        -iadc       -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o clfilt_highpass.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1 - an unfiltered noise waveform.
instr 1
; White noise signal
asig rand 22050

out asig
endin

; Instrument #2 - a filtered noise waveform.
instr 2
; White noise signal
asig rand 22050

; Highpass filter signal asig with a 6-pole Chebyshev
```



```
; Type I at 20 Hz with 3 dB of passband ripple.
al clfilt asig, 20, 1, 6, 1, 3

out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Erik Spjut

New in version 4.20

clip

clip — Clips a signal to a predefined limit.

Description

Clips an a-rate signal to a predefined limit, in a “soft” manner, using one of three methods.

Syntax

```
ares clip asig, imeth, ilimit [, iarg]
```

Initialization

imeth -- selects the clipping method. The default is 0. The methods are:

- 0 = Bram de Jong method (default)
- 1 = sine clipping
- 2 = tanh clipping

ilimit -- limiting value

iarg (optional, default=0.5) -- when *imeth* = 0, indicates the point at which clipping starts, in the range 0 - 1. Not used when *imeth* = 1 or *imeth* = 2. Default is 0.5.

Performance

asig -- a-rate input signal

The Bram de Jong method (*imeth* = 0) applies the algorithm:

$$|x| > a: \quad f(x) = \sin(x) * (a + (x-a)/(1 + ((x-a)/(1-a))^2) \quad |x| > 1: f(x) = \sin(x) * (a+1)/2$$

This method requires that *asig* be normalized to 1.

The second method (*imeth* = 1) is the sine clip:

$$|x| < limit: f(x) = limit * \sin(\pi * x / (2 * limit)) \quad f(x) = limit * \sin(x)$$

The third method (*imeth* = 3) is the tanh clip:

$$|x| < limit: f(x) = limit * \tanh(x/limit) / \tanh(1) \quad f(x) = limit * \sin(x)$$



Note

Method 1 appears to be non-functional at release of Csound version 4.07.

Examples

Here is an example of the clip opcode. It uses the file *clip.csd* [examples/clip.csd].

Example 82. Example of the clip opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o clip.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a noisy waveform.
arnd rand 44100
; Clip the noisy waveform's amplitude to 20,000
a1 clip arnd, 2, 20000

out a1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch
University of Bath, Codemist Ltd.
Bath, UK
August, 2000

New in Csound version 4.07

clock

clock — Deprecated.

Description

Deprecated. Use the *rtclock* opcode instead.

clockoff

clockoff — Stops one of a number of internal clocks.

Description

Stops one of a number of internal clocks.

Syntax

```
clockoff inum
```

Initialization

inum -- the number of a clock. There are 32 clocks numbered 0 through 31. All other values are mapped to clock number 32.

Performance

Between a *clockon* and a *clockoff* opcode, the CPU time used is accumulated in the clock. The precision is machine dependent but is the millisecond range on UNIX and Windows systems. The *readclock* opcode reads the current value of a clock at initialization time.

Examples

See the *readclock* opcode for an example.

See Also

clockon, *readclock*

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
July, 1999

New in Csound version 3.56

clockon

clockon — Starts one of a number of internal clocks.

Description

Starts one of a number of internal clocks.

Syntax

```
clockon inum
```

Initialization

inum -- the number of a clock. There are 32 clocks numbered 0 through 31. All other values are mapped to clock number 32.

Performance

Between a *clockon* and a *clockoff* opcode, the CPU time used is accumulated in the clock. The precision is machine dependent but is the millisecond range on UNIX and Windows systems. The *readclock* opcode reads the current value of a clock at initialization time.

Examples

See the *readclock* opcode for an example.

See Also

clockoff, *readclock*

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
July, 1999

New in Csound version 3.56

cngoto

cngoto — Transfers control on every pass when a condition is not true.

Description

Transfers control on every pass when the condition is *not* true.

Syntax

```
cngoto condition, label
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

Examples

Here is an example of the cngoto opcode. It uses the file *cngoto.csd* [examples/cngoto.csd].

Example 83. Example of the cngoto opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cngoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Change kval linearly from 0 to 2 over
; the period set by the third p-field.
kval line 0, p3, 2

; If kval *is not* greater than or equal to 1 then play
; the high note. Otherwise, play the low note.
cngoto (kval >= 1), highnote
      kgoto lownote

highnote:
  kfreq = 880
  goto playit

lownote:
  kfreq = 440
  goto playit

playit:
; Print the values of kval and kfreq.
printks "kval = %f, kfreq = %f\\n", 1, kval, kfreq

  a1 oscil 10000, kfreq, 1
  out a1
endin
```

```
</CsInstruments>
<CsScore>

; Table: a simple sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
kval = 0.000000, kfreq = 880.000000
kval = 0.999732, kfreq = 880.000000
kval = 1.999639, kfreq = 440.000000
```

See Also

cggoto, cigoto, ckgoto, goto, if, igoto, tigoto, timeout

Credits

Example written by Kevin Conder.

New in version 4.21

comb

comb — Reverberates an input signal with a “colored” frequency response.

Description

Reverberates an input signal with a “colored” frequency response.

Syntax

```
ares comb asig, krvt, ilpt [, iskip] [, insmps]
```

Initialization

ilpt -- loop time in seconds, which determines the “echo density” of the reverberation. This in turn characterizes the “color” of the *comb* filter whose frequency response curve will contain *ilpt* * *sr*/2 peaks spaced evenly between 0 and *sr*/2 (the Nyquist frequency). Loop time can be as large as available memory will permit. The space required for an *n* second loop is $4n*sr$ bytes. Delay space is allocated and returned as in *delay*.

iskip (optional, default=0) -- initial disposition of delay-loop data space (cf. *reson*). The default value is 0.

insmps (optional, default=0) -- delay amount, as a number of samples.

Performance

krvt -- the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude).

This filter reiterates input with an echo density determined by loop time *ilpt*. The attenuation rate is independent and is determined by *krvt*, the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude). Output from a comb filter will appear only after *ilpt* seconds.

Examples

Here is an example of the comb opcode. It uses the file *comb.csd* [examples/comb.csd].

Example 84. Example of the comb opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o comb.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
```

```
nchnls = 1

; Initialize the audio mixer.
gamix init 0

; Instrument #1.
instr 1
  ; Generate a source signal.
  al oscili 30000, cpspch(p4), 1
  ; Output the direct sound.
  out al

  ; Add the source signal to the audio mixer.
  gamix = gamix + al
endin

; Instrument #99 (highest instr number executed last)
instr 99
  krvt = 1.5
  ilpt = 0.1

  ; Comb-filter the mixed signal.
  a99 comb gamix, krvt, ilpt
  ; Output the result.
  out a99

  ; Empty the mixer for the next pass.
  gamix = 0
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 128 10 1

; p4 = frequency (in a pitch-class)
; Play Instrument #1 for a tenth of a second, p4=7.00
i 1 0 0.1 7.00
; Play Instrument #1 for a tenth of a second, p4=7.02
i 1 1 0.1 7.02
; Play Instrument #1 for a tenth of a second, p4=7.04
i 1 2 0.1 7.04
; Play Instrument #1 for a tenth of a second, p4=7.06
i 1 3 0.1 7.06

; Make sure the comb-filter remains active.
i 99 0 5
e

</CsScore>
</CsoundSynthesizer>
```

See Also

alpass, *reverb*, *valpass*, *vcomb*

Credits

Author: William “Pete” Moss (*vcomb* and *valpass*)
University of Texas at Austin
Austin, Texas USA
January 2002

Example written by Kevin Conder.

compress

compress — Compress, limit, expand, duck or gate an audio signal.

Description

This unit functions as an audio compressor, limiter, expander, or noise gate, using either soft-knee or hard-knee mapping, and with dynamically variable performance characteristics. It takes two audio input signals, *aasig* and *acsig*, the first of which is modified by a running analysis of the second. Both signals can be the same, or the first can be modified by a different controlling signal.

compress first examines the controlling *acsig* by performing envelope detection. This is directed by two control values *katt* and *krel*, defining the attack and release time constants (in seconds) of the detector. The detector rides the peaks (not the RMS) of the control signal. Typical values are .01 and .1, the latter usually being similar to *ilook*.

The running envelope is next converted to decibels, then passed through a mapping function to determine what compressor action (if any) should be taken. The mapping function is defined by four decibel control values. These are given as positive values, where 0 db corresponds to an amplitude of 1, and 90 db corresponds to an amplitude of 32768.

Syntax

```
ar compress aasig, acsig, kthresh, kloknee, khiknee, kratio, katt, krel, ilook
```

Initialization

ilook -- lookahead time in seconds, by which an internal envelope release can sense what is coming. This induces a delay between input and output, but a small amount of lookahead improves the performance of the envelope detector. Typical value is .05 seconds, sufficient to sense the peaks of the lowest frequency in *acsig*.

Performance

kthresh -- sets the lowest decibel level that will be allowed through. Normally 0 or less, but if higher the threshold will begin removing low-level signal energy such as background noise.

kloknee, *khiknee* -- decibel break-points denoting where compression or expansion will begin. These set the boundaries of a soft-knee curve joining the low-amplitude 1:1 line and the higher-amplitude compression ratio line. Typical values are 48 and 60 db. If the two breakpoints are equal, a hard-knee (angled) map will result.

kratio -- ratio of compression when the signal level is above the knee. The value 2 will advance the output just one decibel for every input gain of two; 3 will advance just one in three; 20 just one in twenty, etc. Inverse ratios will cause signal expansion: .5 gives two for one, .25 four for one, etc. The value 1 will result in no change.

The actions of **compress** will depend on the parameter settings given. A hard-knee compressor-limiter, for instance, is obtained from a near-zero attack time, equal-value break-points, and a very high ratio (say 100). A noise-gate plus expander is obtained from some positive threshold, and a fractional ratio above the knee. A voice-activated music compressor (ducker) will result from feeding the music into *aasig* and the speech into *acsig*. A voice de-esser will result from feeding the voice into both, with the *acsig* version being preceded by a band-pass filter that emphasizes the sibilants. Each application will require some experimentation to find the best parameter settings; these have been made k-variable to make this practical.

Examples

```
aout compress amus, avoc, 0, 40, 60, 3, .1, .5, .02 ; voice-activated compressor  
; with low-level sensitivity
```

Credits

Written by Barry L. Vercoe for Extended Csound and released in csound5.02.

control

control — Configurable slider controls for realtime user input.

Description

Configurable slider controls for realtime user input. Requires Winsound or TCL/TK. *control* reads a slider's value.

Syntax

```
kres control knum
```

Performance

knun -- number of the slider to be read.

Calling *control* will create a new slider on the screen. There is no theoretical limit to the number of sliders. Windows and TCL/TK use only integers for slider values, so the values may need rescaling. GUIs usually pass values at a fairly slow rate, so it may be advisable to pass the output of control through *port*.

Examples

See the *setctrl* opcode for an example.

See Also

setctrl

Credits

Author: John ffitch
University of Bath, Codemist. Ltd.
Bath, UK
May, 2000

New in Csound version 4.06

convle

convle — Same as the convolve opcode.

Description

Same as the *convolve* opcode.

convolve

convolve — Convolves a signal and an impulse response.

Description

Output is the convolution of signal *ain* and the impulse response contained in *ifilcod*. If more than one output signal is supplied, each will be convolved with the same impulse response. Note that it is considerably more efficient to use one instance of the operator when processing a mono input to create stereo, or quad, outputs.

Note: this opcode can also be written as *convle*.

Syntax

```
ar1 [, ar2] [, ar3] [, ar4] convolve ain, ifilcod [, ichannel]
```

Initialization

ifilcod -- integer or character-string denoting an impulse response data file. An integer denotes the suffix of a file *convolve.m*; a character string (in double quotes) gives a filename, optionally a full pathname. If not a fullpath, the file is sought first in the current directory, then in the one given by the environment variable SADIR (if defined). The data file contains the Fourier transform of an impulse response. Memory usage depends on the size of the data file, which is read and held entirely in memory during computation, but which is shared by multiple calls.

ichannel (optional) -- which channel to use from the impulse response data file.

Performance

ain -- input audio signal.

convolve implements Fast Convolution. The output of this operator is delayed with respect to the input. The following formulas should be used to calculate the delay:

```
For (1/kr) <= IRdur:
  Delay = ceil(IRdur * kr) / kr
For (1/kr) > IRdur:
  Delay = IRdur * ceil(1/(kr*IRdur))
Where:
  kr = Csound control rate
  IRdur = duration, in seconds, of impulse response
  ceil(n) = smallest integer not smaller than n
```

One should be careful to also take into account the initial delay, if any, of the impulse response. For example, if an impulse response is created from a recording, the soundfile may not have the initial delay included. Thus, one should either ensure that the soundfile has the correct amount of zero padding at the start, or, preferably, compensate for this delay in the orchestra. (the latter method is more efficient). To compensate for the delay in the orchestra, subtract the initial delay from the result calculated using the above formula(s), when calculating the required delay to introduce into the 'dry' audio path.

For typical applications, such as reverb, the delay will be in the order of 0.5 to 1.5 seconds, or even longer. This renders the current implementation unsuitable for real time applications. It could conceivably be used for real time filtering however, if the number of taps is small enough.

The author intends to create a higher-level operator at some stage, that would mix the wet & dry signals, using the correct amount of delay automatically.

Examples

Create frequency domain impulse response file using the *cvanal utility*:

```
csound -Ucvanal 11_44.wav 11_44.cv
```

Determine duration of impulse response. For high accuracy, determine the number of sample frames in the impulse response soundfile, and then compute the duration with:

$\text{duration} = (\text{sample frames}) / (\text{sample rate of soundfile})$

This is due to the fact that the *sndinfo utility* only reports the duration to the nearest 10ms. If you have a utility that reports the duration to the required accuracy, then you can simply use the reported value directly.

```
sndinfo 11_44.wav
```

length = 60822 samples, sample rate = 44100

Duration = $60822 / 44100 = 1.379\text{s}$.

Determine initial delay, if any, of impulse response. If the impulse response has not had the initial delay removed, then you can skip this step. If it has been removed, then the only way you will know the initial delay is if the information has been provided separately. For this example, let's assume that the initial delay is 60ms. (0.06s)

Determine the required delay to apply to the dry signal, to align it with the convolved signal:

If $kr = 441$:

$1/kr = 0.0023$, which is $\leq IR_{dur}$ (1.379s), so:

$\text{Delay1} = \text{ceil}(IR_{dur} * kr) / kr$
 $= \text{ceil}(608.14) / 441$
 $= 609 / 441$
 $= 1.38\text{s}$

Accounting for the initial delay:

$\text{Delay2} = 0.06\text{s}$
 $\text{Total delay} = \text{delay1} - \text{delay2}$
 $= 1.38 - 0.06$
 $= 1.32\text{s}$

Create .orc file, e.g.:


```
; Simple demonstration of CONVOLVE operator, to apply reverb.
sr = 44100
kr = 441
ksmps = 100
nchnls = 2
instr 1
imix = 0.22 ; Wet/dry mix. Vary as desired.
; NB: 'Small' reverbs often require a much higher
; percentage of wet signal to sound interesting. 'Large'
; reverbs seem require less. Experiment! The wet/dry mix is
; very important - a small change can make a large difference.
ivol = 0.9 ; Overall volume level of reverb. May need to adjust
; when wet/dry mix is changed, to avoid clipping.
idel = 1.32 ; Required delay to align dry audio with output of convolve.
; This can be automatically calculated within the orc file,
; if desired.
adry      soundin "anechoic.wav"      ; input (dry) audio
awet1,awet2 convolve adry,"l1_44.cv"  ; stereo convolved (wet) audio
adrydel   delay (1-imix)*adry,idel    ; Delay dry signal, to align it with
; convolved signal. Apply level
; adjustment here too.
outs      ivol*(adrydel+imix*awet1),ivol*(adrydel+imix*awet2)
; Mix wet & dry signals, and output
endin
```

See also

pconvolve, dconv, cvanal.

Credits

Author: Greg Sullivan

1996

New in version 3.28

COS

cos — Performs a cosine function.

Description

Returns the cosine of x (x in radians).

Syntax

`cos(x)` (no rate restriction)

Examples

Here is an example of the cos opcode. It uses the file *cos.csd* [examples/cos.csd].

Example 85. Example of the cos opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cos.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  irad = 25
  i1 = cos(irad)

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  i1 = 0.991
```

See Also

cosh, *cosinv*, *sin*, *sinh*, *sininv*, *tan*, *tanh*, *taninv*

Credits

Example written by Kevin Conder.

cosh

cosh — Performs a hyperbolic cosine function.

Description

Returns the hyperbolic cosine of x (x in radians).

Syntax

`cosh(x)` (no rate restriction)

Examples

Here is an example of the cosh opcode. It uses the file *cosh.csd* [examples/cosh.csd].

Example 86. Example of the cosh opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cosh.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  irad = 1
  i1 = cosh(irad)

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  i1 = 1.543
```

See Also

cos, cosinv, sin, sinh, sininv, tan, tanh, taninv

Credits

Author: John ffitch

New in version 3.47

Example written by Kevin Conder.

cosinv

cosinv — Performs an arccosine function.

Description

Returns the arccosine of x (x in radians).

Syntax

`cosinv(x)` (no rate restriction)

Examples

Here is an example of the cosinv opcode. It uses the file *cosinv.csd* [examples/cosinv.csd].

Example 87. Example of the cosinv opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cosinv.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  irad = 0.5
  i1 = cosinv(irad)

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  i1 = 1.047
```

See Also

cos, *cosh*, *sin*, *sinh*, *sininv*, *tan*, *tanh*, *taninv*

Credits

Author: John ffitch

New in version 3.48

Example written by Kevin Conder.

cps2pch

`cps2pch` — Converts a pitch-class value into cycles-per-second (Hz) for equal divisions of the octave.

Description

Converts a pitch-class value into cycles-per-second (Hz) for equal divisions of the octave.

Syntax

```
icps cps2pch ipch, iequal
```

Initialization

ipch -- Input number of the form 8ve.pc, indicating an 'octave' and which note in the octave.

iequal -- If positive, the number of equal intervals into which the 'octave' is divided. Must be less than or equal to 100. If negative, is the number of a table of frequency multipliers.



Note

1. The following are essentially the same

```
ia = cpspch(8.02)
ib  cps2pch 8.02, 12
ic  cpsxpch 8.02, 12, 2, 1.02197503906
```

2. These are opcodes not functions
3. Negative values of *ipch* are allowed.

Examples

Here is an example of the `cps2pch` opcode. It uses the file *cps2pch.csd* [examples/cps2pch.csd].

Example 88. Example of the `cps2pch` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cps2pch.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
```



```
nchnls = 1

; Instrument #1.
instr 1
; Use a normal twelve-tone scale.
ipch = 8.02
iequal = 12

icps cps2pch ipch, iequal

print icps
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1: icps = 293.666
```

Here is an example of the `cps2pch` opcode using a table of frequency multipliers. It uses the file `cps2pch_ftable.csd` [examples/cps2pch_ftable.csd].

Example 89. Example of the `cps2pch` opcode using a table of frequency multipliers.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cps2pch_ftable.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
ipch = 8.02

; Use Table #1, a table of frequency multipliers.
icps cps2pch ipch, -1

print icps
endin

</CsInstruments>
<CsScore>

; Table #1: a table of frequency multipliers.
; Creates a 10-note scale of unequal divisions.
f 1 0 16 -2 1 1.1 1.2 1.3 1.4 1.6 1.7 1.8 1.9

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  icps = 313.951
```

Here is an example of the `cps2pch` opcode using a 19ET scale. It uses the file `cps2pch_19et.csd` [examples/cps2pch_19et.csd].

Example 90. Example of the `cps2pch` opcode using a 19ET scale.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cps2pch_19et.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use 19ET scale.
ipch = 8.02
iequal = 19

icps cps2pch ipch, iequal

print icps
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  icps = 281.429
```

See Also

cpspch, *cpsxpch*

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
1997

New in Csound version 3.492

cpsmidi

cpsmidi — Get the note number of the current MIDI event, expressed in cycles-per-second.

Description

Get the note number of the current MIDI event, expressed in cycles-per-second.

Syntax

icps **cpsmidi**

Performance

Get the note number of the current MIDI event, expressed in cycles-per-second units, for local processing.



cpsmidi vs. cpsmidinn

The *cpsmidi* opcode only produces meaningful results in a Midi-activated note (either real-time or from a Midi score with the -F flag). With *cpsmidi*, the Midi note number value is taken from the Midi event that is internally associated with the instrument instance. On the other hand, the *cpsmidinn* opcode may be used in any Csound instrument instance whether it is activated from a Midi event, score event, line event, or from another instrument. The input value for *cpsmidinn* might for example come from a p-field in a textual score or it may have been retrieved from the real-time Midi event that activated the current note using the *notnum* opcode.

Examples

Here is an example of the cpsmidi opcode. It uses the file *cpsmidi.csd* [examples/cpsmidi.csd].

Example 91. Example of the cpsmidi opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o cpsmidi.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 cpsmidi

  print i1
endin
```

```
</CsInstruments>
<CsScore>

; Play Instrument #1 for 12 seconds.
i 1 0 12
e

</CsScore>
</CsoundSynthesizer>
```

See Also

aftouch, ampmidi, cpsmidib, cpstmid, midictrl, notnum, octmidi, octmidib, pchbend, pchmidi, pchmidib, veloc, cpsmidinn, octmidinn, pchmidinn

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

cpsmidib

cpsmidib — Get the note number of the current MIDI event and modify it by the current pitch-bend value, express it in cycles-per-second.

Description

Get the note number of the current MIDI event and modify it by the current pitch-bend value, express it in cycles-per-second.

Syntax

```
icps cpsmidib [irange]
```

```
kcps cpsmidib [irange]
```

Initialization

irange (optional) -- the pitch bend range in semitones.

Performance

Get the note number of the current MIDI event, modify it by the current pitch-bend value, and express the result in cycles-per-second units. Available as an i-time value or as a continuous k-rate value.

Examples

Here is an example of the **cpsmidib** opcode. It uses the file *cpsmidib.csd* [examples/cpsmidib.csd].

Example 92. Example of the **cpsmidib** opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o cpsmidib.wav -W   ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  il cpsmidib

  print il
endin

</CsInstruments>
<CsScore>
```

```
; Play Instrument #1 for 12 seconds.  
i 1 0 12  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

aftouch, ampmidi, cpsmidi, midictrl, notnum, octmidi, octmidib, pchbend, pchmidi, pchmidib, veloc

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

cpsmidinn

`cpsmidinn` — Converts a Midi note number value to cycles-per-second.

Description

Converts a Midi note number value to cycles-per-second.

Syntax

```
cpsmidinn (MidiNoteNumber) (init- or control-rate args only)
```

where the argument within the parentheses may be a further expression.

Performance

cpsmidinn is a function that takes an i-rate or k-rate value representing a Midi note number and returns the equivalent frequency value in cycles-per-second (Hertz). This conversion assumes that Middle C is Midi note number 60 and that Middle A is tuned to 440 Hz. Midi note number values are typically integers in the range from 0 to 127 but fractional values or values outside of this range will be interpreted consistently.



cpsmidinn vs. cpsmidi

The *cpsmidinn* opcode may be used in any Csound instrument instance whether it is activated from a Midi event, score event, line event, or from another instrument. The input value for *cpsmidinn* might for example come from a p-field in a textual score or it may have been retrieved from the real-time Midi event that activated the current note using the *notnum* opcode. You must specify an i-rate or k-rate expression for the Midi note number that is to be converted. On the other hand, the *cpsmidi* opcode only produces meaningful results in a Midi-activated note (either real-time or from a Midi score with the -F flag). With *cpsmidi*, the Midi note number value is taken from the Midi event associated with the instrument instance, and no location or expression for this value may be specified.

cpsmidinn and its related opcodes are really *value converters* with a special function of manipulating pitch data.

Data concerning pitch and frequency can exist in any of the following forms:

Table 7. Pitch and Frequency Values

Name	Abbreviation
octave point pitch-class (8ve.pc)	pch
octave point decimal	oct
cycles per second	cps
Midi note number (0-127)	midinn

The first two forms consist of a whole number, representing octave registration, followed by a specially interpreted fractional part. For *pch*, the fraction is read as two decimal digits representing the 12 equal-tempered pitch classes from .00 for C to .11 for B. For *oct*, the fraction is interpreted as a true decimal fractional part of an octave. The two fractional forms are thus related by the factor 100/12. In both forms, the fraction is preceded by a whole number octave index such that 8.00 rep-

resents Middle C, 9.00 the C above, etc. Midi note number values range between 0 and 127 (inclusively) with 60 representing Middle C, and are usually whole numbers. Thus A440 can be represented alternatively by 440 (*cps*), 69 (*midinn*), 8.09 (*pch*), or 8.75 (*oct*). Microtonal divisions of the *pch* semitone can be encoded by using more than two decimal places.

The mnemonics of the pitch conversion units are derived from morphemes of the forms involved, the second morpheme describing the source and the first morpheme the object (result). Thus *cpspch*(8.09) will convert the pitch argument 8.09 to its *cps* (or Hertz) equivalent, giving the value of 440. Since the argument is constant over the duration of the note, this conversion will take place at i-time, before any samples for the current note are produced.

By contrast, the conversion *cpsoct*(8.75 + *k1*) which gives the value of A440 transposed by the octave interval *k1*. The calculation will be repeated every *k*-period since that is the rate at which *k1* varies.



Note

The conversion from *pch*, *oct*, or *midinn* into *cps* is not a linear operation but involves an exponential process that could be time-consuming when executed repeatedly. Csound now uses a built-in table lookup to do this efficiently, even at audio rates. Because the table index is truncated without interpolation, pitch resolution when using one of these opcodes is limited to 8192 discrete and equal divisions of the octave, and some pitches of the standard 12-tone equally-tempered scale are very slightly mistuned (by at most 0.15 cents).

Examples

Here is an example of the *cpsmidinn* opcode. It uses the file *cpsmidinn.csd* [examples/cpsmidinn.csd].

Example 93. Example of the *cpsmidinn* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform.
; This example produces no audio, so we render in
; non-realtime and turn off sound to disk:
-n
</CsOptions>
<CsInstruments>

instr 1
; i-time loop to print conversion table
imidiNN = 0
loop1:
  icps = cpsmidinn(imidiNN)
  ioct = octmidinn(imidiNN)
  ipch = pchmidinn(imidiNN)

  print imidiNN, icps, ioct, ipch

  imidiNN = imidiNN + 1
  if (imidiNN < 128) igoto loop1
endin

instr 2
; test k-rate converters
kMiddleC = 60
kcps = cpsmidinn(kMiddleC)
koct = octmidinn(kMiddleC)
kpch = pchmidinn(kMiddleC)

printks "%d %f %f %f\n", 1.0, kMiddleC, kcps, koct, kpch
endin

</CsInstruments>
```



```
<CsScore>  
i1 0 0  
i2 0 0.1  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

octmidinn, pchmidinn, cpsmidi, notnum, cpspch, cpsoct, octcps, octpch, pchoct

Credits

Derived from original value converters by Barry Vercoe.

New in version 5.07

cpsoct

`cpsoct` — Converts an octave-point-decimal value to cycles-per-second.

Description

Converts an octave-point-decimal value to cycles-per-second.

Syntax

```
cpsoct (oct) (no rate restriction)
```

where the argument within the parentheses may be a further expression.

Performance

`cpsoct` and its related opcodes are really *value converters* with a special function of manipulating pitch data.

Data concerning pitch and frequency can exist in any of the following forms:

Table 8. Pitch and Frequency Values

Name	Abbreviation
octave point pitch-class (8ve.pc)	pch
octave point decimal	oct
cycles per second	cps
Midi note number (0-127)	midinn

The first two forms consist of a whole number, representing octave registration, followed by a specially interpreted fractional part. For *pch*, the fraction is read as two decimal digits representing the 12 equal-tempered pitch classes from .00 for C to .11 for B. For *oct*, the fraction is interpreted as a true decimal fractional part of an octave. The two fractional forms are thus related by the factor 100/12. In both forms, the fraction is preceded by a whole number octave index such that 8.00 represents Middle C, 9.00 the C above, etc. Midi note number values range between 0 and 127 (inclusively) with 60 representing Middle C, and are usually whole numbers. Thus A440 can be represented alternatively by 440 (*cps*), 69 (*midinn*), 8.09 (*pch*), or 8.75 (*oct*). Microtonal divisions of the *pch* semitone can be encoded by using more than two decimal places.

The mnemonics of the pitch conversion units are derived from morphemes of the forms involved, the second morpheme describing the source and the first morpheme the object (result). Thus `cpspch(8.09)` will convert the pitch argument 8.09 to its *cps* (or Hertz) equivalent, giving the value of 440. Since the argument is constant over the duration of the note, this conversion will take place at i-time, before any samples for the current note are produced.

By contrast, the conversion `cpsoct(8.75 + k1)` which gives the value of A440 transposed by the octave interval *k1*. The calculation will be repeated every k-period since that is the rate at which *k1* varies.



Note

The conversion from *pch*, *oct*, or *midinn* into *cps* is not a linear operation but involves an exponential process that could be time-consuming when executed repeatedly. Csound now uses a built-in table lookup to do this efficiently, even at audio rates. Be-

cause the table index is truncated without interpolation, pitch resolution when using one of these opcodes is limited to 8192 discrete and equal divisions of the octave, and some pitches of the standard 12-tone equally-tempered scale are very slightly mistuned (by at most 0.15 cents).

Examples

Here is an example of the `cpsoct` opcode. It uses the file `cpsoct.csd` [examples/cpsoct.csd].

Example 94. Example of the `cpsoct` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cpsoct.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Convert an octave-point-decimal value into a
; cycles-per-second value.
ioct = 8.75
icps = cpsoct(ioct)

print icps
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  icps = 440.000
```

See Also

cpspch, *octcps*, *octpch*, *pchoct*, *cpsmidinn*, *octmidinn*, *pchmidinn*

Credits

Example written by Kevin Conder.

cpspch

cpspch — Converts a pitch-class value to cycles-per-second.

Description

Converts a pitch-class value to cycles-per-second.

Syntax

cpspch (pch) (init- or control-rate args only)

where the argument within the parentheses may be a further expression.

Performance

cpspch and its related opcodes are really *value converters* with a special function of manipulating pitch data.

Data concerning pitch and frequency can exist in any of the following forms:

Table 9. Pitch and Frequency Values

Name	Abbreviation
octave point pitch-class (8ve.pc)	pch
octave point decimal	oct
cycles per second	cps
Midi note number (0-127)	midinn

The first two forms consist of a whole number, representing octave registration, followed by a specially interpreted fractional part. For *pch*, the fraction is read as two decimal digits representing the 12 equal-tempered pitch classes from .00 for C to .11 for B. For *oct*, the fraction is interpreted as a true decimal fractional part of an octave. The two fractional forms are thus related by the factor 100/12. In both forms, the fraction is preceded by a whole number octave index such that 8.00 represents Middle C, 9.00 the C above, etc. Midi note number values range between 0 and 127 (inclusively) with 60 representing Middle C, and are usually whole numbers. Thus A440 can be represented alternatively by 440 (*cps*), 69 (*midinn*), 8.09 (*pch*), or 8.75 (*oct*). Microtonal divisions of the *pch* semitone can be encoded by using more than two decimal places.

The mnemonics of the pitch conversion units are derived from morphemes of the forms involved, the second morpheme describing the source and the first morpheme the object (result). Thus *cpspch*(8.09) will convert the pitch argument 8.09 to its *cps* (or Hertz) equivalent, giving the value of 440. Since the argument is constant over the duration of the note, this conversion will take place at i-time, before any samples for the current note are produced.

By contrast, the conversion *cpsoct*(8.75 + k1) which gives the value of A440 transposed by the octave interval *k1*. The calculation will be repeated every k-period since that is the rate at which *k1* varies.



Note

The conversion from *pch*, *oct*, or *midinn* into *cps* is not a linear operation but involves an exponential process that could be time-consuming when executed repeatedly. Csound now uses a built-in table lookup to do this efficiently, even at audio rates. Be-

cause the table index is truncated without interpolation, pitch resolution when using one of these opcodes is limited to 8192 discrete and equal divisions of the octave, and some pitches of the standard 12-tone equally-tempered scale are very slightly mistuned (by at most 0.15 cents).

Examples

Here is an example of the `cpspch` opcode. It uses the file `cpspch.csd` [examples/cpspch.csd].

Example 95. Example of the `cpspch` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cpspch.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Convert a pitch-class value into a
; cycles-per-second value.
ipch = 8.09
icps = cpspch(ipch)

print icps
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  icps = 440.000
```

See Also

cps2pch, *cpsoct*, *cpsxpch*, *octcps*, *octpch*, *pchoct*, *cpsmidinn*, *octmidinn*, *pchmidinn*

Credits

Example written by Kevin Conder.

cpstmid

cpstmid — Get a MIDI note number (allows customized micro-tuning scales).

Description

This unit is similar to *cpsmidi*, but allows fully customized micro-tuning scales.

Syntax

```
icps cpstmid ifn
```

Initialization

ifn -- function table containing the parameters (*numgrades*, *interval*, *basefreq*, *basekeymidi*) and the tuning ratios.

Performance

Init-rate only

cpsmid requires five parameters, the first, *ifn*, is the function table number of the tuning ratios, and the other parameters must be stored in the function table itself. The function table *ifn* should be generated by *GEN02*, with normalization inhibited. The first four values stored in this function are:

1. *numgrades* -- the number of grades of the micro-tuning scale
2. *interval* -- the frequency range covered before repeating the grade ratios, for example 2 for one octave, 1.5 for a fifth etc.
3. *basefreq* -- the base frequency of the scale in Hz
4. *basekeymidi* -- the MIDI note number to which *basefreq* is assigned unmodified

After these four values, the user can begin to insert the tuning ratios. For example, for a standard 12 note scale with the base frequency of 261 Hz assigned to the key number 60, the corresponding f-statement in the score to generate the table should be:

```
;      numgrades interval basefreq basekeymidi tuning ratios (equal temp)
f1 0 64 -2 12    2    261    60    1 1.059463094359 1.122462048309 1.189207115003 ..etc...
```

Another example with a 24 note scale with a base frequency of 440 assigned to the key number 48, and a repetition interval of 1.5:

```
;      numgrades interval basefreq basekeymidi tuning-ratios (equal temp)
f1 0 64 -2 24    1.5    440    48    1 1.01 1.02 1.03 ..etc...
```

Examples

Here is an example of the `cpstmidi` opcode. It uses the file `cpstmidi.csd` [examples/cpstmidi.csd].

Example 96. Example of the `cpstmidi` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d          -MO    ;;RT audio I/O with MIDI in
; For Non-realtime output leave only the line below:
; -o cpstmidi.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Table #1, a normal 12-tone equal temperament scale.
; numgrades = 12 (twelve tones)
; interval = 2 (one octave)
; basefreq = 261.659 (Middle C)
; basekeymidi = 60 (Middle C)
gitemp ftgen 1, 0, 64, -2, 12, 2, 261.659, 60, 1.00, \
          1.059, 1.122, 1.189, 1.260, 1.335, 1.414, \
          1.498, 1.588, 1.682, 1.782, 1.888, 2.000

; Instrument #1.
instr 1
; Use Table #1.
ifn = 1
il cpstmidi ifn

print il
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 12 seconds.
i 1 0 12
e

</CsScore>
</CsoundSynthesizer>
```

See Also

cpsmidi, GEN02

Credits

Author: Gabriel Maldonado
Italy
1998

Example written by Kevin Conder.

New in Csound version 3.492

cpstun

cpstun — Returns micro-tuning values at k-rate.

Description

Returns micro-tuning values at k-rate.

Syntax

kcps **cpstun** ktrig, kindex, kfn

Performance

kcps -- Return value in cycles per second.

ktrig -- A trigger signal used to trigger the evaluation.

kindex -- An integer number denoting an index of scale.

kfn -- Function table containing the parameters (numgrades, interval, basefreq, basekeymidi) and the tuning ratios.

These opcodes are similar to *cpstmid*, but work without necessity of MIDI.

cpstun works at k-rate. It allows fully customized micro-tuning scales. It requires a function table number containing the tuning ratios, and some other parameters stored in the function table itself.

kindex arguments should be filled with integer numbers expressing the grade of given scale to be converted in cps. In *cpstun*, a new value is evaluated only when *ktrig* contains a non-zero value. The function table *kfn* should be generated by *GEN02* and the first four values stored in this function are parameters that express:

- numgrades -- The number of grades of the micro-tuning scale.
- interval -- The frequency range covered before repeating the grade ratios, for example 2 for one octave, 1.5 for a fifth etcetera.
- basefreq -- The base frequency of the scale in cycles per second.
- basekey -- The integer index of the scale to which to assign basefreq unmodified.

After these four values, the user can begin to insert the tuning ratios. For example, for a standard 12-grade scale with the base-frequency of 261 cps assigned to the key-number 60, the corresponding f-statement in the score to generate the table should be:

```
;          numgrades  basefreq  tuning-ratios (eq.temp) .....  
;          interval   basekey  
f1 0 64 -2 12      2      261    60    1    1.059463 1.12246 1.18920 ..etc...
```

Another example with a 24-grade scale with a base frequency of 440 assigned to the key-number 48, and a repetition interval of 1.5:


```
;               numgrades      basefreq      tuning-ratios .....
;               interval      basekey
f1 0 64 -2      24      1.5      440      48      1      1.01 1.02 1.03      ..etc...
```

Examples

Here is an example of the `cpstun` opcode. It uses the file `cpstun.csd` [examples/cpstun.csd].

Example 97. Example of the `cpstun` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cpstun.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Table #1, a normal 12-tone equal temperament scale.
; numgrades = 12 (twelve tones)
; interval = 2 (one octave)
; basefreq = 261.659 (Middle C)
; basekeymidi = 60 (Middle C)
gitemp ftgen 1, 0, 64, -2, 12, 2, 261.659, 60, 1.00, \
          1.059, 1.122, 1.189, 1.260, 1.335, 1.414, \
          1.498, 1.588, 1.682, 1.782, 1.888, 2.000

; Instrument #1.
instr 1
; Set the trigger.
ktrig init 1

; Use Table #1.
kfn init 1

; If the base key (note #60) is C, then 9 notes
; above it (note #60 + 9 = note #69) should be A.
kindex init 69

k1 cpstun ktrig, kindex, kfn

printk2 k1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
i1      440.11044
```

See Also

cpstmid, *cpstuni*, *GEN02*

Credits

Example written by Kevin Conder.

cpstuni

cpstuni — Returns micro-tuning values at init-rate.

Description

Returns micro-tuning values at init-rate.

Syntax

```
icps cpstuni index, ifn
```

Initialization

icps -- Return value in cycles per second.

index -- An integer number denoting an index of scale.

ifn -- Function table containing the parameters (numgrades, interval, basefreq, basekeymidi) and the tuning ratios.

Performance

These opcodes are similar to *cpstmid*, but work without necessity of MIDI.

cpstuni works at init-rate. It allows fully customized micro-tuning scales. It requires a function table number containing the tuning ratios, and some other parameters stored in the function table itself.

The *index* argument should be filled with integer numbers expressing the grade of given scale to be converted in cps. The function table *ifn* should be generated by *GEN02* and the first four values stored in this function are parameters that express:

- numgrades -- The number of grades of the micro-tuning scale.
- interval -- The frequency range covered before repeating the grade ratios, for example 2 for one octave, 1.5 for a fifth etcetera.
- basefreq -- The base frequency of the scale in cycles per second.
- basekey -- The integer index of the scale to which to assign basefreq unmodified.

After these four values, the user can begin to insert the tuning ratios. For example, for a standard 12-grade scale with the base-frequency of 261 cps assigned to the key-number 60, the corresponding f-statement in the score to generate the table should be:

```
;          numgrades  basefreq  tuning-ratios (eq.temp) .....  
;          interval   basekey  
f1 0 64 -2 12      2      261   60   1   1.059463 1.12246 1.18920 ..etc...
```

Another example with a 24-grade scale with a base frequency of 440 assigned to the key-number 48, and a repetition interval of 1.5:

```
;          numgrades  basefreq  tuning-ratios .....
```

```
;
f1 0 64 -2          24          interval      basekey
1.5          440      48          1      1.01  1.02  1.03  ..etc...
```

Examples

Here is an example of the `cpstuni` opcode. It uses the file `cpstuni.csd` [examples/cpstuni.csd].

Example 98. Example of the `cpstuni` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac            -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cpstuni.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Table #1, a normal 12-tone equal temperament scale.
; numgrades = 12 (twelve tones)
; interval = 2 (one octave)
; basefreq = 261.659 (Middle C)
; basekeymidi = 60 (Middle C)
gitemp ftgen 1, 0, 64, -2, 12, 2, 261.659, 60, 1.00, \
1.059, 1.122, 1.189, 1.260, 1.335, 1.414, \
1.498, 1.588, 1.682, 1.782, 1.888, 2.000

; Instrument #1.
instr 1
; Use Table #1.
ifn = 1

; If the base key (note #60) is C, then 9 notes
; above it (note #60 + 9 = note #69) should be A.
index = 69

il cpstuni index, ifn

print il
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  il = 440.110
```

See Also

cpstmid, *cpstun*, *GEN02*

Credits

Written by Gabriel Maldonado.

Example written by Kevin Conder.

cpsxpch

cpsxpch — Converts a pitch-class value into cycles-per-second (Hz) for equal divisions of any interval.

Description

Converts a pitch-class value into cycles-per-second (Hz) for equal divisions of any interval. There is a restriction of no more than 100 equal divisions.

Syntax

icps **cpsxpch** ipch, iequal, irepeat, ibase

Initialization

ipch -- Input number of the form 8ve.pc, indicating an 'octave' and which note in the octave.

iequal -- if positive, the number of equal intervals into which the 'octave' is divided. Must be less than or equal to 100. If negative, is the number of a table of frequency multipliers.

irepeat -- Number indicating the interval which is the 'octave.' The integer 2 corresponds to octave divisions, 3 to a twelfth, 4 is two octaves, and so on. This need not be an integer, but must be positive.

ibase -- The frequency which corresponds to pitch 0.0



Note

1. The following are essentially the same

```
ia = cpspch(8.02)
ib  cps2pch 8.02, 12
ic  cpsxpch 8.02, 12, 2, 1.02197503906
```

2. These are opcodes not functions
3. Negative values of *ipch* are allowed, but not negative *irepeat*, *iequal* or *ibase*.

Examples

Here is an example of the cpsxpch opcode. It uses the file *cpsxpch.csd* [examples/cpsxpch.csd].

Example 99. Example of the cpsxpch opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
```

```
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cpsxpch.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a normal twelve-tone scale.
ipch = 8.02
iequal = 12
irepeat = 2
ibase = 1.02197503906

icps cpsxpch ipch, iequal, irepeat, ibase

print icps
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  icps = 293.666
```

Here is an example of the cpsxpch opcode using a 10.5 ET scale. It uses the file *cpsxpch_105et.csd* [examples/cpsxpch_105et.csd].

Example 100. Example of the cpsxpch opcode using a 10.5 ET scale.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cpsxpch_105et.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a 10.5ET scale.
ipch = 4.02
iequal = 21
irepeat = 4
ibase = 16.35160062496

icps cpsxpch ipch, iequal, irepeat, ibase

print icps
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
```

```
i 1 0 1
e
```

```
</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  icps = 4776.824
```

Here is an example of the `cpsxpch` opcode using a Pierce scale centered on middle A. It uses the file `cpsxpch_pierce.csd` [examples/cpsxpch_pierce.csd].

Example 101. Example of the `cpsxpch` opcode using a Pierce scale centered on middle A.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cpsxpch_pierce.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a Pierce scale centered on middle A.
ipch = 2.02
iequal = 12
irepeat = 3
ibase = 261.62561

icps cpsxpch ipch, iequal, irepeat, ibase

print icps
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  icps = 2827.762
```

See Also

cpspch, *cps2pch*

Credits

Author: John ffitch
University of Bath/Codemist Ltd.

Bath, UK
1997

New in Csound version 3.492

cpuprc

cpuprc — Control allocation of cpu resources on a per-instrument basis, to optimize realtime output.

Description

Control allocation of cpu resources on a per-instrument basis, to optimize realtime output.

Syntax

```
cpuprc insnum, ipercent
```

Initialization

insnum -- instrument number

ipercent -- percent of cpu processing-time to assign. Can also be expressed as a fractional value.

Performance

cpuprc sets the cpu processing-time percent usage of an instrument, in order to avoid buffer under-run in realtime performances, enabling a sort of polyphony threshold. The user must set *ipercent* value for each instrument to be activated in realtime. Assuming that the total theoretical processing time of the cpu of the computer is 100%, this percent value can only be defined empirically, because there are too many factors that contribute to limiting realtime polyphony in different computers.

For example, if *ipercent* is set to 5% for instrument 1, the maximum number of voices that can be allocated in realtime, is 20 ($5\% * 20 = 100\%$). If the user attempts to play a further note while the 20 previous notes are still playing, Csound inhibits the allocation of that note and will display the following warning message:

```
can't allocate last note because it exceeds 100% of cpu time
```

In order to avoid audio buffer underruns, it is suggested to set the maximum number of voices slightly lower than the real processing power of the computer. Sometimes an instrument can require more processing time than normal. If, for example, the instrument contains an oscillator which reads a table that doesn't fit in cache memory, it will be slower than normal. In addition, any program running concurrently in multitasking, can subtract processing power to varying degrees.

At the start, all instruments are set to a default value of *ipercent* = 0.0% (i.e. zero processing time or rather infinite cpu processing-speed). This setting is OK for deferred-time sessions.

All instances of *cpuprc* must be defined in the header section, not in the instrument body.

Examples

Here is an example of the *cpuprc* opcode. It uses the file *cpuprc.csd* [examples/cpuprc.csd].

Example 102. Example of the cpuprc opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cpuprc.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Limit Instrument #1 to 5% of the CPU processing time.
cpuprc 1, 5

; Instrument #1
instr 1
  al oscil 10000, 440, 1
  out al
endin

</CsInstruments>
<CsScore>

; Just generate a nice, ordinary sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

maxalloc, prealloc

Credits

Author: Gabriel Maldonado
Italy
July, 1999

Example written by Kevin Conder.

New in Csound version 3.57

cross2

cross2 — Cross synthesis using FFT's.

Description

This is an implementation of cross synthesis using FFT's.

Syntax

```
ares cross2 ain1, ain2, isize, ioverlap, iwin, kbias
```

Initialization

isize -- This is the size of the FFT to be performed. The larger the size the better the frequency response but a sloppy time response.

ioverlap -- This is the overlap factor of the FFT's, must be a power of two. The best settings are 2 and 4. A big overlap takes a long time to compile.

iwin -- This is the function table that contains the window to be used in the analysis. One can use the *GEN20* routine to create this window.

Performance

ain1 -- The stimulus sound. Must have high frequencies for best results.

ain2 -- The modulating sound. Must have a moving frequency response (like speech) for best results.

kbias -- The amount of cross synthesis. 1 is the normal, 0 is no cross synthesis.

Examples

Here is an example of the cross2 opcode. It uses the file *cross2.csd* [examples/cross2.csd] and *beats.wav* [examples/beats.wav].

Example 103. Example of the cross2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cross2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - Play an audio file.
```

```
instr 1
; Use the "beats.wav" audio file.
aout soundin "beats.wav"
out aout
endin

; Instrument #2 - Cross-synthesize!
instr 2
; Use the "ahh" sound stored in Table #1.
ain1 loscil 30000, 1, 1, 1
; Use the "beats.wav" audio file.
ain2 soundin "beats.wav"

isize = 4096
ioverlap = 2
iwin = 2
kbias init 1

aout cross2 ain1, ain2, isize, ioverlap, iwin, kbias

out aout
endin

</CsInstruments>
<CsScore>

; Table #1: An audio file.
f 1 0 128 1 "ahh.aiff" 0 4 0
; Table #2: A windowing function.
f 2 0 2048 20 2

; Play Instrument #1 for 2 seconds.
i 1 0 2
; Play Instrument #2 for 2 seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Paris Smaragdis
MIT, Cambridge
1997

Example written by Kevin Conder.

crunch

crunch — Semi-physical model of a crunch sound.

Description

crunch is a semi-physical model of a crunch sound. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares crunch iamp, idettack [, inum] [, idamp] [, imaxshake]
```

Initialization

iamp -- Amplitude of output. Note: As these instruments are stochastic, this is only a approximation.

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 7.

idamp (optional) -- the damping factor, as part of this equation:

$$\text{damping_amount} = 0.998 + (\text{idamp} * 0.002)$$

The default *damping_amount* is 0.99806 which means that the default value of *idamp* is 0.03. The maximum *damping_amount* is 1.0 (no damping). This means the maximum value for *idamp* is 1.0.

The recommended range for *idamp* is usually below 75% of the maximum value.

imaxshake (optional) -- amount of energy to add back into the system. The value should be in range 0 to 1.

Examples

Here is an example of the crunch opcode. It uses the file *crunch.csd* [examples/crunch.csd].

Example 104. Example of the crunch opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o crunch.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

;orchestra -----

sr =          44100
kr =          4410
ksmps =       10
nchnls =      1
```

```
instr 01                                ;an example of a crunch
al crunch p4, 0.01
    out al
endin

</CsInstruments>
<CsScore>

;score -----

i1 0 1 26000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

cabasa, sandpaper, sekere, stix

Credits

Author: Perry Cook, part of the PhOLIES (Physically-Oriented Library of Imitated Environmental Sounds)

Adapted by John fitch

University of Bath, Codemist Ltd.

Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

ctrl14

ctrl14 — Allows a floating-point 14-bit MIDI signal scaled with a minimum and a maximum range.

Description

Allows a floating-point 14-bit MIDI signal scaled with a minimum and a maximum range.

Syntax

```
idest ctrl14 ichan, ictlno1, ictlno2, imin, imax [, ifn]
```

```
kdest ctrl14 ichan, ictlno1, ictlno2, kmin, kmax [, ifn]
```

Initialization

idest -- output signal

ichan -- MIDI channel number (1-16)

ictlno -- most-significant byte controller number (0-127)

ictlno2 -- least-significant byte controller number (0-127)

imin -- user-defined minimum floating-point value of output

imax -- user-defined maximum floating-point value of output

ifn (optional) -- table to be read when indexing is required. Table must be normalized. Output is scaled according to *imax* and *imin* val.

Performance

kdest -- output signal

kmin -- user-defined minimum floating-point value of output

kmax -- user-defined maximum floating-point value of output

ctrl14 (i- and k-rate 14 bit MIDI control) allows a floating-point 14-bit MIDI signal scaled with a minimum and a maximum range. The minimum and maximum values can be varied at k-rate. It can use optional interpolated table indexing. It requires two MIDI controllers as input.

ctrl14 differs from *midic14* because it can be included in score-oriented instruments without Csound crashes. It needs the additional parameter *ichan* containing the MIDI channel of the controller. MIDI channel is the same for all the controllers used in a single *ctrl14* opcode.

See Also

ctrl7, *ctrl21*, *initc7*, *initc14*, *initc21*, *midic7*, *midic14*, *midic21*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

ctrl21

ctrl21 — Allows a floating-point 21-bit MIDI signal scaled with a minimum and a maximum range.

Description

Allows a floating-point 21-bit MIDI signal scaled with a minimum and a maximum range.

Syntax

```
idest ctrl21 ichan, ictlno1, ictlno2, ictlno3, imin, imax [, ifn]
```

```
kdest ctrl21 ichan, ictlno1, ictlno2, ictlno3, kmin, kmax [, ifn]
```

Initialization

idest -- output signal

ichan -- MIDI channel number (1-16)

ictlno1 -- most-significant byte controller number (0-127)

ictlno2 -- mid-significant byte controller number (0-127)

ictlno3 -- least-significant byte controller number (0-127)

imin -- user-defined minimum floating-point value of output

imax -- user-defined maximum floating-point value of output

ifn (optional) -- table to be read when indexing is required. Table must be normalized. Output is scaled according to *imax* and *imin* val.

Performance

kdest -- output signal

kmin -- user-defined minimum floating-point value of output

kmax -- user-defined maximum floating-point value of output

ctrl21 (i- and k-rate 21 bit MIDI control) allows a floating-point 21-bit MIDI signal scaled with a minimum and a maximum range. Minimum and maximum values can be varied at k-rate. It can use optional interpolated table indexing. It requires three MIDI controllers as input.

ctrl21 differs from *midic21* because it can be included in score oriented instruments without Csound crashes. It needs the additional parameter *ichan* containing the MIDI channel of the controller. MIDI channel is the same for all the controllers used in a single *ctrl21* opcode.

See Also

ctrl7, *ctrl14*, *initc7*, *initc14*, *initc21*, *midic7*, *midic14*, *midic21*

Credits

Author: Gabriel Maldonado

Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

ctrl7

ctrl7 — Allows a floating-point 7-bit MIDI signal scaled with a minimum and a maximum range.

Description

Allows a floating-point 7-bit MIDI signal scaled with a minimum and a maximum range.

Syntax

```
idest ctrl7 ichan, ictlno, imin, imax [, ifn]

kdest ctrl7 ichan, ictlno, kmin, kmax [, ifn]

adest ctrl7 ichan, ictlno, kmin, kmax [, ifn] [, icutoff]
```

Initialization

idest -- output signal

ichan -- MIDI channel (1-16)

ictlno -- MIDI controller number (0-127)

imin -- user-defined minimum floating-point value of output

imax -- user-defined maximum floating-point value of output

ifn (optional) -- table to be read when indexing is required. Table must be normalized. Output is scaled according to *imax* and *imin* val.

icutoff (optional) -- low pass filter cut-off frequency for smoothing a-rate output.

Performance

kdest, *adest* -- output signal

kmin -- user-defined minimum floating-point value of output

kmax -- user-defined maximum floating-point value of output

ctrl7 (i- and k-rate 7 bit MIDI control) allows a floating-point 7-bit MIDI signal scaled with a minimum and a maximum range. It also allows optional non-interpolated table indexing. Minimum and maximum values can be varied at k-rate.

ctrl7 differs from *midic7* because it can be included in score-oriented instruments without Csound crashes. It also needs the additional parameter *ichan* containing the MIDI channel of the controller.

The a-rate version of *ctrl7* outputs an a-rate variable, which is low-pass filtered (smoothed). It contains an optional *icutoff* parameter, to set the cutoff frequency for the low-pass filter. The default is 5.

See Also

ctrl14, *ctrl21*, *initc7*, *initc14*, *initc21*, *midic7*, *midic14*, *midic21*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

The a-rate version of *ctrl7* was added in version 5.06

ctrlinit

ctrlinit — Sets the initial values for a set of MIDI controllers.

Description

Sets the initial values for a set of MIDI controllers.

Syntax

```
ctrlinit ichnl, ictlno1, ival1 [, ictlno2] [, ival2] [, ictlno3] \  
        [, ival3] [...ival32]
```

Initialization

ichnl -- MIDI channel number (1-16)

ictlno1, *ictlno1*, etc. -- MIDI controller numbers (0-127)

ival1, *ival2*, etc. -- initial value for corresponding MIDI controller number

Performance

Sets the initial values for a set of MIDI controllers.

See Also

massign

Credits

Author: Barry L. Vercoe - Mike Berry
MIT, Cambridge, Mass.

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

cuserrnd

cuserrnd — Continuous USER-defined-distribution RaNDom generator.

Description

Continuous USER-defined-distribution RaNDom generator.

Syntax

```
aout cuserrnd kmin, kmax, ktableNum
```

```
iout cuserrnd imin, imax, itableNum
```

```
kout cuserrnd kmin, kmax, ktableNum
```

Initialization

imin -- minimum range limit

imax -- maximum range limit

itableNum -- number of table containing the random-distribution function. Such table is generated by the user. See GEN40, GEN41, and GEN42. The table length does not need to be a power of 2

Performance

ktableNum -- number of table containing the random-distribution function. Such table is generated by the user. See GEN40, GEN41, and GEN42. The table length does not need to be a power of 2

kmin -- minimum range limit

kmax -- maximum range limit

cuserrnd (continuous user-defined-distribution random generator) generates random values according to a continuous random distribution created by the user. In this case the shape of the distribution histogram can be drawn or generated by any GEN routine. The table containing the shape of such histogram must then be translated to a distribution function by means of GEN40 (see GEN40 for more details). Then such function must be assigned to the XtableNum argument of cuserrnd. The output range can then be rescaled according to the Xmin and Xmax arguments. cuserrnd linearly interpolates between table elements, so it is not recommended for discrete distributions (GEN41 and GEN42).

For a tutorial about random distribution histograms and functions see:

- D. Lorrain. "A panoply of stochastic cannons". In C. Roads, ed. 1989. Music machine. Cambridge, Massachusetts: MIT press, pp. 351 - 379.

See Also

duserrnd, *urd*

Credits

Author: Gabriel Maldonado

New in Version 4.16

dam

dam — A dynamic compressor/expander.

Description

This opcode dynamically modifies a gain value applied to the input sound *ain* by comparing its power level to a given threshold level. The signal will be compressed/expanded with different factors regarding that it is over or under the threshold.

Syntax

```
ares dam asig, kthreshold, icompl, icmp2, irtime, iftime
```

Initialization

icompl -- compression ratio for upper zone.

icmp2 -- compression ratio for lower zone

irtime -- gain rise time in seconds. Time over which the gain factor is allowed to raise of one unit.

iftime -- gain fall time in seconds. Time over which the gain factor is allowed to decrease of one unit.

Performance

asig -- input signal to be modified

kthreshold -- level of input signal which acts as the threshold. Can be changed at k-time (e.g. for ducking)

Note on the compression factors: A compression ratio of one leaves the sound unchanged. Setting the ratio to a value smaller than one will compress the signal (reduce its volume) while setting the ratio to a value greater than one will expand the signal (augment its volume).

Examples

Because the results of the *dam* opcode can be subtle, I recommend looking at them in a graphical audio editor program like *audacity*. *audacity* is available for Linux, Windows, and the MacOS and may be downloaded from <http://audacity.sourceforge.net> [<http://audacity.sourceforge.net/>].

Here is an example of the *dam* opcode. It uses the file *dam.csd* [examples/dam.csd], and *beats.wav* [examples/beats.wav].

Example 105. An example of the *dam* opcode compressing an audio signal.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
;-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave the line below:
```

```
; -o dam.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1, uncompressed signal.
instr 1
; Use the "beats.wav" audio file.
asig soundin "beats.wav"

out asig
endin

; Instrument #2, compressed signal.
instr 2
; Use the "beats.wav" audio file.
asig soundin "beats.wav"

; Compress the audio signal.
kthreshold = 25000
icomp1 = 0.5
icomp2 = 0.763
irtime = 0.1
iftime = 0.1
a1 dam asig, kthreshold, icomp1, icomp2, irtime, iftime

out a1
endin

; Instrument #3, compressed signal.
instr 3
; Use the "beats.wav" audio file.
asig soundin "beats.wav"

; Compress the audio signal.
kthreshold line 25000, p3, 4410000
icomp1 = 0.5
icomp2 = 0.763
irtime = 0.1
iftime = 0.1
a1 dam asig, kthreshold, icomp1, icomp2, irtime, iftime

out a1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 2 seconds.
i 1 0 2
; Play Instrument #2 for 2 seconds.
i 2 2 2
; Play Instrument #3 for 2 seconds.
i 3 4 2
e

</CsScore>
</CsoundSynthesizer>
```

This example compresses the audio file “beats.wav”. You should hear a drum pattern repeat twice. The second time, the sound should be quieter (compressed) than the first.

Here is another example of the dam opcode. It uses the file *dam_expanded.csd* [examples/dam_expanded.csd], and *mary.wav* [examples/mary.wav].

Example 106. An example of the dam opcode expanding an audio signal.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
```

```
; -o dam_expanded.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1, normal audio signal.
instr 1
; Use the "mary.wav" audio file.
asig soundin "mary.wav"

out asig
endin

; Instrument #2, expanded audio signal.
instr 2
; Use the "mary.wav" audio file.
asig soundin "mary.wav"

; Expand the audio signal.
kthreshold init 7500
icompl = 2.25
icomp2 = 2.25
irtime = 0.1
iftime = 0.6
a1 dam asig, kthreshold, icomp1, icomp2, irtime, iftime

out a1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1.
i 1 0.0 3.5
; Play Instrument #2.
i 2 3.5 3.5
e

</CsScore>
</CsoundSynthesizer>
```

This example expands the audio file “mary.wav”. You should hear a melody repeat twice. The second time, the sound should be louder (expanded) than the first.

Credits

Author: Marc Resibois
Belgium
1997

New in version 3.47

Examples written by Kevin Conder.

date

date — Returns the number seconds since 1 January 1970.

Description

Returns the number seconds since 1 January 1970, using the operating system's clock.

Syntax

```
ir date
```

Initialization

ir -- value at i-time, of the system clock in seconds since the start of the epoch.

Examples

Here is an example of the date opcode. It uses the file *date.csd* [examples/date.csd].

Example 107. Example of the date opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o date.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
instr 1
  ii date
  print ii
  Sa dates ii
  prints Sa
  Ss dates -1
  prints Ss
  St dates 1
  prints St
endin
</CsInstruments>
<CsScore>
i 1 0 1
e
</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  ii = 1165665152.000
Sat Dec  9 11:52:32 2006
Sat Dec  9 11:51:46 2006
Thu Jan  1 01:00:01 1970
```

See Also

dates

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
December 2006

New in Csound version 5.05

dates

`dates` — Returns as a string the date and time specified.

Description

Returns as a string the date and time specified.

Syntax

```
Sir dates [ itime]
```

Initialization

itime -- the time is seconds since the start of the epoch. If omitted or negative the current time is taken.

Sir -- the date and time as a string.

Examples

Here is an example of the `dates` opcode. It uses the file *date.csd* [examples/date.csd].

Example 108. Example of the `dates` opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o date.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
instr 1
  ii date
  print ii
  Sa dates ii
  prints Sa
  Ss dates -1
  prints Ss
  St dates 1
  prints St
endin
</CsInstruments>
<CsScore>
i 1 0 1
e
</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  ii = 1165665152.000
Sat Dec  9 11:52:32 2006
Sat Dec  9 11:51:46 2006
Thu Jan  1 01:00:01 1970
```

See Also

date

Credits

Author: John fitch
University of Bath/Codemist Ltd.
Bath, UK
December 2006

New in Csound version 5.05

db

db — Returns the amplitude equivalent for a given decibel amount.

Description

Returns the amplitude equivalent for a given decibel amount. This opcode is the same as *ampdb*.

Syntax

db(*x*)

This function works at a-rate, i-rate, and k-rate.

Initialization

x -- a value expressed in decibels.

Performance

Returns the amplitude for a given decibel amount.

Examples

Here is an example of the db opcode. It uses the file *db.csd* [examples/db.csd].

Example 109. Example of the db opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o db.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Calculate the amplitude of 40 decibels.
idecibels = 40
iamp = db(idecibels)

print iamp
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e
```



```
</CsScore>  
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  iamp = 100.000
```

See Also

ampdb, cent, octave, semitone

Credits

Example written by Kevin Conder.

New in version 4.16

dbamp

dbamp — Returns the decibel equivalent of the raw amplitude x .

Description

Returns the decibel equivalent of the raw amplitude x .

Syntax

dbamp(x) (init-rate or control-rate args only)

Examples

Here is an example of the dbamp opcode. It uses the file *dbamp.csd* [examples/dbamp.csd].

Example 110. Example of the dbamp opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac        -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o dbamp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  iamp = 30000
  idb = dbamp(iamp)

  print idb
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  idb = 89.542
```

See Also

ampdb, *ampdbfs*, *dbfsamp*

Credits

Example written by Kevin Conder.

dbfsamp

dbfsamp — Returns the decibel equivalent of the raw amplitude x , relative to full scale amplitude.

Description

Returns the decibel equivalent of the raw amplitude x , relative to full scale amplitude. Full scale is assumed to be 16 bit. New in Csound version 4.10.

Syntax

dbfsamp(x) (init-rate or control-rate args only)

Examples

Here is an example of the dbfsamp opcode. It uses the file *dbfsamp.csd* [examples/dbfsamp.csd].

Example 111. Example of the dbfsamp opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o dbfsamp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  iamp = 30000
  idb = dbfsamp(iamp)

  print idb
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1: idb = -0.767
```

See Also

ampdb, ampdbfs, dbamp

Credits

Example written by Kevin Conder.

dcblock

dcblock — A DC blocking filter.

Description

Implements the DC blocking filter

$$Y[i] = X[i] - X[i-1] + (igain * Y[i-1])$$

Based on work by Perry Cook.

Syntax

```
ares dcblock ain [, igain]
```

Initialization

igain -- the gain of the filter, which defaults to 0.99

Performance

ain -- audio signal input



Note

The new *dcblock2* opcode is an improved method of removing DC from an audio signal.

Examples

Here is an example of the dcblock opcode. It uses the file *dcblock.csd* [examples/dcblock.csd], and *beats.wav* [examples/beats.wav].

Example 112. Example of the dcblock opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o dcblock.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

```
; Instrument #1 -- normal audio signal.
instr 1
  asig soundin "beats.wav"
  out asig
endin

; Instrument #2 -- dcblock-ed audio signal.
instr 2
  asig soundin "beats.wav"

  igain = 0.75
  al dcblock asig, igain

  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 2 seconds.
i 1 0 2
; Play Instrument #2 for 2 seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See also

dcblock2

Credits

Author: John ffitch
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.49

February 2003: Thanks to a note from Anders Andersson, corrected the formula.

dcblock2

dcblock2 — A DC blocking filter.

Description

Implements a DC blocking filter with improved DC attenuation.

Syntax

```
ares dcblock2 ain [, iorder] [, iskip]
```

Initialization

iorder -- filter order, minimum 4th order, defaults to 128.

iskip -- set to 1 to skip initialization (defaults to 0).

Performance

ares -- filtered audio signal

ain -- input audio signal



Note

Using a value for *iorder* less than *ksmps* will not reduce DC offset efficiently.

See also

dcblock

Credits

By: Victor Lazzarini

New in Csound version 5.09

dconv

dconv — A direct convolution opcode.

Description

A direct convolution opcode.

Syntax

```
ares dconv asig, isize, ifn
```

Initialization

isize -- the size of the convolution buffer to use. if the buffer size is smaller than the size of ifn, then only the first isize values will be used from the table.

ifn -- table number of a stored function containing the impulse response for convolution.

Performance

Rather than the analysis/resynthesis method of the convolve opcode, *dconv* uses direct convolution to create the result. For small tables it can do this quite efficiently, however larger table require much more time to run. *dconv* does (isize * ksmpls) multiplies on every k-cycle. Therefore, reverb and delay effects are best done with other opcodes (unless the times are short).

dconv was designed to be used with time varying tables to facilitate new realtime filtering capabilities.

Examples

Here is an example of the dconv opcode. It uses the file *dconv.csd* [examples/dconv.csd].

Example 113. Example of the dconv opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o dconv.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

#define RANDI(A) #kout randi 1, kfq, $A*.001+iseed, 1
tablew kout, $A, itable#

instr 1
itable init 1
iseed init .6
isize init ftlen(itable)
kfq line 1, p3, 10
```

```
$RANDI(0)
$RANDI(1)
$RANDI(2)
$RANDI(3)
$RANDI(4)
$RANDI(5)
$RANDI(6)
$RANDI(7)
$RANDI(8)
$RANDI(9)
$RANDI(10)
$RANDI(11)
$RANDI(12)
$RANDI(13)
$RANDI(14)
$RANDI(15)

asig      rand      10000, .5, 1
asig      butlp     asig, 5000
asig      dconv     asig, isize, itable

          out       asig *.5
endin

</CsInstruments>
<CsScore>

f1 0 16 10 1
i1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

See also

pconvolve, *convolve.*, *convolve*

Credits

Author: William “Pete” Moss
2001

New in version 4.12

delay

delay — Delays an input signal by some time interval.

Description

A signal can be read from or written into a delay path, or it can be automatically delayed by some time interval.

Syntax

```
ares delay asig, idlt [, iskip]
```

Initialization

idlt -- requested delay time in seconds. This can be as large as available memory will permit. The space required for *n* seconds of delay is $4n * sr$ bytes. It is allocated at the time the instrument is first initialized, and returned to the pool at the end of a score section.

iskip (optional, default=0) -- initial disposition of delay-loop data space (see *reson*). The default value is 0.

Performance

asig -- audio signal

delay is a composite of *delayr* and *delayw*, both reading from and writing into its own storage area. It can thus accomplish signal time-shift, although modified feedback is not possible. There is no minimum delay period.

Examples

Here is an example of the delay opcode. It uses the file *delay.csd* [examples/delay.csd].

Example 114. Example of the delay opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o delay.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Instrument #1 -- Delayed beeps.
instr 1
; Make a basic sound.
a beep vco 20000, 440, 1
```

```
; Delay the beep by .1 seconds.
idlt = 0.1
adel delay abeep, idlt

; Send the beep to the left speaker and
; the delayed beep to the right speaker.
outs abeep, adel
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Keep the score running for 2 seconds.
f 0 2

; Play Instrument #1.
i 1 0.0 0.2
i 1 0.5 0.2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

delay1, delayr, delayw

Credits

Example written by Kevin Conder.

delay1

delay1 — Delays an input signal by one sample.

Description

Delays an input signal by one sample.

Syntax

```
ares delay1 asig [, iskip]
```

Initialization

iskip (optional, default=0) -- initial disposition of delay-loop data space (see *reson*). The default value is 0.

Performance

delay1 is a special form of delay that serves to delay the audio signal *asig* by just one sample. It is thus functionally equivalent to the *delay* opcode but is more efficient in both time and space. This unit is particularly useful in the fabrication of generalized non-recursive filters.

See Also

delay, *delayr*, *delayw*

delayk

delayk — Delays an input signal by some time interval.

Description

k-rate delay opcodes

Syntax

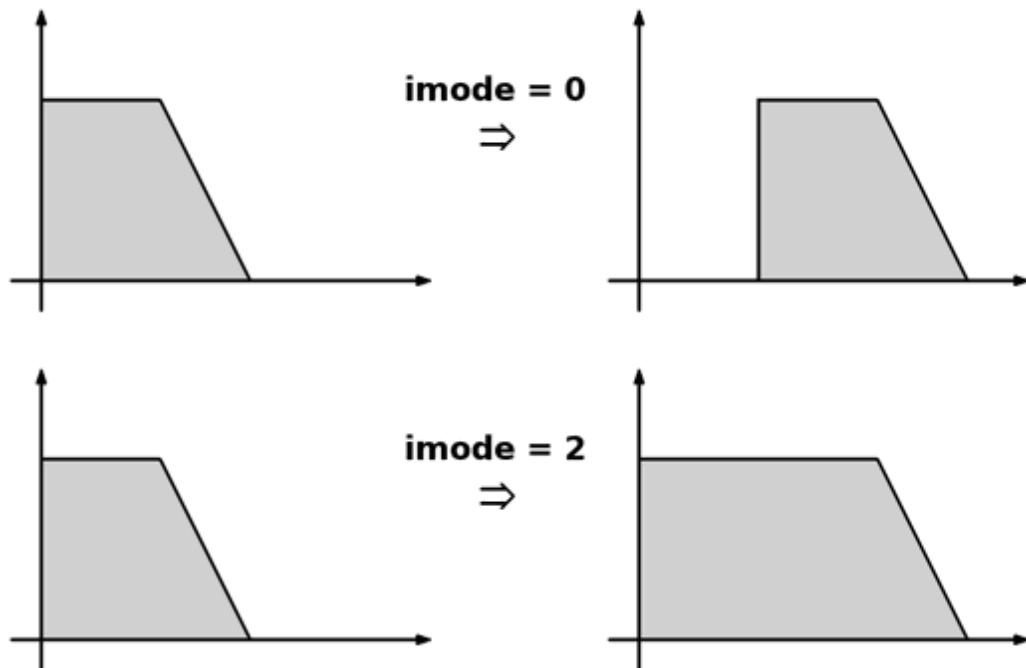
```
kr delayk    ksig, idel[, imode]
```

```
kr vdel_k    ksig, kdel, imdel[, imode]
```

Initialization

idel -- delay time (in seconds) for delayk. It is rounded to the nearest integer multiple of a k-cycle (i.e. $1/kr$).

imode -- sum of 1 for skipping initialization (e.g. in tied notes) and 2 for holding the first input value during the initial delay, instead of outputting zero. This is mainly of use when delaying envelopes that do not start at zero.



imdel -- maximum delay time for *vdel_k*, in seconds.

Performance

kr -- the output signal. Note: neither of the opcodes interpolate the output.

ksig -- the input signal.

kdel -- delay time (in seconds) for *vdel_k*. It is rounded to the nearest integer multiple of a k-cycle (i.e. $1/kr$).

Credits

Istvan Varga.

delayr

delayr — Reads from an automatically established digital delay line.

Description

Reads from an automatically established digital delay line.

Syntax

```
ares delayr idlt [, iskip]
```

Initialization

idlt -- requested delay time in seconds. This can be as large as available memory will permit. The space required for *n* seconds of delay is $4n * sr$ bytes. It is allocated at the time the instrument is first initialized, and returned to the pool at the end of a score section.

iskip (optional, default=0) -- initial disposition of delay-loop data space (see *reson*). The default value is 0.

Performance

delayr reads from an automatically established digital delay line, in which the signal retrieved has been resident for *idlt* seconds. This unit must be paired with and precede an accompanying *delayw* unit. Any other Csound statements can intervene.

Examples

See the example for *delayw*.

See Also

delay, *delayl*, *delayw*

delayw

delayw — Writes the audio signal to a digital delay line.

Description

Writes the audio signal to a digital delay line.

Syntax

```
delayw asig
```

Performance

delayw writes *asig* into the delay area established by the preceding *delayr* unit. Viewed as a pair, these two units permit the formation of modified feedback loops, etc. However, there is a lower bound on the value of *idlt*, which must be at least 1 control period (or $1/kr$).

Examples

Here is an example of the *delayw* opcode. It uses the file *delayw.csd* [examples/delayw.csd].

Example 115. Example of the *delayw* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o delayw.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Instrument #1 -- Delayed beeps.
instr 1
; Make a basic sound.
a beep vco 20000, 440, 1

; Set up a delay line.
idlt = 0.1
adel delayr idlt

; Write the beep to the delay line.
delayw a beep

; Send the beep to the left speaker and
; the delayed beep to the right speaker.
outs a beep, adel
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1
```

```
; Keep the score running for 2 seconds.  
f 0 2  
  
; Play Instrument #1.  
i 1 0.0 0.2  
i 1 0.5 0.2  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

delay, delay1, delayr

Credits

Example written by Kevin Conder.

deltap

deltap — Taps a delay line at variable offset times.

Description

Tap a delay line at variable offset times.

Syntax

```
ares deltap kdl1
```

Performance

kdl1 -- specifies the tapped delay time in seconds. Each can range from 1 control period to the full delay time of the read/write pair; however, since there is no internal check for adherence to this range, the user is wholly responsible. Each argument can be a constant, a variable, or a time-varying signal.

deltap extracts sound by reading the stored samples directly.

This opcode can tap into a *delayr/delayw* pair, extracting delayed audio from the *idlt* seconds of stored sound. There can be any number of *deltap* and/or *deltapi* units between a read/write pair. Each receives an audio tap with no change of original amplitude.

This opcode can provide multiple delay taps for arbitrary delay path and feedback networks. They can deliver either constant-time or time-varying taps, and are useful for building chorus effects, harmonizers, and Doppler shifts. Constant-time delay taps (and some slowly changing ones) do not need interpolated readout; they are well served by *deltap*. Medium-paced or fast varying dlt's, however, will need the extra services of *deltapi*.

delayr/delayw pairs may be interleaved. To associate a delay tap unit with a specific *delayr* unit, it not only has to be located between that *delayr* and the appropriate *delayw* unit, but must also precede any following *delayr* units. See Example 2. (This feature added in Csound version 3.57 by Jens Groh and John ffitch).

N.B. k-rate delay times are not internally interpolated, but rather lay down stepped time-shifts of audio samples; this will be found quite adequate for slowly changing tap times. For medium to fast-paced changes, however, one should provide a higher resolution audio-rate timeshift as input.

Examples

Example 116. deltap example #1

```
asource  buzz      1, 440, 20, 1
atime    linseg     1, p3/2,.01, p3/2,1 ; trace a distance in secs
ampfac   =          1/atime/atime       ; and calc an amp factor
adump    delayr     1                   ; set maximum distance
amove    deltapi    atime               ; move sound source past
          delayw     asource            ; the listener
          out        amove * ampfac
```

Example 117. *deltap* example #2

```
ainput1 = .....
ainput2 = .....
kdlyt1  = .....
kdlyt2  = .....

;Read delayed signal, first delayr instance:
  adump   delayr  4.0
  adly1   deltap kdlyt1           ;associated with first delayr instance

;Read delayed signal, second delayr instance:
  adump   delayr  4.0
  adly2   deltap kdlyt2           ; associated with second delayr instance

;Do some cross-coupled manipulation:
  afdbk1  =      0.7 * adly1 + 0.7 * adly2 + ainput1
  afdbk2  =     -0.7 * adly1 + 0.7 * adly2 + ainput2

;Feed back signal, associated with first delayr instance:
      delayw  afdbk1

;Feed back signal, associated with second delayr instance:
      delayw  afdbk2
      outs    adly1, adly2
```

See Also

deltap3, deltapi, deltapn

deltap3

deltap — Taps a delay line at variable offset times, uses cubic interpolation.

Description

Taps a delay line at variable offset times, uses cubic interpolation.

Syntax

```
ares deltap3 xdlrt
```

Performance

xdlrt -- specifies the tapped delay time in seconds. Each can range from 1 control period to the full delay time of the read/write pair; however, since there is no internal check for adherence to this range, the user is wholly responsible. Each argument can be a constant, a variable, or a time-varying signal; the *xdlrt* argument in *deltap3* implies that an audio-varying delay is permitted there.

deltap3 is experimental, and uses cubic interpolation. (New in Csound version 3.50.)

This opcode can tap into a *delayr/delayw* pair, extracting delayed audio from the *idlt* seconds of stored sound. There can be any number of *deltap* and/or *deltapi* units between a read/write pair. Each receives an audio tap with no change of original amplitude.

This opcode can provide multiple delay taps for arbitrary delay path and feedback networks. They can deliver either constant-time or time-varying taps, and are useful for building chorus effects, harmonizers, and Doppler shifts. Constant-time delay taps (and some slowly changing ones) do not need interpolated readout; they are well served by *deltap*. Medium-paced or fast varying dlt's, however, will need the extra services of *deltapi*.

delayr/delayw pairs may be interleaved. To associate a delay tap unit with a specific *delayr* unit, it not only has to be located between that *delayr* and the appropriate *delayw* unit, but must also precede any following *delayr* units. See Example 2. (This feature added in Csound version 3.57 by Jens Groh and John ffitth).

N.B. k-rate delay times are not internally interpolated, but rather lay down stepped time-shifts of audio samples; this will be found quite adequate for slowly changing tap times. For medium to fast-paced changes, however, one should provide a higher resolution audio-rate timeshift as input.

Examples

Example 118. deltap example #1

```
asource  buzz      1, 440, 20, 1
atime    linseg     1, p3/2,.01, p3/2,1 ; trace a distance in secs
ampfac   =          1/atime/atime       ; and calc an amp factor
adump    delayr     1                   ; set maximum distance
amove    deltapi    atime                ; move sound source past
          delayw     asource             ; the listener
          out        amove * ampfac
```

Example 119. *deltap* example #2

```
ainput1 = .....
ainput2 = .....
kdlyt1  = .....
kdlyt2  = .....

;Read delayed signal, first delayr instance:
  adump  delayr  4.0
  adly1  deltap kdlyt1      ;associated with first delayr instance

;Read delayed signal, second delayr instance:
  adump  delayr  4.0
  adly2  deltap kdlyt2      ; associated with second delayr instance

;Do some cross-coupled manipulation:
  afdbk1 =      0.7 * adly1 + 0.7 * adly2 + ainput1
  afdbk2 =     -0.7 * adly1 + 0.7 * adly2 + ainput2

;Feed back signal, associated with first delayr instance:
  delayw  afdbk1

;Feed back signal, associated with second delayr instance:
  delayw  afdbk2
  outs    adly1, adly2
```

See Also

deltap, *deltapi*, *deltapn*

deltapi

deltapi — Taps a delay line at variable offset times, uses interpolation.

Description

Taps a delay line at variable offset times, uses interpolation.

Syntax

```
ares deltapi xdl t
```

Performance

xdl t -- specifies the tapped delay time in seconds. Each can range from 1 control period to the full delay time of the read/write pair; however, since there is no internal check for adherence to this range, the user is wholly responsible. Each argument can be a constant, a variable, or a time-varying signal; the *xdl t* argument in *deltapi* implies that an audio-varying delay is permitted there.

deltapi extracts sound by interpolated readout. By interpolating between adjacent stored samples *deltapi* represents a particular delay time with more accuracy, but it will take about twice as long to run.

This opcode can tap into a *delayr/delayw* pair, extracting delayed audio from the *idlt* seconds of stored sound. There can be any number of *deltap* and/or *deltapi* units between a read/write pair. Each receives an audio tap with no change of original amplitude.

This opcode can provide multiple delay taps for arbitrary delay path and feedback networks. They can deliver either constant-time or time-varying taps, and are useful for building chorus effects, harmonizers, and Doppler shifts. Constant-time delay taps (and some slowly changing ones) do not need interpolated readout; they are well served by *deltap*. Medium-paced or fast varying dlt's, however, will need the extra services of *deltapi*.

delayr/delayw pairs may be interleaved. To associate a delay tap unit with a specific *delayr* unit, it not only has to be located between that *delayr* and the appropriate *delayw* unit, but must also precede any following *delayr* units. See Example 2. (This feature added in Csound version 3.57 by Jens Groh and John ffitch).

N.B. k-rate delay times are not internally interpolated, but rather lay down stepped time-shifts of audio samples; this will be found quite adequate for slowly changing tap times. For medium to fast-paced changes, however, one should provide a higher resolution audio-rate timeshift as input.

Examples

Example 120. deltap example #1

```
asource  buzz      1, 440, 20, 1
atime    linseg     1, p3/2,.01, p3/2,1 ; trace a distance in secs
ampfac   =          1/atime/atime       ; and calc an amp factor
adump    delayr     1                   ; set maximum distance
amove    deltapi    atime                ; move sound source past
          delayw     asource              ; the listener
          out        amove * ampfac
```

Example 121. *deltap* example #2

```
ainput1 = .....
ainput2 = .....
kdlyt1  = .....
kdlyt2  = .....

;Read delayed signal, first delayr instance:
adump   delayr 4.0
adly1   deltap kdlyt1           ;associated with first delayr instance

;Read delayed signal, second delayr instance:
adump   delayr 4.0
adly2   deltap kdlyt2           ; associated with second delayr instance

;Do some cross-coupled manipulation:
afdbk1  =      0.7 * adly1 + 0.7 * adly2 + ainput1
afdbk2  =     -0.7 * adly1 + 0.7 * adly2 + ainput2

;Feed back signal, associated with first delayr instance:
delayw  afdbk1

;Feed back signal, associated with second delayr instance:
delayw  afdbk2
outs    adly1, adly2
```

See Also

deltap, *deltap3*, *deltapn*

deltapn

deltapn — Taps a delay line at variable offset times.

Description

Tap a delay line at variable offset times.

Syntax

```
ares deltapn xnumsamps
```

Performance

xnumsamps -- specifies the tapped delay time in number of samples. Each can range from 1 control period to the full delay time of the read/write pair; however, since there is no internal check for adherence to this range, the user is wholly responsible. Each argument can be a constant, a variable, or a time-varying signal.

deltapn is identical to *deltapi*, except delay time is specified in number of samples, instead of seconds (Hans Mikelson).

This opcode can tap into a *delayr/delayw* pair, extracting delayed audio from the *idlt* seconds of stored sound. There can be any number of *deltap* and/or *deltapi* units between a read/write pair. Each receives an audio tap with no change of original amplitude.

This opcode can provide multiple delay taps for arbitrary delay path and feedback networks. They can deliver either constant-time or time-varying taps, and are useful for building chorus effects, harmonizers, and Doppler shifts. Constant-time delay taps (and some slowly changing ones) do not need interpolated readout; they are well served by *deltap*. Medium-paced or fast varying dlt's, however, will need the extra services of *deltapi*.

delayr/delayw pairs may be interleaved. To associate a delay tap unit with a specific *delayr* unit, it not only has to be located between that *delayr* and the appropriate *delayw* unit, but must also precede any following *delayr* units. See Example 2. (This feature added in Csound version 3.57 by Jens Groh and John ffitch).

N.B. k-rate delay times are not internally interpolated, but rather lay down stepped time-shifts of audio samples; this will be found quite adequate for slowly changing tap times. For medium to fast-paced changes, however, one should provide a higher resolution audio-rate timeshift as input.

Examples

Example 122. deltap example #1

```
asource  buzz      1, 440, 20, 1
atime    linseg     1, p3/2,.01, p3/2,1 ; trace a distance in secs
ampfac   =          1/atime/atime       ; and calc an amp factor
adump    delayr     1                    ; set maximum distance
amove    deltapi    atime                ; move sound source past
          delayw     asource              ; the listener
          out        amove * ampfac
```

Example 123. *deltap* example #2

```
ainput1 = .....
ainput2 = .....
kdlyt1  = .....
kdlyt2  = .....

;Read delayed signal, first delayr instance:
  adump  delayr  4.0
  adly1  deltap kdlyt1      ;associated with first delayr instance

;Read delayed signal, second delayr instance:
  adump  delayr  4.0
  adly2  deltap kdlyt2      ; associated with second delayr instance

;Do some cross-coupled manipulation:
  afdbk1 =      0.7 * adly1 + 0.7 * adly2 + ainput1
  afdbk2 =     -0.7 * adly1 + 0.7 * adly2 + ainput2

;Feed back signal, associated with first delayr instance:
      delayw  afdbk1

;Feed back signal, associated with second delayr instance:
      delayw  afdbk2
      outs    adly1, adly2
```

See Also

deltap, *deltap3*, *deltapi*

deltapx

deltapx — Read to or write from a delay line with interpolation.

Description

deltapx is similar to *deltapi* or *deltap3*. However, it allows higher quality interpolation. This opcode can read from and write to a delayr/delayw delay line with interpolation.

Syntax

```
aout deltapx adel, iwsiz
```

Initialization

iwsiz -- interpolation window size in samples. Allowed values are integer multiplies of 4 in the range 4 to 1024. *iwsiz* = 4 uses cubic interpolation. Increasing *iwsiz* improves sound quality at the expense of CPU usage, and minimum delay time.

Performance

aout -- Output signal

adel -- Delay time in seconds.

```
a1      delayr idlr
          deltapxw a2, adl1, iws1
a3      deltapx adl2, iws2
          deltapxw a4, adl3, iws3
          delayw a5
```

Minimum and maximum delay times:

$idlr \geq 1/kr$	Delay line length
$adl1 \geq (iws1/2)/sr$	Write before read
$adl1 \leq idlr - (1 + iws1/2)/sr$	(allows shorter delays)
$adl2 \geq 1/kr + (iws2/2)/sr$	Read time
$adl2 \leq idlr - (1 + iws2/2)/sr$	
$adl2 \geq adl1 + (iws1 + iws2) / (2*sr)$	
$adl2 \geq 1/kr + adl3 + (iws2 + iws3) / (2*sr)$	
$adl3 \geq (iws3/2)/sr$	Write after read
$adl3 \leq idlr - (1 + iws3/2)/sr$	(allows feedback)



Note

Window sizes for opcodes other than *deltapx* are: *deltap*, *deltapn*: 1, *deltapi*: 2 (linear), *deltap3*: 4 (cubic)

Examples

```
a1      phasor 300.0
a1      = a1 - 0.5
a_      delayr 1.0
adel    phasor 4.0
adel    = sin (2.0 * 3.14159265 * adel) * 0.01 + 0.2
        deltapxw a1, adel, 32
adel    phasor 2.0
adel    = sin (2.0 * 3.14159265 * adel) * 0.01 + 0.2
        deltapxw a1, adel, 32
adel    = 0.3
a2      deltapx adel, 32
a1      = 0
        delayw a1

        out a2 * 20000.0
```

See Also

deltapxw

Credits

Author: Istvan Varga
August 2001

New in version 4.13

deltapxw

deltapxw — Mixes the input signal to a delay line.

Description

deltapxw mixes the input signal to a delay line. This opcode can be mixed with reading units (*deltap*, *deltapn*, *deltapi*, *deltap3*, and *deltapx*) in any order; the actual delay time is the difference of the read and write time. This opcode can read from and write to a *delayr/delayw* delay line with interpolation.

Syntax

```
deltapxw ain, adel, iwsiz
```

Initialization

iwsiz -- interpolation window size in samples. Allowed values are integer multiplies of 4 in the range 4 to 1024. *iwsiz* = 4 uses cubic interpolation. Increasing *iwsiz* improves sound quality at the expense of CPU usage, and minimum delay time.

Performance

ain -- Input signal

adel -- Delay time in seconds.

```
a1      delayr idlr
deltapxw a2, adl1, iws1
a3      deltapx adl2, iws2
deltapxw a4, adl3, iws3
delayw a5
```

Minimum and maximum delay times:

$idlr \geq 1/kr$	Delay line length
$adl1 \geq (iws1/2)/sr$	Write before read
$adl1 \leq idlr - (1 + iws1/2)/sr$	(allows shorter delays)
$adl2 \geq 1/kr + (iws2/2)/sr$	Read time
$adl2 \leq idlr - (1 + iws2/2)/sr$	
$adl2 \geq adl1 + (iws1 + iws2) / (2*sr)$	
$adl2 \geq 1/kr + adl3 + (iws2 + iws3) / (2*sr)$	
$adl3 \geq (iws3/2)/sr$	Write after read
$adl3 \leq idlr - (1 + iws3/2)/sr$	(allows feedback)



Note

Window sizes for opcodes other than *deltapx* are: *deltap*, *deltapn*: 1, *deltapi*: 2 (linear), *deltap3*: 4 (cubic)

Examples

```
a1      phasor 300.0
a1      = a1 - 0.5
a_      delayr 1.0
adel    phasor 4.0
adel    = sin (2.0 * 3.14159265 * adel) * 0.01 + 0.2
        deltapxw a1, adel, 32
adel    phasor 2.0
adel    = sin (2.0 * 3.14159265 * adel) * 0.01 + 0.2
        deltapxw a1, adel, 32
adel    = 0.3
a2      deltapx adel, 32
a1      = 0
        delayw a1

out a2 * 20000.0
```

See Also

deltapx

Credits

Author: Istvan Varga
August 2001

New in version 4.13

denorm

denorm — Mixes low level noise to a list of a-rate signals

Description

Mixes low level ($\sim 1e-20$ for floats, and $\sim 1e-56$ for doubles) noise to a list of a-rate signals. Can be used before IIR filters and reverbs to avoid denormalized numbers which may otherwise result in significantly increased CPU usage.

Syntax

```
denorm a1[, a2[, a3[, ... ]]]
```

Performance

a1[, a2[, a3[, ...]]] -- signals to mix noise with

Some processor architectures (particularly Pentium IVs) are very slow at processing extremely small numbers. These small numbers can appear as a result of some decaying feedback process like reverb and IIR filters. Low level noise can be added so that very small numbers are never reached, and they are 'absorbed' by this 'noise floor'.

If CPU usage goes to 100% at the end of reverb tails, or you get audio glitches in processes that shouldn't use too much CPU, using *denorm* before the culprit opcode or process might solve the problem.

Credits

Author: Istvan Varga
2005

diff

diff — Modify a signal by differentiation.

Description

Modify a signal by differentiation.

Syntax

```
ares diff asig [, iskip]
```

```
kres diff ksig [, iskip]
```

Initialization

iskip (optional) -- initial disposition of internal save space (see *reson*). The default value is 0.

Performance

integ and *diff* perform integration and differentiation on an input control signal or audio signal. Each is the converse of the other, and applying both will reconstruct the original signal. Since these units are special cases of low-pass and high-pass filters, they produce a scaled (and phase shifted) output that is frequency-dependent. Thus *diff* of a sine produces a cosine, with amplitude $2 * \sin(\pi * Hz / sr)$ that of the original (for each component partial); *integ* will inversely affect the magnitudes of its component inputs. With this understanding, these units can provide useful signal modification.

Examples

Here is an example of the diff opcode. It uses the file *diff.csd* [examples/diff.csd].

Example 124. Example of the diff opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o diff.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 -- a normal instrument.
instr 1
; Generate a band-limited pulse train.
asrc buzz 20000, 440, 20, 1

out asrc
endin

; Instrument #2 -- a differentiated instrument.
```



```
instr 2
; Generate a band-limited pulse train.
asrc buzz 20000, 440, 20, 1

; Emphasize the highs.
al diff asrc

out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 1 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

downsamp, integ, interp, samphold, upsamp

Credits

Example written by Kevin Conder.

diskgrain

diskgrain — Synchronous granular synthesis, using a soundfile as source.

Description

diskgrain implements synchronous granular synthesis. The source sound for the grains is obtained by reading a soundfile containing the samples of the source waveform.

Syntax

```
asig diskgrain Sfname, kamp, kfreq, kpitch, kgrsize, kprate, \  
      ifun, iolaps[, ioffset, imaxgrsize]
```

Initialization

Sfilename -- source soundfile.

ifun -- grain envelope function table.

iolaps -- maximum number of overlaps, $\max(kfreq) \times \max(kgrsize)$. Estimating a large value should not affect performance, but exceeding this value will probably have disastrous consequences.

ioffset -- start offset in secs from beginning of file (default: 0).

imaxgrsize -- max grain size in secs (default 1.0).

Performance

kamp -- amplitude scaling

kfreq -- frequency of grain generation, or density, in grains/sec.

kpitch -- grain pitch scaling (1=normal pitch, < 1 lower, > 1 higher; negative, backwards)

kgrsize -- grain size in secs.

kprate -- readout pointer rate, in grains. The value of 1 will advance the reading pointer 1 grain ahead in the source table. Larger values will time-compress and smaller values will time-expand the source signal. Negative values will cause the pointer to run backwards and zero will freeze it.

The grain generator has full control of frequency (grains/sec), overall amplitude, grain pitch (a sampling increment) and grain size (in secs), both as fixed or time-varying (signal) parameters. An extra parameter is the grain pointer speed (or rate), which controls which position the generator will start reading samples in the file for each successive grain. It is measured in fractions of grain size, so a value of 1 (the default) will make each successive grain read from where the previous grain should finish. A value of 0.5 will make the next grain start at the midway position from the previous grain start and finish, etc.. A value of 0 will make the generator read always from a fixed position (wherever the pointer was last at). A negative value will decrement pointer positions. This control gives extra flexibility for creating timescale modifications in the resynthesis.

Diskgrain will generate any number of parallel grain streams (which will depend on grain density/frequency), up to the *olaps* value (default 100). The number of streams (overlapped grains) is determined by $grainsize \times grain_freq$. More grain overlaps will demand more calculations and the synthesis might not run in realtime (depending on processor power).

Diskgrain can simulate FOF-like formant synthesis, provided that a suitable shape is used as grain envelope and a sinewave as the grain wave. For this use, grain sizes of around 0.04 secs can be used.

The formant centre frequency is determined by the grain pitch. Since this is a sampling increment, in order to use a frequency in Hz, that value has to be scaled by `tablesiz/sr`. Grain frequency will determine the fundamental.

This opcode is a variation on the *syncgrain* opcode.

Examples

Here is an example of the *diskgrain* opcode. It uses the file *diskgrain.csd* [examples/diskgrain.csd].

Example 125. Example of the *diskgrain* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr = 48000
ksmps = 128

instr 1
iolaps = 2
igrsize = 0.04
ifreq = iolaps/igrsize
ips = 1/iolaps

istr = p4 /* timescale */
ipitch = p5 /* pitchscale */

a1 diskgrain "mary.wav", 32000, ifreq, ipitch, igrsize, ips*istr, 1, iolaps

out a1
endin

</CsInstruments>
<CsScore>
f 1 0 8192 20 1 1 ;Hamming function

;          timescale  pitchscale
i 1 0 5 1 1
i 1 + 5 2 1
i 1 + 5 1 0.75
i 1 + 5 1.5 1.5
i 1 + 5 0.5 1.5

e
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Victor Lazzarini;
May 2007
New in Csound 5.06

diskin

diskin — Reads audio data from an external device or stream and can alter its pitch.

Description

Reads audio data from an external device or stream and can alter its pitch.

Syntax

```
ar1 [, ar2 [, ar3 [, ... ar24]]] diskin ifilcod, kpitch [, iskptim] \  
[, iwraparound] [, iformat] [, iskipinit]
```

Initialization

ifilcod -- integer or character-string denoting the source soundfile name. An integer denotes the file soundin.filcod ; a character-string (in double quotes, spaces permitted) gives the filename itself, optionally a full pathname. If not a full path, the named file is sought first in the current directory, then in that given by the environment variable SSDIR (if defined) then by SFDIR. See also *GEN01*.

iskptim (optional) -- time in seconds of input sound to be skipped. The default value is 0.

iformat (optional) -- specifies the audio data file format:

- 1 = 8-bit signed char (high-order 8 bits of a 16-bit integer)
- 2 = 8-bit A-law bytes
- 3 = 8-bit U-law bytes
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = 8-bit unsigned int (not available in Csound versions older than 5.00)
- 8 = 24-bit int (not available in Csound versions older than 5.00)
- 9 = 64-bit doubles (not available in Csound versions older than 5.00)

iwraparound -- 1 = on, 0 = off (wraps around to end of file either direction)

iskipinit switches off all initialisation if non zero (default =0). This was introduced in 4_23f13 and csound5.

If *iformat* = 0 it is taken from the soundfile header, and if no header from the Csound *-o* command-line flag. The default value is 0.

Performance

kpitch -- can be any real number. a negative number signifies backwards playback. The given number is a pitch ratio, where:

- 1 = normal pitch

- 2 = 1 octave higher
- 3 = 12th higher, etc.
- .5 = 1 octave lower
- .25 = 2 octaves lower, etc.
- -1 = normal pitch backwards
- -2 = 1 octave higher backwards, etc.

diskin is identical to *soundin* except that it can alter the pitch of the sound that is being read.



Note to Windows users

Windows users typically use back-slashes, “\”, when specifying the paths of their files. As an example, a Windows user might use the path “c:\music\samples\loop001.wav”. This is problematic because back-slashes are normally used to specify special characters.

To correctly specify this path in Csound, one may alternately:

- Use forward slashes: c:/music/samples/loop001.wav
- Use back-slash special characters, “\\”: c:\\music\\samples\\loop001.wav

Examples

Here is an example of the *diskin* opcode. It uses the file *diskin.csd* [examples/diskin.csd], *beats.wav* [examples/beats.wav].

Example 126. Example of the *diskin* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o diskin.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1 - play an audio file.
instr 1
; Play the audio file backwards.
asig diskin "beats.wav", -1
out asig
endin

</CsInstruments>
<CsScore>
```

```
; Play Instrument #1, the audio file, for three seconds.  
i 1 0 3  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

in, *inh*, *ino*, *inq*, *ins*, *soundin* and *diskin2*

Credits

Authors: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

New in versin 3.46

Example written by Kevin Conder.

Warning to Windows users added by Kevin Conder, April 2002

diskin2

diskin2 — Reads audio data from a file, and can alter its pitch using one of several available interpolation types, as well as convert the sample rate to match the orchestra sr setting.

Description

Reads audio data from a file, and can alter its pitch using one of several available interpolation types, as well as convert the sample rate to match the orchestra sr setting. diskin2 can also read multichannel files with any number of channels in the range 1 to 24. diskin2 allows more control and higher sound quality than diskin, but there is also the disadvantage of higher CPU usage.

Syntax

```
a1[, a2[, ... a24]] diskin2 ifilcod, kpitch[, iskiptim \
[, iwrap[, iformat [, iwsizel[, ibufsize[, iskipinit]]]]]]
```

Initialization

ifilcod -- integer or character-string denoting the source soundfile name. An integer denotes the file soundin.ifilcod; a character-string (in double quotes, spaces permitted) gives the filename itself, optionally a full pathname. If not a full path, the named file is sought first in the current directory, then in those given by the environment variable SSDIR (if defined) then by SFDIR. See also GEN01. Note: files longer than $2^{31}-1$ sample frames may not be played correctly on 32 bit platforms; this means a maximum length about 3 hours with a sample rate of 192000 Hz.

iskiptim (optional, defaults to zero) -- time in seconds of input sound to be skipped, assuming *kpitch*=1. Can be negative, to add $-\text{iskiptim}/\text{kpitch}$ seconds of delay instead of skipping sound.

iwrap (optional, defaults to zero) -- if set to any non-zero value, read locations that are negative or are beyond the end of the file are wrapped to the duration of the sound file instead of assuming zero samples. Useful for playing a file in a loop.



Note

If *iwrap* is enabled, the file length should not be shorter than the interpolation window size (see below), otherwise there may be clicks in the sound output.

iformat (optional, defaults to zero) -- sample format, for raw (headerless) files only. This parameter is ignored if the file has a header. Allowed values are:

- 0: 16-bit short integers
- 1: 8-bit signed char (high-order 8 bits of a 16-bit integer)
- 2: 8-bit A-law bytes
- 3: 8-bit U-law bytes
- 4: 16-bit short integers
- 5: 32-bit long integers
- 6: 32-bit floats
- 7: 8-bit unsigned int

- 8: 24-bit int
- 9: 64-bit doubles

*iwsiz*e (optional, defaults to zero) -- interpolation window size, in samples. Can be one of the following:

- 1: round to nearest sample (no interpolation, for *kpitch*=1)
- 2: linear interpolation
- 4: cubic interpolation
- >= 8: *iwsiz*e point sinc interpolation with anti-aliasing (slow)

Zero or negative values select the default, which is cubic interpolation.



Note

If interpolation is used, *kpitch* is automatically scaled by the ratio of the sample rate of the sound file and the orchestra, so that the file will always be played at the original pitch if *kpitch* is 1. However, the sample rate conversion is disabled if *iwsiz*e is 1.

*ibufsiz*e (optional, defaults to 0) -- buffer size in mono samples (not sample frames). This is only the suggested value, the actual setting will be rounded so that the number of sample frames is an integer power of two and is in the range 128 (or *iwsiz*e if greater than 128) to 1048576. The default, which is 4096, and is enabled by zero or negative values, should be suitable for most uses, but for non-realtime mixing of many large sound files, a high buffer setting is recommended to improve the efficiency of disk reads. For real time audio output, reading the files from a fast RAM file system (on platforms where this option is available) with a small buffer size may be preferred.

iskipinit (optional, defaults to 0) -- skip initialization if set to any non-zero value.

Performance

a1 ... a24 -- output signals, in the range -0dbfs to 0dbfs. Any samples before the beginning (i.e. negative location) and after the end of the file are assumed to be zero, unless *iwrap* is non-zero. The number of output arguments must be the same as the number of sound file channels - which can be determined with the *filenchnls* opcode, otherwise an init error will occur.



Note

It is more efficient to read a single file with many channels, than many files with only a single channel, especially with high *iwsiz*e settings.

kpitch -- transpose the pitch of input sound by this factor (e.g. 0.5 means one octave lower, 2 is one octave higher, and 1 is the original pitch). Fractional and negative values are allowed (the latter results in playing the file backwards, however, in this case the skip time parameter should be set to some positive value, e.g. the length of the file, or *iwrap* should be non-zero, otherwise nothing would be played). If interpolation is enabled, and the sample rate of the file differs from the orchestra sample rate, the transpose ratio is automatically adjusted to make sure that *kpitch*=1 plays at the original pitch. Using a high *iwsiz*e setting (40 or more) can significantly improve sound quality when transposing up, although at the expense of high CPU usage.

Example

```
<CsoundSynthesizer>  
<CsOptions>
```



```
; set this to a directory where beats.wav can be found
--env:SSDIR+=/Csound/Documentation/manual/examples
</CsOptions>
<CsInstruments>
sr      = 48000
ksmps   = 32
nchnls  = 2

instr 1

ktrans  linseg 1, 5, 2, 10, -2
a1      diskin2 "beats.wav", ktrans, 0, 1, 0, 32
outs    a1, a1
endin

</CsInstruments>
<CsScore>

i 1 0 15
e

</CsScore>
</CsoundSynthesizer>
```

See Also

in, inh, ino, inq, ins, soundin and *diskin2*

Credits

Author: Istvan Varga
2005

New in version 5.00

dispfft

displayfft — Displays the Fourier Transform of an audio or control signal.

Description

These units will print orchestra init-values, or produce graphic display of orchestra control signals and audio signals. Uses X11 windows if enabled, else (or if -g flag is set) displays are approximated in ASCII characters.

Syntax

```
dispfft xsig, iprd, iwsiz [, iwtyp] [, idbout] [, iwtflg]
```

Initialization

iprd -- the period of display in seconds.

iwsiz -- size of the input window in samples. A window of *iwsiz* points will produce a Fourier transform of *iwsiz*/2 points, spread linearly in frequency from 0 to sr/2. *iwsiz* must be a power of 2, with a minimum of 16 and a maximum of 4096. The windows are permitted to overlap.

iwtyp (optional, default=0) -- window type. 0 = rectangular, 1 = Hanning. The default value is 0 (rectangular).

idbout (optional, default=0) -- units of output for the Fourier coefficients. 0 = magnitude, 1 = decibels. The default is 0 (magnitude).

iwtflg (optional, default=0) -- wait flag. If non-zero, each display is held until released by the user. The default value is 0 (no wait).

Performance

dispfft -- displays the Fourier Transform of an audio or control signal (*asig* or *ksig*) every *iprd* seconds using the Fast Fourier Transform method.

Examples

Here is an example of the dispfft opcode. It uses the file *dispfft.csd* [examples/dispfft.csd] and *beats.wav* [examples/beats.wav].

Example 127. Example of the dispfft opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac        -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o dispfft.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
```

```
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  asig soundin "beats.wav"
  dispfft asig, 1, 512
  out asig
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

display, print

Credits

Comments about the *inprds* parameter contributed by Rasmus Ekman.

Example written by Kevin Conder.

display

display — Displays the audio or control signals as an amplitude vs. time graph.

Description

These units will print orchestra init-values, or produce graphic display of orchestra control signals and audio signals. Uses X11 windows if enabled, else (or if -g flag is set) displays are approximated in ASCII characters.

Syntax

```
display xsig, iprd [, inprds] [, iwtflg]
```

Initialization

iprd -- the period of display in seconds.

inprds (optional, default=1) -- Number of display periods retained in each display graph. A value of 2 or more will provide a larger perspective of the signal motion. The default value is 1 (each graph completely new).

inprds (optional, default=1) -- a scaling factor for the displayed waveform, controlling how many iprd-sized frames of samples are drawn in the window (the default and minimum value is 1.0). Higher inprds values are slower to draw (more points to draw) but will show the waveform scrolling through the window, which is useful with low iprd values.

iwtflg (optional, default=0) -- wait flag. If non-zero, each display is held until released by the user. The default value is 0 (no wait).

Performance

display -- displays the audio or control signal *xsig* every *iprd* seconds, as an amplitude vs. time graph.

Examples

Here is an example of the display opcode. It uses the file *display.csd* [examples/display.csd].

Example 128. Example of the display opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o display.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
```

```
nchnls = 1

; Instrument #1.
instr 1
; Go from 1000 to 0 linearly, over the period defined by p3.
klin line 1000, p3, 0

; Create a new display each second, wait for the user.
display klin, 1, 1, 1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 5 seconds.
i 1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

See Also

dispfft, print

Credits

Comments about the *inprds* parameter contributed by Rasmus Ekman.

Example written by Kevin Conder.

distort

distort — Distort an audio signal via waveshaping and optional clipping.

Description

Syntax

```
ar distort asig, kdist, ifn[, ihp, istor]
```

Initialization

ifn -- table number of a waveshaping function with extended guard point. The function can be of any shape, but it should pass through 0 with positive slope at the table mid-point. The table size need not be large, since it is read with interpolation.

ihp -- (optional) half-power point (in cps) of an internal low-pass filter. The default value is 10.

istor -- (optional) initial disposition of internal data space (see *reson*). The default value is 0.

Performance

asig -- Audio signal to be processed

kdist -- Amount of distortion (usually between 0 and 1)

This unit distorts an incoming signal using a waveshaping function *ifn* and a distortion index *kdist*. The input signal is first compressed using a running rms, then passed through a waveshaping function which may modify its shape and spectrum. Finally it is rescaled to approximately its original power.

The amount of distortion depends on the nature of the shaping function and on the value of *kdist*, which generally ranges from 0 to 1. For low values of *kdist*, we should like the shaping function to pass the signal almost unchanged. This will be the case if, at the mid-point of the table, the shaping function is near-linear and is passing through 0 with positive slope. A line function from -1 to +1 will satisfy this requirement; so too will a sigmoid (sinusoid from 270 to 90 degrees). As *kdist* is increased, the compressed signal is expanded to encounter more and more of the shaping function, and if this becomes non-linear the signal is increasingly *bent* on read-through to cause distortion.

When *kdist* becomes large enough, the read-through process will eventually hit the outer limits of the table. The table is not read with wrap-around, but will 'stick' at the end-points as the incoming signal exceeds them; this introduces clipping, an additional form of signal distortion. The point at which clipping begins will depend on the complexity (rms-to-peak value) of the input signal. For a pure sinusoid, clipping will begin only as *kdist* exceeds 0.7; for a more complex input, clipping might begin at a *kdist* of 0.5 or much less. *kdist* can exceed the clip point by any amount, and may be greater than 1.

The shaping function can be made arbitrarily complex for extra effect. It should generally be continuous, though this is not a requirement. It should also be well-behaved near the mid-point, and roughly balanced positive-negative overall, else some excessive DC offset may result. The user might experiment with more aggressive functions to suit the purpose. A generally positive slope allows the distorted signal to be mixed with the source without phase cancellation.

distort is useful as an effects process, and is usually combined with reverb and chorusing on effects busses. However, it can alternatively be used to good effect within a single instrument.

Examples

```
gifn  ftgen      0,0, 257, 9, .5,1,270      ; define a sigmoid, or better
gifn  ftgen      0,0, 257, 9, .5,1,270,1.5,.33,90,2.5,.2,270,3.5,.143,90,4.5,.111,270

kdist  line      0, 10, 1.2                  ; and over 10 seconds
aout   distort   asig, kdist, gifn           ; gradually increase the distortion
```

Credits

Written by Barry L. Vercoe for Extended Csound and released in csound5.

distort1

distort1 — Modified hyperbolic tangent distortion.

Description

Implementation of modified hyperbolic tangent distortion. *distort1* can be used to generate wave shaping distortion based on a modification of the *tanh* function.

$$aout = \frac{\exp(asig * (shape1 + pregain)) - \exp(asig * (shape2 - pregain))}{\exp(asig * pregain) + \exp(-asig * pregain)}$$

Syntax

ares **distort1** asig, kpregain, kpostgain, kshape1, kshape2[, imode]

Initialization

imode (Csound version 5.00 and later only; optional, defaults to 0) -- scales kpregain, kpostgain, kshape1, and kshape2 for use with audio signals in the range -32768 to 32768 (*imode*=0), -0dbfs to 0dbfs (*imode*=1), or disables scaling of kpregain and kpostgain and scales kshape1 by kpregain and kshape2 by -kpregain (*imode*=2).

Performance

asig -- is the input signal.

kpregain -- determines the amount of gain applied to the signal before waveshaping. A value of 1 gives slight distortion.

kpostgain -- determines the amount of gain applied to the signal after waveshaping.

kshape1 -- determines the shape of the positive part of the curve. A value of 0 gives a flat clip, small positive values give sloped shaping.

kshape2 -- determines the shape of the negative part of the curve.

Examples

Here is an example of the distort1 opcode. It uses the file *distort1.csd* [examples/distort1.csd].

Example 129. Example of the distort1 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
```



```
; For Non-realtime ouput leave only the line below:
; -o distort1.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

gadist init 0

instr 1
  iamp = p4
  ifqc = cpspch(p5)
  asig pluck iamp, ifqc, ifqc, 0, 1
  gadist = gadist + asig
endin

instr 50
  kpre init p4
  kpost init p5
  kshap1 init p6
  kshap2 init p7
  aout distort1 gadist, kpre, kpost, kshap1, kshap2

  outs aout, aout

  gadist = 0
endin

</CsInstruments>
<CsScore>

;   Sta   Dur   Amp   Pitch
i1 0.0  3.0  10000  6.00
i1 0.5  2.5  10000  7.00
i1 1.0  2.0  10000  7.07
i1 1.5  1.5  10000  8.00

;   Sta   Dur   PreGain PostGain Shape1 Shape2
i50 0    3    2        1        0        0
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Hans Mikelson
December 1998

New in Csound version 3.50

divz

divz — Safely divides two numbers.

Syntax

```
ares divz xa, xb, ksubst
```

```
ires divz ia, ib, isubst
```

```
kres divz ka, kb, ksubst
```

Description

Safely divides two numbers.

Initialization

Whenever b is not zero, set the result to the value a / b ; when b is zero, set it to the value of *subst* instead.

Examples

Here is an example of the divz opcode. It uses the file *divz.csd* [examples/divz.csd].

Example 130. Example of the divz opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o divz.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Define the numbers to be divided.
ka init 200
; Linearly change the value of kb from 200 to 0.
kb line 0, p3, 200
; If a "divide by zero" error occurs, substitute -1.
ksubst init -1

; Safely divide the numbers.
kresults divz ka, kb, ksubst

; Print out the results.
printks "%f / %f = %f\\n", 0.1, ka, kb, kresults
endin

</CsInstruments>
```

```
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
200.000000 / 0.000000 = -1.000000
200.000000 / 19.999887 = 10.000056
200.000000 / 40.000027 = 4.999997
```

See Also

=, init, tival

Credits

Author: John ffitch after an idea by Barry L. Vercoe

Example written by Kevin Conder.

downsamp

downsamp — Modify a signal by down-sampling.

Description

Modify a signal by down-sampling.

Syntax

```
kres downsamp asig [, iwlen]
```

Initialization

iwlen (optional) -- window length in samples over which the audio signal is averaged to determine a downsampled value. Maximum length is *ksmps*; 0 and 1 imply no window averaging. The default value is 0.

Performance

downsamp converts an audio signal to a control signal by downsampling. It produces one kval for each audio control period. The optional window invokes a simple averaging process to suppress fold-over.

Examples

Here is an example of the *downsamp* opcode. It uses the file *downsamp.csd* [examples/downsamp.csd].

Example 131. Example of the *downsamp* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o downsamp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create a noise signal at a-rate.
anoise noise 20000, 0.2

; Downsample the noise signal to k-rate.
knoise downsamp anoise

; Use the noise signal at k-rate.
al oscil 30000, knoise, 1
out anoise
endin
```

```
</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

diff, integ, interp, samphold, upsamp

Credits

Example written by Kevin Conder.

dripwater

dripwater — Semi-physical model of a water drop.

Description

dripwater is a semi-physical model of a water drop. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares dripwater kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] \  
    [, ifreq1] [, ifreq2]
```

Initialization

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 10.

idamp (optional) -- the damping factor, as part of this equation:

$$\text{damping_amount} = 0.996 + (\text{idamp} * 0.002)$$

The default *damping_amount* is 0.996 which means that the default value of *idamp* is 0. The maximum *damping_amount* is 1.0 (no damping). This means the maximum value for *idamp* is 2.0.

The recommended range for *idamp* is usually below 75% of the maximum value. Rasmus Ekman suggests a range of 1.4-1.75. He also suggests a maximum value of 1.9 instead of the theoretical limit of 2.0.

imaxshake (optional, default=0) -- amount of energy to add back into the system. The value should be in range 0 to 1.

ifreq (optional) -- the main resonant frequency. The default value is 450.

ifreq1 (optional) -- the first resonant frequency. The default value is 600.

ifreq2 (optional) -- the second resonant frequency. The default value is 750.

Performance

kamp -- Amplitude of output. Note: As these instruments are stochastic, this is only an approximation.

Examples

Here is an example of the dripwater opcode. It uses the file *dripwater.csd* [examples/dripwater.csd].

Example 132. Example of the dripwater opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-o dac      -i adc      -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o dripwater.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 01 ;example of a water drip
a1 line 5, p3, 5 ;preset an amplitude boost
a2 dripwater p4, 0.01, 0, .9 ;dripwater needs a little amplitude help at these values
a3 product a1, a2 ;increase amplitude
out a3
endin

</CsInstruments>
<CsScore>

i1 0 1 20000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

bamboo, guiro, sleighbells, tambourine

Credits

Author: Perry Cook, part of the PhISEM (Physically Informed Stochastic Event Modeling)
Adapted by John ffitch
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

dssiactivate

dssiactivate — Activates or deactivates a DSSI or LADSPA plugin.

Syntax

```
dssiactivate ihandle, ktoggle
```

Description

dssiactivate is used to activate or deactivate a DSSI or LADSPA plugin. It calls the plugin's `activate()` and `deactivate()` functions if they are provided.

Initialization

ihandle - the number which identifies the plugin, generated by *dssiinit*.

Performance

ktoggle - Selects between activation (*ktoggle*=1) and deactivation (*ktoggle*=0).

dssiactivate is used to turn on and off plugins if they provide this facility. This may help conserve CPU processing in some cases. For consistency, all plugins must be activated to produce sound. An inactive plugin produces silence.

Depending on the plugin's implementation, this may cause interruptions in the realtime audio process, so use with caution.

dssiactivate may cause audio stream breakups when used in realtime, so it is recommended to load all plugins to be used before playing.



Warning

Please note that even if `activate()` and `deactivate()` functions are not present in a plugin, *dssiactivate* must be called for the plugin to produce sound.

Credits

2005

By: Andrés Cabrera

Uses code from Richard Furse's LADSPA sdk.

dssiaudio

dssiaudio — Processes audio using a LADSPA or DSSI plugin.

Syntax

```
aout1 [, aout2, aout3, aout4] dssiaudio ihandle, ain1 [,ain2, ain3, ain4]
```

Description

dssiaudio generates audio by processing an input signal through a LADSPA plugin.

Initialization

ihandle - handle for the plugin returned by *dssiinit*

Performance

aout1, aout2, etc - Audio output generated by the plugin

ain1, ain2, etc - Audio provided to the plugin for processing

dssiaudio runs a plugin on the provided audio and produces audio output. Currently upto four inputs and outputs are provided. You should provide signal for all the plugins audio inputs, otherwise unpredictable results may occur. If the plugin doesn't have any input (e.g Noise generator) you must still provide at least one input variable, which will be ignored with a message.

Only one *dssiaudio* should be executed once per plugin, or strange results may occur.

Credits

2005

By: Andrés Cabrera

Uses code from Richard Furse's LADSPA sdk.

dssictls

dssictls — Send control information to a LADSPA or DSSI plugin.

Syntax

```
dssictls ihandle, iport, kvalue, ktrigger
```

Description

dssictls sends control values to a plugin's control port

Initialization

ihandle - handle for the plugin returned by *dssiinit*

iport - control port number

Performance

kvalue - value to be assigned to the port

ktrigger - determines whether the control information will be sent (*ktrigger* = 1) or not. This is useful for thinning control information, generating *ktrigger* with *metro*

dssictls sends control information to a LADSPA or DSSI plugin's control port. The valid control ports and ranges are given by *dssiinit*. Using values outside the ranges may produce unspecified behaviour.

Credits

2005

By: Andrés Cabrera

Uses code from Richard Furse's LADSPA sdk.

dssiinit

dssiinit — Loads a DSSI or LADSPA plugin.

Syntax

```
ihandle dssiinit ilibraryname, ipluginindex [, iverbose]
```

Description

dssiinit is used to load a DSSI or LADSPA plugin into memory for use with the other dssi4cs opcodes. Both LADSPA effects and DSSI instruments can be used.

Initialization

ihandle - the number which identifies the plugin, to be passed to other dssi4cs opcodes.

ilibraryname - the name of the .so (shared object) file to load.

ipluginindex - The index of the plugin to be used.

iverbose (optional) - show plugin information and parameters when loading. (default = 1)

dssiinit looks for *ilibraryname* on LADSPA_PATH and DSSI_PATH. One of these variables must be set, otherwise *dssiinit* will return an error. LADSPA and DSSI libraries may contain more than one plugin which must be referenced by its index. *dssiinit* then attempts to find plugin index *ipluginindex* in the library and load the plugin into memory if it is found. To find out which plugins you have available and their index numbers you can use: *dssilist*.

If *iverbose* is not 0 (the default), information about the plugin detailing its characteristics and its ports will be shown. This information is important for opcodes like *dssictls*.

Plugins are set to inactive by default, so you **must** use *dssiactivate* to get the plugin to produce sound. This is required even if the plugin doesn't provide an activate() function.

dssiinit may cause audio stream breakups when used in realtime, so it is recommended to load all plugins to be used before playing.

Examples

Here is an example of the dssinit opcode. It uses the file *dssi4cs.csd* [examples/dssi4cs.csd].

Example 133. Example of the dssiinit opcode. (Remember to change the Library name)

```
<CsoundSynthesizer>
<CsOptions>
;use appropriate realtime options
</CsOptions>
<CsInstruments>
ksmps = 256
nchnls = 2

dssilist

gihandle dssiinit "amp.so", 0, 1
;gihandle dssiinit "cmt.so", 30, 2
;gihandle2 dssiinit "cmt.so", 8, 1
;gihandle dssiinit "delayorama_1402", 0
```

```
gihandle2 dssiinit "cmt.so", 49 , 1
;gihandle dssiinit "freq_tracker_1418.so", 0 , 1, 1
;gihandle dssiinit "g2reverb.so", 0, 1
;gihandle2 dssiinit "declip_1195.so", 0, 1
;gihandle2 dssiinit "revdelay_1605.so", 0, 1
;gihandle2 dssiinit "tap_chorusflanger.so", 0, 1
;gihandle2 dssiinit "plate_1423.so", 0, 1
gihandle3 dssiinit "gate_1410.so", 0, 1
;gihandle3 dssiinit "hexter.so", 0, 1

instr 1
print p4
dssiactivate gihandle, p4
dssiactivate gihandle2, p4
dssiactivate gihandle3, p4
endin

instr 2
ain1 inch 1
ain2 inch 2
;aout1,aout2 dssiaudio gihandle, ain1, ain2
aout1 dssiaudio gihandle, ain1
outs aout1,aout1
endin

instr 3
kval linen 1, p3 /3, p3, p3/ 3
dssiactls gihandle, p4, kval, 1
endin

instr 4
ain1 inch 1
aout1 dssiaudio gihandle2, ain1
outs aout1,aout1
endin

</CsInstruments>
<CsScore>

i 1 1 1 1

i 2 2 15 ;plugin 1

i 3 3 12 0 ;Control port 0

i 4 8 2 ;plugin 2
e
</CsScore>
</CsoundSynthesizer>
```

Credits

2005

By: Andrés Cabrera

Uses code from Richard Furse's LADSPA sdk.

dssilist

dssilist — Lists all available DSSI and LADSPA plugins.

Syntax

`dssilist`

Description

dssilist checks the variables DSSI_PATH and LADSPA_PATH and lists all plugins available in all plugin libraries there.

LADSPA and DSSI libraries may contain more than one plugin which must be referenced by the index provided by *dssilist*.

This opcode produces a long printout which may interrupt realtime audio output, so it should be run at the start of a performance.

Credits

2005

By: Andrés Cabrera

Uses code from Richard Furse's LADSPA sdk.

dumpk

dumpk — Periodically writes an orchestra control-signal value to an external file.

Description

Periodically writes an orchestra control-signal value to a named external file in a specific format.

Syntax

```
dumpk ksig, ifilename, iformat, iprd
```

Initialization

ifilename -- character string (in double quotes, spaces permitted) denoting the external file name. May either be a full path name with target directory specified or a simple filename to be created within the current directory

iformat -- specifies the output data format:

- 1 = 8-bit signed char(high order 8 bits of a 16-bit integer)
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = ASCII long integers
- 8 = ASCII floats (2 decimal places)

Note that A-law and U-law output are not available, and that all formats except the last two are binary. The output file contains no header information.

iprd -- the period of *ksig* output *i* seconds, rounded to the nearest orchestra control period. A value of 0 implies one control period (the enforced minimum), which will create an output file sampled at the orchestra control rate.

Performance

ksig -- a control-rate signal

This opcode allows a generated control signal value to be saved in a named external file. The file contains no self-defining header information. But it contains a regularly sampled time series, suitable for later input or analysis. There may be any number of *dumpk* opcodes in an instrument or orchestra but each must write to a different file.

Examples

Here is an example of the dumpk opcode. It uses the file *dumpk.csd* [examples/dumpk.csd].

Example 134. Example of the dumpk opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o dumpk.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 20
nchnls = 1

; By Andres Cabrera 2008

instr 1
; Write fibonacci numbers to file "fibonacci.txt"
; as ascii long integers (mode 7), using the orchestra's
; control rate (iprd = 0)

knumber init 0
koldnumber init 1
ktrans init 1
ktrans = knumber
knumber = knumber + koldnumber
koldnumber = ktrans
dumpk knumber, "fibonacci.txt", 7, 0
printk2 knumber
endin

</CsInstruments>
<CsScore>

;Write to the file for 1 second. Since control rate is 20, 20 values will be written
i 1 0 1

</CsScore>
</CsoundSynthesizer>
```

See Also

dumpk2, dumpk3, dumpk4, readk, readk2, readk3, readk4

Credits

By: John ffitch and Barry Vercoe

1999 or earlier

dumpk2

dumpk2 — Periodically writes two orchestra control-signal values to an external file.

Description

Periodically writes two orchestra control-signal values to a named external file in a specific format.

Syntax

```
dumpk2 ksig1, ksig2, ifilename, iformat, iprd
```

Initialization

ifilename -- character string (in double quotes, spaces permitted) denoting the external file name. May either be a full path name with target directory specified or a simple filename to be created within the current directory

iformat -- specifies the output data format:

- 1 = 8-bit signed char(high order 8 bits of a 16-bit integer)
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = ASCII long integers
- 8 = ASCII floats (2 decimal places)

Note that A-law and U-law output are not available, and that all formats except the last two are binary. The output file contains no header information.

iprd -- the period of *ksig* output *i* seconds, rounded to the nearest orchestra control period. A value of 0 implies one control period (the enforced minimum), which will create an output file sampled at the orchestra control rate.

Performance

ksig1, *ksig2* -- control-rate signals.

This opcode allows two generated control signal values to be saved in a named external file. The file contains no self-defining header information. But it contains a regularly sampled time series, suitable for later input or analysis. There may be any number of *dumpk2* opcodes in an instrument or orchestra but each must write to a different file.

Examples

See the example for *dumpk*. The only difference between *dumpk* and *dumpk2* is that *dumpk2* can write two values at a time from the file.

See Also

dumpk, dumpk3, dumpk4, readk, readk2, readk3, readk4

Credits

By: John ffitch and Barry Vercoe

1999 or earlier

dumpk3

dumpk3 — Periodically writes three orchestra control-signal values to an external file.

Description

Periodically writes three orchestra control-signal values to a named external file in a specific format.

Syntax

```
dumpk3 ksig1, ksig2, ksig3, ifilename, iformat, iprd
```

Initialization

ifilename -- character string (in double quotes, spaces permitted) denoting the external file name. May either be a full path name with target directory specified or a simple filename to be created within the current directory

iformat -- specifies the output data format:

- 1 = 8-bit signed char(high order 8 bits of a 16-bit integer)
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = ASCII long integers
- 8 = ASCII floats (2 decimal places)

Note that A-law and U-law output are not available, and that all formats except the last two are binary. The output file contains no header information.

iprd -- the period of *ksig* output *i* seconds, rounded to the nearest orchestra control period. A value of 0 implies one control period (the enforced minimum), which will create an output file sampled at the orchestra control rate.

Performance

ksig1, *ksig2*, *ksig3* -- control-rate signals

This opcode allows three generated control signal values to be saved in a named external file. The file contains no self-defining header information. But it contains a regularly sampled time series, suitable for later input or analysis. There may be any number of *dumpk3* opcodes in an instrument or orchestra but each must write to a different file.

Examples

See the example for *dumpk*. The only difference between *dumpk* and *dumpk3* is that *dumpk3* can write three values at a time from the file.

See Also

dumpk, dumpk2, dumpk4, readk, readk2, readk3, readk4

Credits

By: John ffitch and Barry Vercoe

1999 or earlier

dumpk4

dumpk4 — Periodically writes four orchestra control-signal values to an external file.

Description

Periodically writes four orchestra control-signal values to a named external file in a specific format.

Syntax

```
dumpk4 ksig1, ksig2, ksig3, ksig4, ifilename, iformat, iprd
```

Initialization

ifilename -- character string (in double quotes, spaces permitted) denoting the external file name. May either be a full path name with target directory specified or a simple filename to be created within the current directory

iformat -- specifies the output data format:

- 1 = 8-bit signed char(high order 8 bits of a 16-bit integer)
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = ASCII long integers
- 8 = ASCII floats (2 decimal places)

Note that A-law and U-law output are not available, and that all formats except the last two are binary. The output file contains no header information.

iprd -- the period of *ksig* output *i* seconds, rounded to the nearest orchestra control period. A value of 0 implies one control period (the enforced minimum), which will create an output file sampled at the orchestra control rate.

Performance

ksig1, ksig2, ksig3, ksig4 -- control-rate signals

This opcode allows four generated control signal values to be saved in a named external file. The file contains no self-defining header information. But it contains a regularly sampled time series, suitable for later input or analysis. There may be any number of *dumpk4* opcodes in an instrument or orchestra but each must write to a different file.

Examples

See the example for *dumpk*. The only difference between *dumpk* and *dumpk4* is that *dumpk4* can write four values at a time from the file.

See Also

dumpk, dumpk2, dumpk3, readk, readk2, readk3, readk4

Credits

By: John ffitch and Barry Vercoe

1999 or earlier

duserrnd

duserrnd — Discrete USER-defined-distribution RaNDom generator.

Description

Discrete USER-defined-distribution RaNDom generator.

Syntax

```
aout duserrnd ktableNum
```

```
iout duserrnd itableNum
```

```
kout duserrnd ktableNum
```

Initialization

itableNum -- number of table containing the random-distribution function. Such table is generated by the user. See GEN40, GEN41, and GEN42. The table length does not need to be a power of 2

Performance

ktableNum -- number of table containing the random-distribution function. Such table is generated by the user. See GEN40, GEN41, and GEN42. The table length does not need to be a power of 2

duserrnd (discrete user-defined-distribution random generator) generates random values according to a discrete random distribution created by the user. The user can create the discrete distribution histogram by using GEN41. In order to create that table, the user has to define an arbitrary amount of number pairs, the first number of each pair representing a value and the second representing its probability (see GEN41 for more details).

When used as a function, the rate of generation depends by the rate type of input variable *XtableNum*. In this case it can be embedded into any formula. Table number can be varied at k-rate, allowing to change the distribution histogram during the performance of a single note. *duserrnd* is designed be used in algorithmic music generation.

duserrnd can also be used to generate values following a set of ranges of probabilities by using distribution functions generated by GEN42 (See GEN42 for more details). In this case, in order to simulate continuous ranges, the length of table *XtableNum* should be reasonably big, as *duserrnd* does not interpolate between table elements.

For a tutorial about random distribution histograms and functions see:

- D. Lorrain. "A panoply of stochastic cannons". In C. Roads, ed. 1989. Music machine. Cambridge, Massachusetts: MIT press, pp. 351 - 379.

See Also

cuserrnd, *urd*

Credits

Author: Gabriel Maldonado

New in Version 4.16

else

else — Executes a block of code when an "if...then" condition is false.

Description

Executes a block of code when an "if...then" condition is false.

Syntax

`else`

Performance

else is used inside of a block of code between the *"if...then"* and *endif* opcodes. It defines which statements are executed when a "if...then" condition is false. Only one *else* statement may occur and it must be the last conditional statement before the *endif* opcode.

Examples

See the example for the *if* opcode.

See Also

elseif, *endif*, *goto*, *if*, *igoto*, *kgoto*, *tigoto*, *timeout*

Credits

New in version 4.21

elseif

elseif — Defines another "if...then" condition when a "if...then" condition is false.

Description

Defines another "if...then" condition when a "if...then" condition is false.

Syntax

```
elseif xa R xb then
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

Performance

elseif is used inside of a block of code between the "*if...then*" and *endif* opcodes. When a "if...then" condition is false, it defines another "if...then" condition to be met. Any number of *elseif* statements are allowed.

Examples

See the example for the *if* opcode.

See Also

else, *endif*, *goto*, *if*, *igoto*, *kgoto*, *tigoto*, *timeout*

Credits

New in version 4.21

endif

endif — Closes a block of code that begins with an "if...then" statement.

Description

Closes a block of code that begins with an "*if...then*" statement.

Syntax

`endif`

Performance

Any block of code that begins with an "*if...then*" statement must end with an *endif* statement.

Examples

See the example for the *if* opcode.

See Also

elseif, else, goto, if, igoto, kgoto, tigoto, timeout

Credits

New in version 4.21

endin

endin — Ends the current instrument block.

Description

Ends the current instrument block.

Syntax

endin

Initialization

Ends the current instrument block.

Instruments can be defined in any order (but they will always be both initialized and performed in ascending instrument number order). Instrument blocks cannot be nested (i.e. one block cannot contain another).



Note

There may be any number of instrument blocks in an orchestra.

Examples

Here is an example of the endin opcode. It uses the file *endin.csd* [examples/endin.csd].

Example 135. Example of the endin opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o endin.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  iamp = 10000
  icps = 440
  iphs = 0

  a1 oscils iamp, icps, iphs
  out a1
endin

</CsInstruments>
<CsScore>
```

```
; Play Instrument #1 for 2 seconds.  
i 1 0 2  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

instr

Credits

Example written by Kevin Conder.

endop

endop — Marks the end of an user-defined opcode block.

Description

Marks the end of an user-defined opcode block.

Syntax

endop

Performance

The syntax of a user-defined opcode block is as follows:

```
opcode name, outtypes, intypes
xinarg1 [, xinarg2] [, xinarg3] ... [xinargN] xin
[setksmps iksmps]
... the rest of the instrument's code.
xout xoutarg1 [, xoutarg2] [, xoutarg3] ... [xoutargN]
endop
```

The new opcode can then be used with the usual syntax:

```
[xinarg1] [, xinarg2] ... [xinargN] name [xoutarg1] [, xoutarg2] ... [xoutargN] [, iksmps]
```

Examples

See the example for the *opcode* opcode.

See Also

opcode, *setksmps*, *xin*, *xout*

Credits

Author: Istvan Varga, 2002; based on code by Matt J. Ingalls

New in version 4.22

envlpx

envlpx — Applies an envelope consisting of 3 segments.

Description

envlpx -- apply an envelope consisting of 3 segments:

1. stored function rise shape
2. modified exponential pseudo steady state
3. exponential decay

Syntax

```
ares envlpx xamp, irise, idur, idec, ifn, iatss, iatdec [, ixmod]
```

```
kres envlpx kamp, irise, idur, idec, ifn, iatss, iatdec [, ixmod]
```

Initialization

irise -- rise time in seconds. A zero or negative value signifies no rise modification.

idur -- overall duration in seconds. A zero or negative value will cause initialization to be skipped.

idec -- decay time in seconds. Zero means no decay. An *idec* > *idur* will cause a truncated decay.

ifn -- function table number of stored rise shape with extended guard point.

iatss -- attenuation factor, by which the last value of the *envlpx* rise is modified during the note's pseudo steady state. A factor greater than 1 causes an exponential growth and a factor less than 1 creates an exponential decay. A factor of 1 will maintain a true steady state at the last rise value. Note that this attenuation is not by fixed rate (as in a piano), but is sensitive to a note's duration. However, if *iatss* is negative (or if steady state < 4 k-periods) a fixed attenuation rate of *abs(iatss)* per second will be used. 0 is illegal.

iatdec -- attenuation factor by which the closing steady state value is reduced exponentially over the decay period. This value must be positive and is normally of the order of .01. A large or excessively small value is apt to produce a cutoff which is audible. A zero or negative value is illegal.

ixmod (optional, between +- .9 or so) -- exponential curve modifier, influencing the steepness of the exponential trajectory during the steady state. Values less than zero will cause an accelerated growth or decay towards the target (e.g. *subito piano*). Values greater than zero will cause a retarded growth or decay. The default value is zero (unmodified exponential).

Performance

kamp, *xamp* -- input amplitude signal.

Rise modifications are applied for the first *irise* seconds, and decay from time *idur* - *idec*. If these periods are separated in time there will be a steady state during which *amp* will be modified by the first exponential pattern. If rise and decay periods overlap then both modifications will be in effect for that time. If the overall duration *idur* is exceeded in performance, the final decay will continue on in the same direction, tending asymptotically to zero.

Examples

Here is an example of the `envlpx` opcode. It uses the file `envlpx.csd` [examples/envlpx.csd].

Example 136. Example of the `envlpx` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o envlpx.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a simple instrument.
instr 1
; Set the amplitude.
kamp init 20000
; Get the frequency from the fourth p-field.
kcps = cpspch(p4)

al vco kamp, kcps, 1
out al
endin

; Instrument #2 - instrument with an amplitude envelope.
instr 2
kamp = 20000
irise = 0.05
idur = p3 - .01
idec = 0.5
ifn = 2
iatss = 1
iatdec = 0.01

; Create an amplitude envelope.
kenv envlpx kamp, irise, idur, idec, ifn, iatss, iatdec

; Get the frequency from the fourth p-field.
kcps = cpspch(p4)

al vco kenv, kcps, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1
; Table #2, a rising envelope.
f 2 0 129 -7 0 128 1

; Set the tempo to 120 beats per minute.
t 0 120

; Make sure the score plays for 33 seconds.
f 0 33

; Play a melody with Instrument #1.
; p4 = frequency in pitch-class notation.
i 1 0 1 8.04
i 1 1 1 8.04
i 1 2 1 8.05
i 1 3 1 8.07
i 1 4 1 8.07
i 1 5 1 8.05
i 1 6 1 8.04
```

```
i 1 7 1 8.02
i 1 8 1 8.00
i 1 9 1 8.00
i 1 10 1 8.02
i 1 11 1 8.04
i 1 12 2 8.04
i 1 14 2 8.02

; Repeat the melody with Instrument #2.
; p4 = frequency in pitch-class notation.
i 2 16 1 8.04
i 2 17 1 8.04
i 2 18 1 8.05
i 2 19 1 8.07
i 2 20 1 8.07
i 2 21 1 8.05
i 2 22 1 8.04
i 2 23 1 8.02
i 2 24 1 8.00
i 2 25 1 8.00
i 2 26 1 8.02
i 2 27 1 8.04
i 2 28 2 8.04
i 2 30 2 8.02
e

</CsScore>
</CsoundSynthesizer>
```

See Also

envlpxr, linen, linenr

Credits

Thanks goes to Luis Jure for pointing out a mistake with *iatss*.

Example written by Kevin Conder.

envlpxr

envlpxr — The *envlpx* opcode with a final release segment.

Description

envlpxr is the same as *envlpx* except that the final segment is entered only on sensing a MIDI note release. The note is then extended by the decay time.

Syntax

```
ares envlpxr xamp, irise, idec, ifn, iatss, iatdec [, ixmod] [, irind]
```

```
kres envlpxr kamp, irise, idec, ifn, iatss, iatdec [, ixmod] [, irind]
```

Initialization

irise -- rise time in seconds. A zero or negative value signifies no rise modification.

idec -- decay time in seconds. Zero means no decay.

ifn -- function table number of stored rise shape with extended guard point.

iatss -- attenuation factor, by which the last value of the *envlpxr* rise is modified during the note's pseudo steady state. A factor greater than 1 causes an exponential growth and a factor less than 1 creates an exponential decay. A factor of 1 will maintain a true steady state at the last rise value. Note that this attenuation is not by fixed rate (as in a piano), but is sensitive to a note's duration. However, if *iatss* is negative (or if steady state < 4 k-periods) a fixed attenuation rate of *abs(iatss)* per second will be used. 0 is illegal.

iatdec -- attenuation factor by which the closing steady state value is reduced exponentially over the decay period. This value must be positive and is normally of the order of .01. A large or excessively small value is apt to produce a cutoff which is audible. A zero or negative value is illegal.

ixmod (optional, between +- .9 or so) -- exponential curve modifier, influencing the steepness of the exponential trajectory during the steady state. Values less than zero will cause an accelerated growth or decay towards the target (e.g. *subito piano*). Values greater than zero will cause a retarded growth or decay. The default value is zero (unmodified exponential).

irind (optional) -- independence flag. If left zero, the release time (*idec*) will influence the extended life of the current note following a note-off. If non-zero, the *idec* time is quite independent of the note extension (see below). The default value is 0.

Performance

kamp, *xamp* -- input amplitude signal.

envlpxr is an example of the special Csound “r” units that contain a note-off sensor and release time extender. When each senses a score event termination or a MIDI noteoff, it will immediately extend the performance time of the current instrument by *idec* seconds unless it is made independent by *irind*. Then it will begin a decay from wherever it was at the time.

You can use other pre-made envelopes which start a release segment upon receiving a note off message, like *linsegr* and *expsegr*, or you can construct more complex envelopes using *xtratim* and *release*. Note that you don't need to use *xtratim* if you are using *envlpxr*, since the time is extended automatically.

These “r” units can also be modified by MIDI noteoff velocities (see veloffs). If the *irind* flag is on (non-zero), the overall performance time is unaffected by note-off and veloff data.

Multiple “r” units. When two or more “r” units occur in the same instrument it is usual to have only one of them influence the overall note duration. This is normally the master amplitude unit. Other units controlling, say, filter motion can still be sensitive to note-off commands while not affecting the duration by making them independent (*irind* non-zero). Depending on their own *idec* (release time) values, independent “r” units may or may not reach their final destinations before the instrument terminates. If they do, they will simply hold their target values until termination. If two or more “r” units are simultaneously master, note extension is by the greatest *idec*.

See Also

envlpx, linen, linenr

Credits

Thanks goes to Luis Jure for pointing out a mistake with *iatss*.

ephasor

ephasor —

Performance

Ephasor has been added to Csound 5.10, but its behavior will change for 5.11. Stay tuned...

Credits

Author: Victor Lazzarini
2008

New in version 5.10

eqfil

eqfil — Equalizer filter

Description

The opcode eqfil is a 2nd order tunable equalisation filter based on Regalia and Mitra design ("Tunable Digital Frequency Response Equalization Filters", IEEE Trans. on Ac., Sp. and Sig Proc., 35 (1), 1987). It provides a peak/notch filter for building parametric/graphic equalisers.

The amplitude response for this filter will be flat (=1) for kgain=1. With kgain is bigger than 1, there will be a peak at the centre frequency, whose width is given by the kbw parameter, but outside this band, the response will tend towards 1. Conversely, if kgain is smaller than 1, a notch will be created around the CF.

Syntax

```
asig eqfil ain, kcf, kbw, kgain[, istor]
```

Initialization

istor --initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- filtered output signal.

ain -- input signal.

kcf -- filter centre frequency

kbw -- peak/notch bandwidth (Hz).

kgain -- peak/notch gain.

Examples

Example 137. Example

```
kfe      expseg 10, p3*0.9, 180, p3*0.1, 175
kenv     linen 1000, 0.05, p3, 0.05
asig     buzz  kenv, kfe, sr/(2*kfe), 1
afil     eqfil asig, 1500, 400, 0.1

out afil
```

Credits

Author: Victor Lazzarini;
April 2007

New in version 5.06

event

event — Generates a score event from an instrument.

Description

Generates a score event from an instrument.

Syntax

```
event "scorechar", kinsnum, kdelay, kdur, [, kp4] [, kp5] [, ...]
```

```
event "scorechar", "insname", kdelay, kdur, [, kp4] [, kp5] [, ...]
```

Initialization

“scorechar” -- A string (in double-quotes) representing the first p-field in a score statement. This is usually *“e”*, *“f”*, or *“i”*.

“insname” -- A string (in double-quotes) representing a named instrument.

Performance

kinsnum -- The instrument to use for the event. This corresponds to the first p-field, p1, in a score statement.

kdelay -- When (in seconds) the event will occur from the current performance time. This corresponds to the second p-field, p2, in a score statement.

kdur -- How long (in seconds) the event will happen. This corresponds to the third p-field, p3, in a score statement.

kp4, *kp5*, ... (optional) -- Parameters representing additional p-field in a score statement. It starts with the fourth p-field, p4.

Examples

Here is an example of the event opcode. It uses the file *event.csd* [examples/event.csd].

Example 138. Example of the event opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o event.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
```

```
nchnls = 1

; Instrument #1 - an oscillator with a high note.
instr 1
; Create a trigger and set its initial value to 1.
ktrigger init 1

; If the trigger is equal to 0, continue playing.
; If not, schedule another event.
if (ktrigger == 0) goto contin
; kscoreop="i", an i-statement.
; kinsnum=2, play Instrument #2.
; kwhen=1, start at 1 second.
; kdur=0.5, play for a half-second.
event "i", 2, 1, 0.5

; Make sure the event isn't triggered again.
ktrigger = 0

contin:
al oscils 10000, 440, 1
out al
endin

; Instrument #2 - an oscillator with a low note.
instr 2
al oscils 10000, 220, 1
out al
endin

</CsInstruments>
<CsScore>

; Make sure the score plays for two seconds.
f 0 2

; Play Instrument #1 for a half-second.
i 1 0 0.5
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the event opcode using a named instrument. It uses the file *event_named.csd* [examples/event_named.csd].

Example 139. Example of the event opcode using a named instrument.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o event_named.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - an oscillator with a high note.
instr 1
; Create a trigger and set its initial value to 1.
ktrigger init 1

; If the trigger is equal to 0, continue playing.
; If not, schedule another event.
if (ktrigger == 0) goto contin
; kscoreop="i", an i-statement.
; kinsnum="low_note", instrument named "low_note".
; kwhen=1, start at 1 second.
; kdur=0.5, play for a half-second.
event "i", "low_note", 1, 0.5
```

```
    ; Make sure the event isn't triggered again.
    ktrigger = 0

contin:
    al oscils 10000, 440, 1
    out al
endin

; Instrument "low_note" - an oscillator with a low note.
instr low_note
    al oscils 10000, 220, 1
    out al
endin

</CsInstruments>
<CsScore>

; Make sure the score plays for two seconds.
f 0 2

; Play Instrument #1 for a half-second.
i 1 0 0.5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Examples written by Kevin Conder.

New in version 4.17

Thanks goes to Matt Ingalls for helping to fix the example.

Thanks goes to Matt Ingalls for helping clarify the kwhen/kdelay parameter.

event_i

event_i — Generates a score event from an instrument.

Description

Generates a score event from an instrument.

Syntax

```
event_i "scorechar", iinsnum, idelay, idur, [, ip4] [, ip5] [, ...]
```

```
event_i "scorechar", "insname", idelay, idur, [, ip4] [, ip5] [, ...]
```

Initialization

“scorechar” -- A string (in double-quotes) representing the first p-field in a score statement. This is usually *“e”*, *“f”*, or *“i”*.

“insname” -- A string (in double-quotes) representing a named instrument.

iinsnum -- The instrument to use for the event. This corresponds to the first p-field, p1, in a score statement.

idelay -- When (in seconds) the event will occur from the current performance time. This corresponds to the second p-field, p2, in a score statement.

idur -- How long (in seconds) the event will happen. This corresponds to the third p-field, p3, in a score statement.

ip4, ip5, ... (optional) -- Parameters representing additional p-field in a score statement. It starts with the fourth p-field, p4.

Performance

The event is added to the queue at initialisation time.

Credits

Written by Istvan Varga.

New in Csound5

exitnow

exitnow — Exit csound as fast as possible, with no cleaning up.

Description

In Csound4 calls an exit function to leave csoundn as fast as possible. On Csound5 exits back to the driving code.

Syntax

`exitnow`

Performance

Stops Csound on the initialisation cycle.

exp

exp — Returns e raised to the x-th power.

Description

Returns e raised to the xth power.

Syntax

exp(x) (no rate restriction)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

Examples

Here is an example of the exp opcode. It uses the file *exp.csd* [examples/exp.csd].

Example 140. Example of the exp opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o exp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = exp(8)
  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 2980.958
```

See Also

abs, frac, int, log, log10, i, sqrt

Credits

Example written by Kevin Conder.

New in version 4.21

expcurve

expcurve — This opcode implements a formula for generating a normalised exponential curve in range 0 - 1. It is based on the Max / MSP work of Eric Singer (c) 1994.

Description

Generates an exponential curve in range 0 to 1 of arbitrary steepness. Steepness index equal to or lower than 1.0 will result in Not-a-Number errors and cause unstable behavior.

The formula used to calculate the curve is:

$$(\exp(x * \log(y)) - 1) / (y - 1)$$

where x is equal to *kindex* and y is equal to *ksteepness*.

Syntax

kout **expcurve** kindex, ksteepness

Performance

kindex -- Index value. Expected range 0 to 1.

ksteepness -- Steepness of the generated curve. Values closer to 1.0 result in a straighter line while larger values steepen the curve.

kout -- Scaled output.

Examples

Here is an example of the expcurve opcode. It uses the file *expcurve.csd* [examples/expcurve.csd].

Example 141. Example of the expcurve opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  Silent
-odac      -idac      -d      ;;realtime output
</CsOptions>
<CsInstruments>

sr = 48000
ksmps = 100
nchnls = 2

/*---- */

instr 1 ; logcurve test

kmod phasor 1/200

kout expcurve kmod, 2

printk2 kmod
printk2 kout

endin
```

```
/*--- ---*/  
</CsInstruments>  
<CsScore>  
  
i1 0 8888  
  
e  
</CsScore>  
</CsoundSynthesizer>
```

See Also

scale, gainslider, logcurve

Credits

Author: David Akbari
October
2006

expon

expon — Trace an exponential curve between specified points.

Description

Trace an exponential curve between specified points.

Syntax

```
ares expon ia, idur, ib
```

```
kres expon ia, idur, ib
```

Initialization

ia -- starting value. Zero is illegal for exponentials.

ib -- value after *idur* seconds. For exponentials, must be non-zero and must agree in sign with *ia*.

idur -- duration in seconds of the segment. A zero or negative value will cause all initialization to be skipped.

Performance

These units generate control or audio signals whose values can pass through 2 specified points. The *idur* value may or may not equal the instrument's performance time: a shorter performance will truncate the specified pattern, while a longer one will cause the defined segment to continue on in the same direction.

Examples

Here is an example of the expon opcode. It uses the file *expon.csd* [examples/expon.csd].

Example 142. Example of the expon opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o expon.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Define kcps as a frequency value that exponentially declines
; from 880 to 220. It declines over the period set by p3.
kcps expon 880, p3, 220
```

```
al oscil 20000, kcps, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

expseg, expsegr, line, linseg, linsegr

Credits

Example written by Kevin Conder.

exprand

exprand — Exponential distribution random number generator (positive values only).

Description

Exponential distribution random number generator (positive values only). This is an x-class noise generator.

Syntax

ares **exprand** klambda

ires **exprand** klambda

kres **exprand** klambda

Performance

klambda -- lambda parameter for the exponential distribution.

The probability density function of an exponential distribution is an exponential curve, whose mean is $0.69515/\text{lambda}$. For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the exprand opcode. It uses the file *exprand.csd* [examples/exprand.csd].

Example 143. Example of the exprand opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o exprand.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random between 0 and 1.
; krange = 1
```

```
i1 exprand 1
print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 0.174
```

See Also

seed, betarand, bexprnd, cauchy, gauss, linrand, pcauchy, poisson, trirand, unirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder.

expseg

expseg — Trace a series of exponential segments between specified points.

Description

Trace a series of exponential segments between specified points.

Syntax

```
ares expseg ia, idur1, ib [, idur2] [, ic] [...]
```

```
kres expseg ia, idur1, ib [, idur2] [, ic] [...]
```

Initialization

ia -- starting value. Zero is illegal for exponentials.

ib, *ic*, etc. -- value after *dur1* seconds, etc. For exponentials, must be non-zero and must agree in sign with *ia*.

idur1 -- duration in seconds of first segment. A zero or negative value will cause all initialization to be skipped.

idur2, *idur3*, etc. -- duration in seconds of subsequent segments. A zero or negative value will terminate the initialization process with the preceding point, permitting the last-defined line or curve to be continued indefinitely in performance. The default is zero.

Performance

These units generate control or audio signals whose values can pass through 2 or more specified points. The sum of *dur* values may or may not equal the instrument's performance time: a shorter performance will truncate the specified pattern, while a longer one will cause the last-defined segment to continue on in the same direction.

Note that the *expseg* opcode does not operate correctly at audio rate when segments are shorter than a k-period. Try the *expsega* opcode instead.

Examples

Here is an example of the *expseg* opcode. It uses the file *expseg.csd* [examples/expseg.csd].

Example 144. Example of the expseg opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o expseg.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```

```
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; p4 = frequency in pitch-class notation.
kcps = cpspch(p4)

; Create an amplitude envelope.
kenv expseg 0.01, p3*0.25, 1, p3*0.75, 0.01
kamp = kenv * 30000

a1 oscil kamp, kcps, 1
out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for a half-second, p4=8.00
i 1 0 0.5 8.00
; Play Instrument #1 for a half-second, p4=8.01
i 1 1 0.5 8.01
; Play Instrument #1 for a half-second, p4=8.02
i 1 2 0.5 8.02
; Play Instrument #1 for a half-second, p4=8.03
i 1 3 0.5 8.03
e

</CsScore>
</CsoundSynthesizer>
```

See Also

expon, expsega, expsegr, line, linseg, linsegr transeg

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

New in Csound 3.57

expsega

expsega — An exponential segment generator operating at a-rate.

Description

An exponential segment generator operating at a-rate. This unit is almost identical to *expseg*, but more precise when defining segments with very short durations (i.e., in a percussive attack phase) at audio rate.

Syntax

```
ares expsega ia, idur1, ib [, idur2] [, ic] [...]
```

Initialization

ia -- starting value. Zero is illegal.

ib, *ic*, etc. -- value after *idur1* seconds, etc. must be non-zero and must agree in sign with *ia*.

idur1 -- duration in seconds of first segment. A zero or negative value will cause all initialization to be skipped.

idur2, *idur3*, etc. -- duration in seconds of subsequent segments. A zero or negative value will terminate the initialization process with the preceding point, permitting the last defined line or curve to be continued indefinitely in performance. The default is zero.

Performance

This unit generate audio signals whose values can pass through two or more specified points. The sum of *dur* values may or may not equal the instrument's performance time. A shorter performance will truncate the specified pattern, while a longer one will cause the last defined segment to continue on in the same direction.

Examples

Here is an example of the expsega opcode. It uses the file *expsega.csd* [examples/expsega.csd].

Example 145. Example of the expsega opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o expsega.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

```
; Instrument #1.
instr 1
; Define a short percussive amplitude envelope that
; goes from 0.01 to 20,000 and back.
aenv expsega 0.01, 0.1, 20000, 0.1, 0.01

al oscil aenv, 440, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #1 for one second.
i 1 1 1
; Play Instrument #1 for one second.
i 1 2 1
; Play Instrument #1 for one second.
i 1 3 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

expseg, expsegr

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

New in Csound 3.57

expsegr

expsegr — Trace a series of exponential segments between specified points including a release segment.

Description

Trace a series of exponential segments between specified points including a release segment.

Syntax

```
ares expsegr ia, idur1, ib [, idur2] [, ic] [...], irel, iz
```

```
kres expsegr ia, idur1, ib [, idur2] [, ic] [...], irel, iz
```

Initialization

ia -- starting value. Zero is illegal for exponentials.

ib, *ic*, etc. -- value after *dur1* seconds, etc. For exponentials, must be non-zero and must agree in sign with *ia*.

idur1 -- duration in seconds of first segment. A zero or negative value will cause all initialization to be skipped.

idur2, *idur3*, etc. -- duration in seconds of subsequent segments. A zero or negative value will terminate the initialization process with the preceding point, permitting the last-defined line or curve to be continued indefinitely in performance. The default is zero.

irel, *iz* -- duration in seconds and final value of a note releasing segment.

Performance

These units generate control or audio signals whose values can pass through 2 or more specified points. The sum of *dur* values may or may not equal the instrument's performance time: a shorter performance will truncate the specified pattern, while a longer one will cause the last-defined segment to continue on in the same direction.

expsegr is amongst the Csound “r” units that contain a note-off sensor and release time extender. When each senses an event termination or MIDI noteoff, it immediately extends the performance time of the current instrument by *irel* seconds, and sets out to reach the value *iz* by the end of that period (no matter which segment the unit is in). “r” units can also be modified by MIDI noteoff velocities. For two or more extenders in an instrument, extension is by the greatest period.

You can use other pre-made envelopes which start a release segment upon receiving a note off message, like *linsegr* and *madsr*, or you can construct more complex envelopes using *xtratim* and *release*. Note that you don't need to use *xtratim* if you are using *expsegr*, since the time is extended automatically.

Examples

Here is an example of the expsegr opcode. It uses the file *expsegr.csd* [examples/expsegr.csd].

Example 146. Example of the expsegr opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o expsegr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; p4 = frequency in pitch-class notation.
kcps = cpspch(p4)

; Use an amplitude envelope with second-long release.
kenv expsegr 0.01, p3/2, 1, p3/2, 0.01, 1, 1
kamp = kenv * 30000

al oscil kamp, kcps, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Make sure the score lasts for four seconds.
f 0 4

; p4 = frequency (in pitch-class notation).
; Play Instrument #1 for a half-second, p4=8.00
i 1 0 0.5 8.00
; Play Instrument #1 for a half-second, p4=8.01
i 1 1 0.5 8.01
; Play Instrument #1 for a half-second, p4=8.02
i 1 2 0.5 8.02
; Play Instrument #1 for a half-second, p4=8.03
i 1 3 0.5 8.03
e

</CsScore>
</CsoundSynthesizer>
```

See Also

linsegr, expsegr, envlpxr, mxadsr, madsr expon, expseg, expsega, xtratim

Credits

Author: Barry L. Vercoe

Example written by Kevin Conder.

New in Csound 3.47

fclose

fclose — Closes a previously opened file.

Description

fclose can be used to close a file which was opened with *fiopen*.

Syntax

```
fclose ihandle
```

```
fclose Sfilename
```

Initialization

ihandle -- a number which identifies this file (generated by a previous *fiopen*).

Sfilename -- A string in double quotes or string variable with the filename. The full path must be given if the file directory is not in the system PATH and is not present in the current directory.

Performance

fclose closes a file which was previously opened with *fiopen*. *fclose* is only needed if you need to read a file written to during the same csound performance, since only when csound ends a performance does it close and save data in all open files. The opcode *fclose* is useful for instance if you want to save presets within files which you want to be accesible without having to terminate csound.



Note

If you don't need this functionality it is safer not to call *fclose*, and just let csound close the files when it exits.

If a files closed with *fclose* is being accessed by another opcode (like *fout* or *foutk*, it will be closed later when it is no longer being used.



Warning

This opcode should be used with care, as the file handle will become invalid, and will cause an init error when an opcode tries to access the closed file.

See Also

fout, *fout*, *fouti*, *foutir*, *foutk*

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 5.02

filelen

filelen — Returns the length of a sound file.

Description

Returns the length of a sound file.

Syntax

```
ir filelen ifilcod, [iallowraw]
```

Initialization

ifilcod -- sound file to be queried

iallowraw -- Allow raw sound files (default=1)

Performance

filelen returns the length of the sound file *ifilcod* in seconds. *filelen* can return the length of convolve and PVOC files if the "allow raw sound file" flag is not zero (it is non-zero by default).

Examples

Here is an example of the filelen opcode. It uses the file *filelen.csd* [examples/filelen.csd], and *mary.wav* [examples/mary.wav].

Example 147. Example of the filelen opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o filelen.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the length of the audio file
; "mary.wav" in seconds.
ilen filelen "mary.wav"
print ilen
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 1 second.
i 1 0 1
```

e

```
</CsScore>  
</CsoundSynthesizer>
```

The audio file “mary.wav” is 3.5 seconds long. So *filelen*'s output should include a line like this:

```
instr 1:  ilen = 3.501
```

See Also

filenchnls, *filepeak*, *filesr*

Credits

Author: Matt Ingalls
July 1999

Example written by Kevin Conder.

New in Csound version 3.57

filenchnls

filenchnls — Returns the number of channels in a sound file.

Description

Returns the number of channels in a sound file.

Syntax

```
ir filenchnls ifilcod [, iallowraw]
```

Initialization

ifilcod -- sound file to be queried

iallowraw -- (Optional) Allow raw sound files (default=1)

Performance

filenchnls returns the number of channels in the sound file *ifilcod*. *filechnls* can return the number of channels of convolve and PVOC files if the *iallowraw* flag is not zero (it is non-zero by default).

Examples

Here is an example of the *filenchnls* opcode. It uses the file *filenchnls.csd* [examples/filenchnls.csd], and *mary.wav* [examples/mary.wav].

Example 148. Example of the *filenchnls* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o filenchnls.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the number of channels in the
; audio file "mary.wav".
ichnls filenchnls "mary.wav"
print ichnls
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 1 second.
i 1 0 1
```

e

```
</CsScore>  
</CsoundSynthesizer>
```

The audio file “mary.wav” is monoaural (1 channel). So *filenchnls*'s output should include a line like this:

```
instr 1:  ichnls = 1.000
```

See Also

filelen, *filepeak*, *filesr*

Credits

Author: Matt Ingalls
July 1999

Example written by Kevin Conder.

New in Csound version 3.57

filepeak

filepeak — Returns the peak absolute value of a sound file.

Description

Returns the peak absolute value of a sound file.

Syntax

```
ir filepeak ifilcod [, ichnl]
```

Initialization

ifilcod -- sound file to be queried

ichnl (optional, default=0) -- channel to be used in calculating the peak value. Default is 0.

- *ichnl* = 0 returns peak value of all channels
- *ichnl* > 0 returns peak value of *ichnl*

Performance

filepeak returns the peak absolute value of the sound file *ifilcod*.

Examples

Here is an example of the filepeak opcode. It uses the file *filepeak.csd* [examples/filepeak.csd], and *mary.wav* [examples/mary.wav].

Example 149. Example of the filepeak opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o filepeak.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the peak absolute value of the
; audio file "mary.wav".
ipeak filepeak "mary.wav"
print ipeak
endin
```

```
</CsInstruments>
<CsScore>

; Play Instrument #1 for 1 second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

The peak absolute value of the audio file “mary.wav” is 0.306902. So *filepeak*'s output should include a line like this:

```
instr 1:  ipeak = 0.307
```

See Also

filelen, *filenchnls*, *filesr*

Credits

Author: Matt Ingalls
July 1999

Example written by Kevin Conder.

New in Csound version 3.57

filesr

filesr — Returns the sample rate of a sound file.

Description

Returns the sample rate of a sound file.

Syntax

```
ir filesr ifilcod [, iallowraw]
```

Initialization

ifilcod -- sound file to be queried

iallowraw -- (Optional) Allow raw sound files (default=1)

Performance

filesr returns the sample rate of the sound file *ifilcod*. *filesr* can return the sample rate of convolve and PVOC files if the *iallowraw* flag is not zero (it is non-zero by default).

Examples

Here is an example of the *filesr* opcode. It uses the file *filesr.csd* [examples/filesr.csd], and *mary.wav* [examples/mary.wav].

Example 150. Example of the *filesr* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o filesr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the sampling rate of the
; audio file "mary.wav".
isr filesr "mary.wav"
print isr
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 1 second.
i 1 0 1
```


e

```
</CsScore>  
</CsoundSynthesizer>
```

The audio file “mary.wav” was sampled at 44.1 KHz. So *filesr*'s output should include a line like this:

```
instr 1:  isr = 44100.000
```

See Also

filelen, *filenchnls*, *filepeak*

Credits

Author: Matt Ingalls
July 1999

Example written by Kevin Conder.

New in Csound version 3.57

filter2

filter2 — Performs filtering using a transposed form-II digital filter lattice with no time-varying control.

Description

General purpose custom filter with time-varying pole control. The filter coefficients implement the following difference equation:

$$(1)*y(n) = b0*x[n] + b1*x[n-1] + \dots + bM*x[n-M] - a1*y[n-1] - \dots - aN*y[n-N]$$

the system function for which is represented by:

$$H(Z) = \frac{B(Z)}{A(Z)} = \frac{b0 + b1*Z^{-1} + \dots + bM*Z^{-M}}{1 + a1*Z^{-1} + \dots + aN*Z^{-N}}$$

Syntax

```
ares filter2 asig, iM, iN, ib0, ib1, ..., ibM, ia1, ia2, ..., iaN
```

```
kres filter2 ksig, iM, iN, ib0, ib1, ..., ibM, ia1, ia2, ..., iaN
```

Initialization

At initialization the number of zeros and poles of the filter are specified along with the corresponding zero and pole coefficients. The coefficients must be obtained by an external filter-design application such as Matlab and specified directly or loaded into a table via *GEN01*.

Performance

The *filter2* opcodes perform filtering using a transposed form-II digital filter lattice with no time-varying control.

Since *filter2* implements generalized recursive filters, it can be used to specify a large range of general DSP algorithms. For example, a digital waveguide can be implemented for musical instrument modeling using a pair of *delayr* and *delayw* opcodes in conjunction with the *filter2* opcode.

Examples

A first-order linear-phase lowpass linear-phase FIR filter operating on a k-rate signal:

```
k1 filter2 ksig, 2, 0, 0.5, 0.5    ;; k-rate FIR filter
```

See Also

zfilter2

Credits

Author: Michael A. Casey
M.I.T.
Cambridge, Mass.
1997

New in version 3.47

fin

fin — Read signals from a file at a-rate.

Description

Read signals from a file at a-rate.

Syntax

```
fin ifilename, iskipframes, iformat, ain1 [, ain2] [, ain3] [...]
```

Initialization

ifilename -- input file name (can be a string or a handle number generated by *fiopen*)

iskipframes -- number of frames to skip at the start (every frame contains a sample of each channel)

iformat -- a number specifying the input file format for headerless files. If a header is found, this argument is ignored.

- 0 - 32 bit floating points without header
- 1 - 16 bit integers without header

Performance

fin (file input) is the complement of *fout*: it reads a multichannel file to generate audio rate signals. The user must be sure that the number of channels of the input file is the same as the number of *ainX* arguments.



Note

Please note that since this opcode generates its output using input parameters (on the right side of the opcode), these variables must be initialized before use, otherwise a 'used before defined' error will occur. You can use the *init* opcode for this.

See Also

fini, *fink*

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

fini

fini — Read signals from a file at i-rate.

Description

Read signals from a file at i-rate.

Syntax

```
fini ifilename, iskipframes, iformat, in1 [, in2] [, in3] [, ...]
```

Initialization

ifilename -- input file name (can be a string or a handle number generated by *flopen*)

iskipframes -- number of frames to skip at the start (every frame contains a sample of each channel)

iformat -- a number specifying the input file format. If a header is found, this argument is ignored.

- 0 - floating points in text format (loop; see below)
- 1 - floating points in text format (no loop; see below)
- 2 - 32 bit floating points in binary format (no loop)

Performance

fini is the complement of *fouti* and *foutir*. It reads the values each time the corresponding instrument note is activated. When *iformat* is set to 0 and the end of file is reached, the file pointer is zeroed. This restarts the scan from the beginning. When *iformat* is set to 1 or 2, no looping is enabled and at the end of file the corresponding variables will be filled with zeroes.



Note

Please note that since this opcode generates its output using input parameters (on the right side of the opcode), these variables must be initialized before use, otherwise a 'used before defined' error will occur. You can use the *init* opcode for this.

See Also

fin, *fink*

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

fink

fink — Read signals from a file at k-rate.

Description

Read signals from a file at k-rate.

Syntax

```
fink ifilename, iskipframes, iformat, kin1 [, kin2] [, kin3] [...]
```

Initialization

ifilename -- input file name (can be a string or a handle number generated by fiopen)

iskipframes -- number of frames to skip at the start (every frame contains a sample of each channel)

iformat -- a number specifying the input file format. If a header is found, this argument is ignored.

- 0 - 32 bit floating points without header
- 1 - 16 bit integers without header

Performance

fink is the same as *fin* but operates at k-rate.



Note

Please note that since this opcode generates its output using input parameters (on the right side of the opcode), these variables must be initialized before use, otherwise a 'used before defined' error will occur. You can use the *init* opcode for this.

See Also

fin, *fini*

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

fiopen

fiopen — Opens a file in a specific mode.

Description

fiopen can be used to open a file in one of the specified modes.

Syntax

```
ihandle fiopen ifilename, imode
```

Initialization

ihandle -- a number which specifies this file.

ifilename -- the output file's name (in double-quotes).

imode -- choose the mode of opening the file. *imode* can be a value chosen among the following:

- 0 - open a text file for writing
- 1 - open a text file for reading
- 2 - open a binary file for writing
- 3 - open a binary file for reading

Performance

fiopen opens a file to be used by the *fout* family of opcodes. It is safer to use it in the header section, external to any instruments. It returns a number, *ihandle*, which unequivocally refers to the opened file.

If *fiopen* is called on an already open file, it just returns the same handle again, and does not close the file.

Notice that *fout* and *foutk* can use either a string containing a file pathname, or a handle-number generated by *fiopen*. Whereas, with *fouti* and *foutir*, the target file can be only specified by means of a handle-number.

See Also

fclose fout, fouti, foutir, foutk

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

flanger

flanger — A user controlled flanger.

Description

A user controlled flanger.

Syntax

```
ares flanger asig, adel, kfeedback [, imaxd]
```

Initialization

imaxd(optional) -- maximum delay in seconds (needed for initial memory allocation)

Performance

asig -- input signal

adel -- delay in seconds

kfeedback -- feedback amount (in normal tasks this should not exceed 1, even if bigger values are allowed)

This unit is useful for generating choruses and flangers. The delay must be varied at a-rate connecting *adel* to an oscillator output. Also the feedback can vary at k-rate. This opcode is implemented to allow *kr* different than *sr* (else delay could not be lower than *ksmps*) enhancing realtime performance. This unit is very similar to *wguide1*, the only difference is *flanger* does not have the lowpass filter.

Examples

Here is an example of the flanger opcode. It uses the file *flanger.csd* [examples/flanger.csd], and *beats.wav* [examples/beats.wav].

Example 151. Example of the flanger opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o flanger.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use the "beat.wav" audio file.
```



```
asig soundin "beats.wav"

; Vary the delay amount from 0 to 0.01 seconds.
adel line 0, p3, 0.01
kfeedback = 0.7

; Apply flange to the input signal.
aflang flanger asig, adel, kfeedback

; It can get loud, so clip its amplitude to 30,000.
al clip aflang, 1, 30000
out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Gabriel Maldonado
Italy

Example written by Kevin Conder.

New in Csound version 3.49

flashtxt

flashtxt — Allows text to be displayed from instruments like sliders

Description

Allows text to be displayed from instruments like sliders etc. (only on Unix and Windows at present)

Syntax

```
flashtxt iwhich, String
```

Initialization

iwhich -- the number of the window.

String -- the string to be displayed.

Performance

A window is created, identified by the *iwhich* argument, with the text string displayed. If the text is replaced by a number then the window id deleted. Note that the text windows are globally numbered so different instruments can change the text, and the window survives the instance of the instrument.

Examples

Here is an example of the flashtxt opcode. It uses the file *flashtxt.csd* [examples/flashtxt.csd].

Example 152. Example of the flashtxt opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc       -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o flashtxt.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

instr 1
  flashtxt 1, "Instr 1 live"
  ao oscil 4000, 440, 1
  out ao
endin

</CsInstruments>
<CsScore>

; Table 1: an ordinary sine wave.
f 1 0 32768 10 1
```

```
; Play Instrument #1 for three seconds.  
i 1 0 3  
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

FLbox

FLbox — A FLTK widget that displays text inside of a box.

Description

A FLTK widget that displays text inside of a box.

Syntax

```
ihandle FLbox "label", itype, ifont, isize, iwidth, iheight, ix, iy [, image]
```

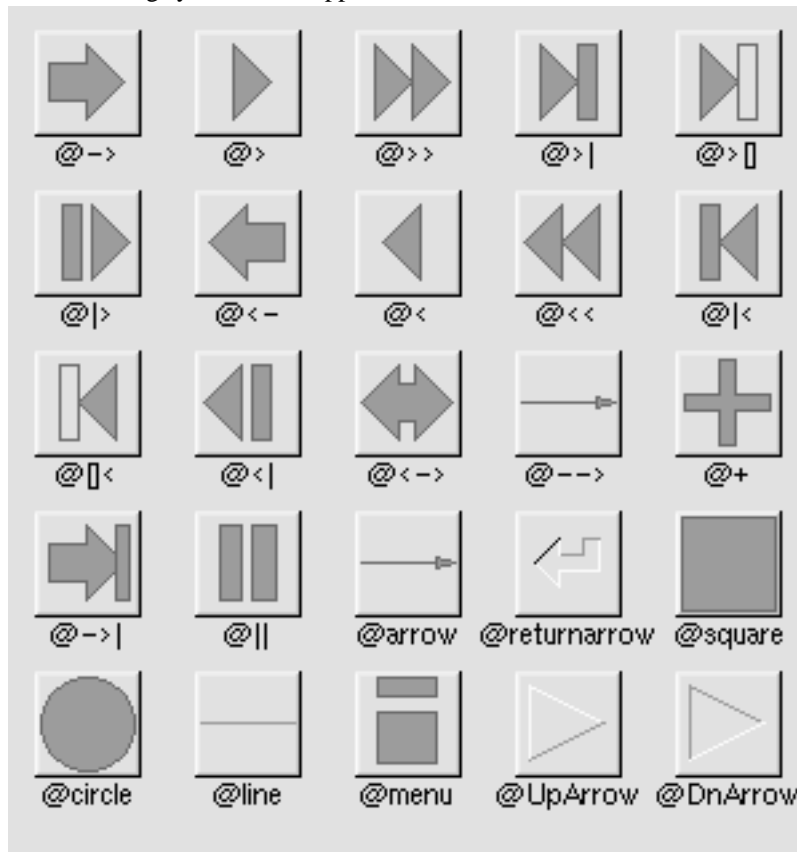
Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLbox* and must not be set by the user label. (The user label is a double-quoted string containing some user-provided text placed near the widget.)

"label" -- a double-quoted string containing some user-provided text, placed near corresponding widget.

Notice that with *FLbox*, it is not necessary to call the *FLsetTextType* opcode at all in order to use a symbol. In this case, it is sufficient to set a label starting with "@" followed by the proper formatting string.

The following symbols are supported:



FLTK label supported symbols.

The @ sign may be followed by the following optional “formatting” characters, in this order:

1. “#” forces square scaling rather than distortion to the widget's shape.
2. +[1-9] or -[1-9] tweaks the scaling a little bigger or smaller.
3. [1-9] rotates by a multiple of 45 degrees. “6” does nothing, the others point in the direction of that key on a numeric keypad.

itype -- an integer number denoting the appearance of the widget.

The following values are legal for *itype*:

- 1 - flat box
- 2 - up box
- 3 - down box
- 4 - thin up box
- 5 - thin down box
- 6 - engraved box
- 7 - embossed box
- 8 - border box
- 9 - shadow box
- 10 - rounded box
- 11 - rounded box with shadow
- 12 - rounded flat box
- 13 - rounded up box
- 14 - rounded down box
- 15 - diamond up box
- 16 - diamond down box
- 17 - oval box
- 18 - oval shadow box
- 19 - oval flat box

ifont -- an integer number denoting the font of *FLbox*.

ifont argument to set the font type. The following values are legal for *ifont*:

- 1 - helvetica (same as "Arial" under Windows)
- 2 - helvetica bold
- 3 - helvetica italic

- 4 - helvetica bold italic
- 5 - courier
- 6 - courier bold
- 7 - courier italic
- 8 - courier bold italic
- 9 - times
- 10 - times bold
- 11 - times italic
- 12 - times bold italic
- 13 - symbol
- 14 - screen
- 15 - screen bold
- 16 - dingbats

isize -- size of the font.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of the upper left corner of the valuator, relative to the upper left corner of corresponding window. (Expressed in pixels.)

iy -- vertical position of the upper left corner of the valuator, relative to the upper left corner of corresponding window. (Expressed in pixels.)

image -- a handle referring to an eventual image opened with *bmopen* opcode. If it is set, it allows a skin for that widget.



Note about the *bmopen* opcode

Although the documentation mentions the *bmopen* opcode, it has not been implemented in Csound 4.22.

Performance

FLbox is useful to show some text in a window. The text is bounded by a box, whose aspect depends on *itype* argument.

Note that *FLbox* is not a valuator and its value is fixed. Its value cannot be modified.

Examples

Here is an example of the *FLbox* opcode. It uses the file *FLbox.csd* [examples/FLbox.csd].

Example 153. Example of the *FLbox* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o FLbox.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "Text Box", 700, 400, 50, 50
; Box border type (7=embossed box)
itype = 7
; Font type (10='Times Bold')
ifont = 10
; Font size
isize = 20
; Width of the flbox
iwidth = 400
; Height of the flbox
iheight = 30
; Distance of the left edge of the flbox
; from the left edge of the panel
ix = 150
; Distance of the upper edge of the flbox
; from the upper edge of the panel
iy = 100

ih3 FLbox "Use Text Boxes For Labelling", itype, ifont, isize, iwidth, iheight, ix, iy
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
endin

</CsInstruments>
<CsScore>

; Real-time performance for 1 hour.
f 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLbutBank, *FLbutton*, *FLprintk*, *FLprintk2*, *FLvalue*

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLbutBank

FLbutBank — A FLTK widget opcode that creates a bank of buttons.

Description

A FLTK widget opcode that creates a bank of buttons.

Syntax

```
kout, ihandle FLbutBank itype, inumx, inumy, iwidth, iheight, ix, iy, \  
iopcode [, kp1] [, kp2] [, kp3] [, kp4] [, kp5] [....] [, kpN]
```

Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLbutBank* and must not be set by the user label. (The user label is a double-quoted string containing some user-provided text placed near the widget.)

itype -- an integer number denoting the appearance of the widget. The valid numbers are:

- 1 - normal button
- 2 - light button
- 3 - check button
- 4 - round button

You can add 20 to the value to create a "plastic" type button. (Note that there is no Plastic Round button. i.e. if you set type to 24 it will look exactly like type 23).

inumx -- number of buttons in each row of the bank.

inumy -- number of buttons in each column of the bank

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window, expressed in pixels

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window, expressed in pixels

iopcode -- score opcode type. You have to provide the ascii code of the letter corresponding to the score opcode. At present time only "i" (ascii code 105) score statements are supported. A zero value refers to a default value of "i". So both 0 and 105 activates the *i* opcode. A value of -1 disables this opcode feature.

Performance

kout -- output value

kp1, kp2, ..., kpN -- arguments of the activated instruments.

The *FLbutBank* opcode creates a bank of buttons. For example, the following line:


```
gkButton,ihbl FLbutBank 22, 8, 8, 380, 180, 50, 350, 0, 7, 0, 0, 5000, 6000
```

will create the this bank:



FLbutBank.

A click to a button checks that button. It may also uncheck a previous checked button belonging to the same bank. So the behaviour is always that of radio-buttons. Notice that each button is labeled with a progressive number. The *kout* argument is filled with that number when corresponding button is checked.

FLbutBank not only outputs a value but can also activate (or schedule) an instrument provided by the user each time a button is pressed. If the *iopcode* argument is set to a negative number, no instrument is activated so this feature is optional. In order to activate an instrument, *iopcode* must be set to 0 or to 105 (the ascii code of character "i", referring to the *i* score opcode). P-fields of the activated instrument are *kp1* (instrument number), *kp2* (action time), *kp3* (duration) and so on with user p-fields.

The *itype* argument sets the type of buttons identically to the *FLbutton* opcode. By adding 10 to the *itype* argument (i.e. by setting 11 for type 1, 12 for type 2, 13 for type 3 and 14 for type 4), it is possible to skip the current *FLbutBank* value when getting/setting snapshots (see *General FLTK Widget-related Opcodes*). You can also add 10 to "plastic" button types (31 for type 1, 32 for type 2, etc.)

FLbutBank is very useful to retrieve snapshots.

Examples

Here is an example of the FLbutBank opcode. It uses the file *FLbutBank.csd* [examples/FLbutBank.csd].

Example 154. Example of the FLbutBank opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLbutton.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
nchnls = 1

FLpanel "Button Bank", 520, 140, 100, 100
;itype = 2      ;Light Buttons
```

```
itype = 22      ;Plastic Light Buttons
inumx = 10
inumpy = 4
iwidth = 500
iheight = 120
ix = 10
iy = 10
iopcode = 0
istarttim = 0
idur = 1

gkbutton, ihbb FLbutBank itype, inumx, inumpy, iwidth, iheight, ix, iy, iopcode, 1, istarttim, i

FLpanelEnd
FLrun

instr 1
    ibutton = i(gkbutton)
    prints "Button %i pushed!\\n", ibutton
endin

</CsInstruments>
<CsScore>

; Real-time performance for 1 hour.
f 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLbox, FLbutton, FLprintk, FLprintk2, FLvalue

Credits

Author: Gabriel Maldonado

New in version 4.22

FLbutton

FLbutton — A FLTK widget opcode that creates a button.

Description

A FLTK widget opcode that creates a button.

Syntax

```
kout, ihandle FLbutton "label", ion, ioff, itype, iwidth, iheight, ix, \
      iy, iopcode [, kp1] [, kp2] [, kp3] [, kp4] [, kp5] [....] [, kpN]
```

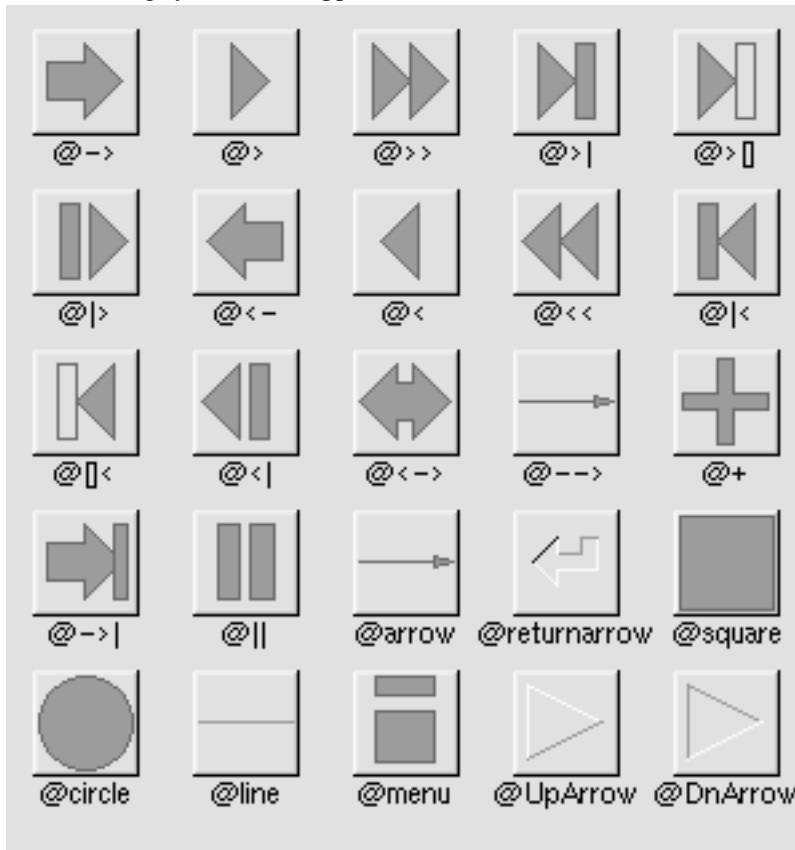
Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLbutton* and must not be set by the user label. (The user label is a double-quoted string containing some user-provided text placed near the widget.)

"label" -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

Notice that with *FLbutton*, it is not necessary to call the *FLsetTextType* opcode at all in order to use a symbol. In this case, it is sufficient to set a label starting with "@" followed by the proper formatting string.

The following symbols are supported:



FLTK label supported symbols.

The @ sign may be followed by the following optional “formatting” characters, in this order:

1. “#” forces square scaling rather than distortion to the widget's shape.
2. +[1-9] or -[1-9] tweaks the scaling a little bigger or smaller.
3. [1-9] rotates by a multiple of 45 degrees. “6” does nothing, the others point in the direction of that key on a numeric keypad.

ion -- value output when the button is checked.

ioff -- value output when the button is unchecked.

itype -- an integer number denoting the appearance of the widget.

Several kind of buttons are possible, according to the value of *itype* argument:

- 1 - normal button
- 2 - light button
- 3 - check button
- 4 - round button

You can add 20 to the value to create a "plastic" type button. (Note that there is no Plastic Round button. i.e. if you set type to 24 it will look exactly like type 23).

This is the appearance of the buttons:



FLbutton.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iopcode -- score opcode type. You have to provide the ascii code of the letter corresponding to the score opcode. At present time only "i" (ascii code 105) score statements are supported. A zero value refers to a default value of "i". So both 0 and 105 activates the *i* opcode. A value of -1 disables this opcode feature.

Performance

kout -- output value

kp1, *kp2*, ..., *kpN* -- arguments of the activated instruments.

Buttons of type 2, 3, and 4 also output (*kout* argument) the value contained in the *ion* argument when checked, and that contained in *ioff* argument when unchecked.

By adding 10 to *itype* argument (i.e. by setting 11 for type 1, 12 for type 2, 13 for type 3 and 14 for type 4) it is possible to skip the button value when getting/setting snapshots (see later section). *FLbutton* not only outputs a value, but can also activate (or schedule) an instrument provided by the user each time a button is pressed. You can also add 10 to "plastic" button types (31 for type 1, 32 for type 2, etc.)

If the *iopcode* argument is set to a negative number, no instrument is activated. So this feature is optional. In order to activate an instrument, *iopcode* must be set to 0 or to 105 (the ascii code of character "i", referring to the *i* score opcode).

P-fields of the activated instrument are *kp1* (instrument number), *kp2* (action time), *kp3* (duration) and so on with user p-fields. Notice that in dual state buttons (light button, check button and round button), the instrument is activated only when button state changes from unchecked to checked (not when passing from checked to unchecked).

Examples

Here is an example of the *FLbutton* opcode. It uses the file *FLbutton.csd* [examples/FLbutton.csd], and *beats.wav* [examples/beats.wav].

Example 155. Example of the *FLbutton* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o FLbutton.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Using FLbuttons to create on screen controls for play,
; stop, fast forward and fast rewind of a sound file
; This example also makes use of a preset graphic for buttons.

sr = 44100
kr = 44100
ksmps = 1
nchnls = 2

FLpanel "Buttons", 240, 400, 100, 100
```

```
ion = 0
ioff = 0
itype = 1
iwidth = 50
iheight = 50
ix = 10
iy = 10
iopcode = 0
istarttim = 0
idur = -1 ;Turn instruments on indefinitely

; Normal speed forwards
gkplay, ihb1 FLbutton "@>", ion, ioff, itype, iwidth, iheight, ix, iy, iopcode, 1, istarttim, i
; Stationary
gkstop, ihb2 FLbutton "@square", ion, ioff, itype, iwidth, iheight, ix+55, iy, iopcode, 2, istarttim, i
; Double speed backwards
gkrew, ihb3 FLbutton "@<<", ion, ioff, itype, iwidth, iheight, ix + 110, iy, iopcode, 1, istarttim, i
; Double speed forward
gkff, ihb4 FLbutton "@>>", ion, ioff, itype, iwidth, iheight, ix+165, iy, iopcode, 1, istarttim, i
; Type 1
gkt1, iht1 FLbutton "1-Normal Button", ion, ioff, 1, 200, 40, ix, iy + 65, -1
; Type 2
gkt2, iht2 FLbutton "2-Light Button", ion, ioff, 2, 200, 40, ix, iy + 110, -1
; Type 3
gkt3, iht3 FLbutton "3-Check Button", ion, ioff, 3, 200, 40, ix, iy + 155, -1
; Type 4
gkt4, iht4 FLbutton "4-Round Button", ion, ioff, 4, 200, 40, ix, iy + 200, -1
; Type 21
gkt5, iht5 FLbutton "21-Plastic Button", ion, ioff, 21, 200, 40, ix, iy + 245, -1
; Type 22
gkt6, iht6 FLbutton "22-Plastic Light Button", ion, ioff, 22, 200, 40, ix, iy + 290, -1
; Type 23
gkt7, iht7 FLbutton "23-Plastic Check Button", ion, ioff, 23, 200, 40, ix, iy + 335, -1
FLpanelEnd
FLrun

; Ensure that only 1 instance of instr 1
; plays even if the play button is clicked repeatedly
insnum = 1
icount = 1
maxalloc insnum, icount

instr 1
  asig diskin "beats.wav", p4, 0, 1
  outs asig, asig
endin

instr 2
  turnoff2 1, 0, 0 ;Turn off instr 1
  turnoff ;Turn off this instrument
endin

</CsInstruments>
<CsScore>

; Real-time performance for 1 hour.
f 0 3600
e

</CsScore>
</CsSoundSynthesizer>
```

See Also

FLbox, FLbutBank, FLprintk, FLprintk2, FLvalue

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLcloseButton

FLcloseButton — A FLTK widget opcode that creates a button that will close the panel window it is a part of.

Description

A FLTK widget opcode that creates a button that will close the panel window it is a part of.

Syntax

```
ihandle FLcloseButton "label", iwidth, iheight, ix, iy
```

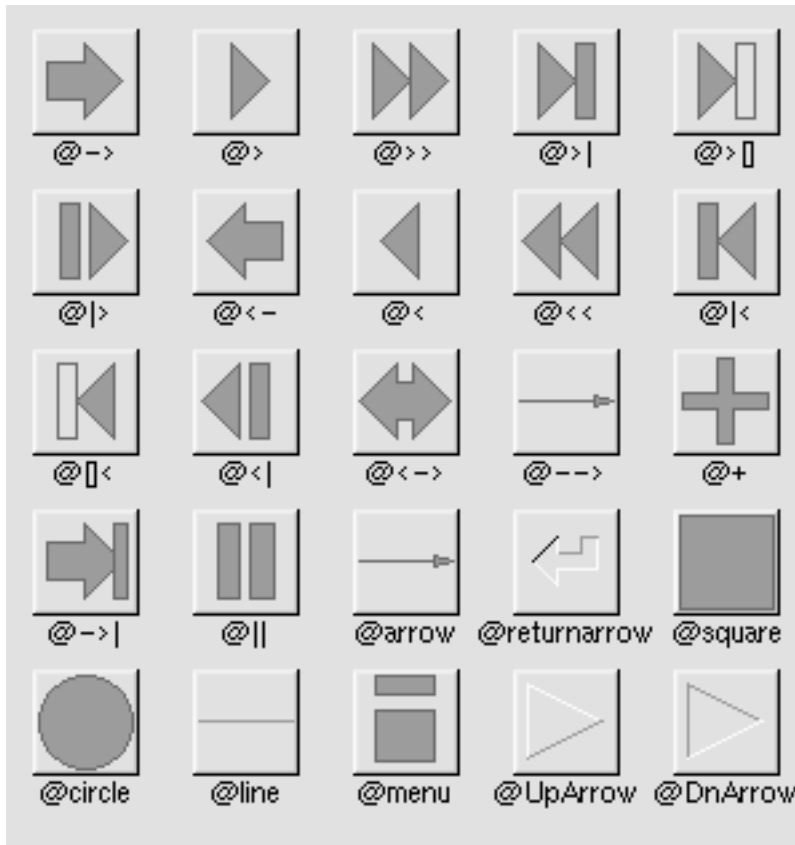
Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLcloseButton* and must not be set by the user label. (The user label is a double-quoted string containing some user-provided text placed near the widget.)

"label" -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

Notice that with *FLcloseButton*, it is not necessary to call the *FLsetTextType* opcode at all in order to use a symbol. In this case, it is sufficient to set a label starting with “@” followed by the proper formatting string.

The following symbols are supported:



FLTK label supported symbols.

The @ sign may be followed by the following optional “formatting” characters, in this order:

1. “#” forces square scaling rather than distortion to the widget's shape.
2. +[1-9] or -[1-9] tweaks the scaling a little bigger or smaller.
3. [1-9] rotates by a multiple of 45 degrees. “6” does nothing, the others point in the direction of that key on a numeric keypad.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

See Also

FLbutton, *FLbox*, *FLbutBank*, *FLprintk*, *FLprintk2*, *FLvalue*

Credits

Author: Steven Yi

New in version 5.05

FLcolor

FLcolor — A FLTK opcode that sets the primary colors.

Description

Sets the primary colors to RGB values given by the user.

Syntax

```
FLcolor ired, igrreen, iblue [, ired2, igrreen2, iblue2]
```

Initialization

ired -- The red color of the target widget. The range for each RGB component is 0-255

igrreen -- The green color of the target widget. The range for each RGB component is 0-255

iblue -- The blue color of the target widget. The range for each RGB component is 0-255

ired2 -- The red component for the secondary color of the target widget. The range for each RGB component is 0-255

igrreen2 -- The green component for the secondary color of the target widget. The range for each RGB component is 0-255

iblue2 -- The blue component for the secondary color of the target widget. The range for each RGB component is 0-255

Performance

These opcodes modify the appearance of other widgets. There are two types of such opcodes, those that don't contain the *ihandle* argument which affect all subsequently declared widgets, and those without *ihandle* which affect only a target widget previously defined.

FLcolor sets the primary colors to RGB values given by the user. This opcode affects the primary color of (almost) all widgets defined next its location. User can put several instances of *FLcolor* in front of each widget he intend to modify. However, to modify a single widget, it would be better to use the opcode belonging to the second type (i.e. those containing *ihandle* argument).

FLcolor is designed to modify the colors of a group of related widgets that assume the same color. The influence of *FLcolor* on subsequent widgets can be turned off by using -1 as the only argument of the opcode. Also, using -2 (or -3) as the only value of *FLcolor* makes all next widget colors randomly selected. The difference is that -2 selects a light random color, while -3 selects a dark random color.

Using *ired2*, *igrreen2*, *iblue2* is equivalent to using a separate *FLcolor2*.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLcolor2

FLcolor2 — A FLTK opcode that sets the secondary (selection) color.

Description

FLcolor2 is the same of *FLcolor* except it affects the secondary (selection) color.

Syntax

```
FLcolor2 ired, igreen, iblue
```

Initialization

ired -- The red color of the target widget. The range for each RGB component is 0-255

igreen -- The green color of the target widget. The range for each RGB component is 0-255

iblue -- The blue color of the target widget. The range for each RGB component is 0-255

Performance

These opcodes modify the appearance of other widgets. There are two types of such opcodes: those that don't contain the *ihandle* argument which affect all subsequently declared widgets, and those without *ihandle* which affect only a target widget previously defined.

FLcolor2 is the same of *FLcolor* except it affects the secondary (selection) color. Setting it to -1 turns off the influence of *FLcolor2* on subsequent widgets. A value of -2 (or -3) makes all next widget secondary colors randomly selected. The difference is that -2 selects a light random color, while -3 selects a dark random color.

See Also

FLcolor, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLcount

FLcount — A FLTK widget opcode that creates a counter.

Description

Allows the user to increase/decrease a value with mouse clicks on a corresponding arrow button.

Syntax

```
kout, ihandle FLcount "label", imin, imax, istep1, istep2, itype, \  
    iwidth, iheight, ix, iy, iopcode [, kp1] [, kp2] [, kp3] [...] [, kpN]
```

Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. Used by further opcodes that changes some valuator's properties. It is automatically set by the corresponding valuator.

"label" -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

imin -- minimum value of output range

imax -- maximum value of output range

istep1 -- a floating-point number indicating the increment of valuator value corresponding to of each mouse click. *istep1* is for fine adjustments.

istep2 -- a floating-point number indicating the increment of valuator value corresponding to of each mouse click. *istep2* is for coarse adjustments.

itype -- an integer number denoting the appearance of the valuator.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

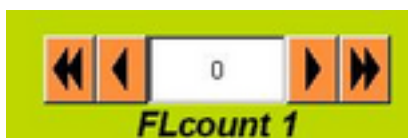
iopcode -- score opcode type. You have to provide the ascii code of the letter corresponding to the score opcode. At present time only "i" (ascii code 105) score statements are supported. A zero value refers to a default value of "i". So both 0 and 105 activates the *i* opcode. A value of -1 disables this opcode feature.

Performance

kout -- output value

kp1, kp2, ..., kpN -- arguments of the activated instruments.

FLcount allows the user to increase/decrease a value with mouse clicks on corresponding arrow buttons:



FLcount.

There are two kind of arrow buttons, for larger and smaller steps. Notice that *FLcount* not only outputs a value and a handle, but can also activate (schedule) an instrument provided by the user each time a button is pressed. P-fields of the activated instrument are *kp1* (instrument number), *kp2* (action time), *kp3* (duration) and so on with user p-fields. If the *iopcode* argument is set to a negative number, no instrument is activated. So this feature is optional.

Examples

Here is an example of the FLcount opcode. It uses the file *FLcount.csd* [examples/FLcount.csd].

Example 156. Example of the FLcount opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLcount.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Demonstration of the flcount opcode
; clicking on the single arrow buttons
; increments the oscillator in semitone steps
; clicking on the double arrow buttons
; increments the oscillator in octave steps
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "Counter", 900, 400, 50, 50
; Minimum value output by counter
imin = 6
; Maximum value output by counter
imax = 12
; Single arrow step size (semitones)
istep1 = 1/12
; Double arrow step size (octave)
istep2 = 1
; Counter type (1=double arrow counter)
itype = 1
; Width of the counter in pixels
iwidth = 200
; Height of the counter in pixels
iheight = 30
; Distance of the left edge of the counter
; from the left edge of the panel
ix = 50
; Distance of the top edge of the counter
; from the top edge of the panel
iy = 50
; Score event type (-1=ignored)
iopcode = -1

gkoct, ihandle FLcount "pitch in oct format", imin, imax, istep1, istep2, itype, iwidth, iheight
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
iamp = 15000
```

```
    ifn = 1
    asig oscili iamp, cpsoct(gkoc), ifn
    out asig
endin

</CsInstruments>
<CsScore>

; Function table that defines a single cycle
; of a sine wave.
f 1 0 1024 10 1

; Instrument 1 will play a note for 1 hour.
i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLjoy, FLknob, FLroller, FLslider, FLtext

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLexecButton

FLexecButton — A FLTK widget opcode that creates a button that executes a command.

Description

A FLTK widget opcode that creates a button that executes a command. Useful for opening up HTML documentation as About text or to start a separate program from an FLTK widget interface.



Warning

Because any command can be executed, the user is advised to be very careful when using this opcode and when running orchestras by others using this opcode.

Syntax

```
ihandle FLexecButton "command", iwidth, iheight, ix, iy
```

Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLexecButton*.

"command" -- a double-quoted string containing a command to execute.

Notice that with *FLexecButton*, the default text for the button is "About" and it is necessary to call the *FLsetText* opcode to change the text of the button.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

Examples

Here is an example of the FLexecButton opcode. It uses the file *FLexecButton.csd* [examples/FLexecButton.csd].

Example 157. Example of the FLexecButton opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No display
-odac        -iadc     -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>
```

```
sr      = 44100
ksmps   = 10
nchnls  = 1

; Example by Jonathan Murphy 2007

;;; reset amplitude range
0dbfs   = 1

;;; set the base colour for the panel
FLcolor 100, 0, 200
;;; define the panel
FLpanel "FLEXecButton", 250, 100, 0, 0
;;; sliders to control time stretch and pitch
gkstr, gistretch FLslider "Time", 0.5, 1.5, 0, 6, -1, 10, 60, 150, 20
gkpch, gipitch  FLslider "Pitch", 0.5, 1.5, 0, 6, -1, 10, 60, 200, 20
;;; set FLEXecButton colour
FLcolor 255, 255, 0
;;; when this button is pressed, fourier analysis is performed on the file
;;; "beats.wav", producing the analysis file "beats.pvx"
gipvoc FLEXecButton "csound -U pvanal beats.wav beats.pvx", 60, 20, 20, 20
;;; set FLEXecButton text
FLsetText "PVOC", gipvoc
;;; when this button is pressed, instr 10000 is called, exiting
;;; Csound immediately

;;; cancel previous colour
FLcolor -1
;;; set colour for kill button
FLcolor 255, 0, 0
gkill, gikill FLbutton "X", 1, 1, 1, 20, 20, 100, 20, 0, 10000, 0, 0.1
;;; cancel previous colour
FLcolor -1
;;; set colour for play/stop and pause buttons
FLcolor 0, 200, 0
;;; pause and play/stop buttons
gkpause, gipause FLbutton "@|", 1, 0, 2, 40, 20, 20, 60, -1
gkplay, gipplay  FLbutton "@>", 1, 0, 2, 40, 20, 80, 60, -1
;;; end the panel
FLpanelEnd
;;; set initial values for time stretch and pitch
FLsetVal_i 1, gistretch
FLsetVal_i 1, gipitch
;;; run the panel
FLrun

instr 1                                     ; trigger play/stop
;;; is the play/stop button on or off?
;;; either way we need to trigger something,
;;; so we can't just use the value of gkplay
kon trigger gkplay, 0, 0
koff trigger gkplay, 1, 1
;;; if on, start instr 2
schedkwhen kon, -1, -1, 2, 0, -1
;;; if off, stop instr 2
schedkwhen koff, -1, -1, -2, 0, -1

endin

instr 2

;;; paused or playing?
if (gkpause == 1) kgoto pause
kgoto start

pause:
;;; if the pause button is on, skip sound production
kgoto end

start:
;;; get the length of the analysis file in seconds
ilen filelen "beats.pvx"
;;; determine base frequency of playback
icps = 1/ilen
;;; create a table over the length of the file
itpt ftgen 0, 0, 513, -7, 0, 512, ilen
;;; phasor for time control
kphs phasor icps * gkstr
;;; use phasor as index into table
kndx = kphs * 512
;;; read table
ktpt tablei kndx, itpt
;;; use value from table as time pointer into file
fsigl pvsfread ktpt, "beats.pvx"
;;; change playback pitch
fsig2 pvscale fsigl, gkpch
;;; resynthesize
aout pvsynth fsig2
```



```
;;; envelope to avoid clicks and clipping
aenv    linsegr 0, 0.3, 0.75, 0.1, 0
aout    = aout * aenv
        out      aout
end:

    endin

    instr 10000                                ; kill

        exitnow

    endin

</CsInstruments>
<CsScore>
i1 0 10000
e
</CsScore>
</CsoundSynthesizer>
```

See Also

FLbutton, FLbox, FLbutBank, FLprintk, FLprintk2, FLvalue

Credits

Author: Steven Yi

Example by: Jonathan Murphy

New in version 5.05

FLgetsnap

FLgetsnap — Retrieves a previously stored FLTK snapshot.

Description

Retrieves a previously stored snapshot (in memory), i.e. sets all valuator to the corresponding values stored in that snapshot.

Syntax

```
inumsnap FLgetsnap index [, igroup]
```

Initialization

inumsnap -- current number of snapshots.

index -- a number referring unequivocally to a snapshot. Several snapshots can be stored in the same bank.

igroup -- (optional) an integer number referring to a snapshot-related group of widget. It allows to get/set, or to load/save the state of a subset of valuators. Default value is zero that refers to the first group. The group number is determined by the opcode *FLsetSnapGroup*.



Note

The *igroup* parameter has not been yet fully implemented in the current version of csound. Please do not rely on it yet.

Performance

FLgetsnap retrieves a previously stored snapshot (in memory), i.e. sets all valuator to the corresponding values stored in that snapshot. The *index* argument unequivocally must refer to an already existing snapshot. If the *index* argument refers to an empty snapshot or to a snapshot that doesn't exist, no action is done. *FLsetsnap* outputs the current number of snapshots (*inumsnap* argument).

For purposes of snapshot saving, widgets can be grouped, so that snapshots affect only a defined group of widgets. The opcode *FLsetSnapGroup* is used to specify the group for all widgets declared after it, until the next *FLsetSnapGroup* statement.

See Also

FLloadsnap, *FLrun*, *FLsavesnap*, *FLsetsnap*, *FLsetSnapGroup*, *FLupdate*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLgroup

FLgroup — A FLTK container opcode that groups child widgets.

Description

A FLTK container opcode that groups child widgets.

Syntax

```
FLgroup "label", iwidth, iheight, ix, iy [, iborder] [, image]
```

Initialization

"label" -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iborder (optional, default=0) -- border type of the container. It is expressed by means of an integer number chosen from the following:

- 0 - no border
- 1 - down box border
- 2 - up box border
- 3 - engraved border
- 4 - embossed border
- 5 - black line border
- 6 - thin down border
- 7 - thin up border

If the integer number doesn't match any of the previous values, no border is provided as the default.

image (optional) -- a handle referring to an eventual image opened with the *bmopen* opcode. If it is set, it allows a skin for that widget.



Note about the *bmopen* opcode

Although the documentation mentions the *bmopen* opcode, it has not been implemented in Csound 4.22.

Performance

Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuator or other kinds of widgets.

There are no k-rate arguments in containers.

See Also

FLgroupEnd, *FLpack*, *FLpackEnd*, *FLpanel*, *FLpanelEnd*, *FLscroll*, *FLscrollEnd*, *FLtabs*, *FLtabsEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLgroupEnd

FLgroupEnd — Marks the end of a group of FLTK child widgets.

Description

Marks the end of a group of FLTK child widgets.

Syntax

`FLgroupEnd`

Performance

Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuator or other kinds of widgets.

There are no k-rate arguments in containers.

See Also

FLgroup, *FLpack*, *FLpackEnd*, *FLpanel*, *FLpanelEnd*, *FLscroll*, *FLscrollEnd*, *FLtabs*, *FLtabsEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLgroupEnd

FLgroup_end — Marks the end of a group of FLTK child widgets.

Description

Marks the end of a group of FLTK child widgets. This is another name for **FLgroupEnd** provides for compatibility. See *FLgroupEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLhide

FLhide — Hides the target FLTK widget.

Description

Hides the target FLTK widget, making it invisible.

Syntax

```
FLhide ihandle
```

Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLbutBank* and must not be set by the user label. (The user label is a double-quoted string containing some user-provided text placed near the widget.)

Performance

FLhide hides target widget, making it invisible.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLhvsBox

FLhvsBox — Displays a box with a grid useful for visualizing two-dimensional Hyper Vectorial Synthesis.

Description

FLhvsBox displays a box with a grid useful for visualizing two-dimensional Hyper Vectorial Synthesis.

Syntax

```
ihandle FLhvsBox inumlinesX, inumlinesY, iwidth, iheight, ix, iy [, image]
```

Initialization

ihandle – an integer number used a univocally-defined handle for identifying a specific HVS box (see below).

inumlinesX, inumlinesY - number of vertical and horizontal lines delimiting the HVS squared areas

iwidth, iheight - width and height of the HVS box

ix, iy - the position of the HVS box

image – (optional, default 0) an integer number denoting an RGB image opened with the *bmopen* opcode. A zero indicates no image.

Performance

FLhvsBox is a widget able to visualize current position of the HVS cursor in an HVS box (i.e. a squared area containing a grid). The number of horizontal and vertical lines of the grid can be defined with the *inumlinesX, inumlinesY* arguments. This opcode has to be declared inside an *FLpanel* - *FLpanelEnd* block. See the entry for *hvs2* for an example of usage of *FLhvsBox*.

FLhvsBoxSetValue is used to set the cursor position of an *FLhvsBox* widget.



Note

The opcode *bmscan* has not been implemented, so currently the parameter *image* has no effect.

See Also

hvs2, FLhvsBoxSetValue

Credits

Author: Gabriel Maldonado

New in version 5.06

FLhvsBoxSetValue

FLhvsBoxSetValue — Sets the cursor position of a previously-declared FLhvsBox widget.

Description

FLhvsBoxSetValue sets the cursor position of a previously-declared *FLhvsBox* widget.

Syntax

FLhvsBox *kx*, *ky*, *ihandle*

Initialization

ihandle – an integer number used a univocally-defined handle for identifying a specific HVS box (see below).

Performance

kx, *ky*– the coordinates of the HVS cursor position to be set.

FLhvsBoxSetValue sets the cursor position of a previously-declared *FLhvsBox* widget. The *kx* and *ky* arguments, denoting the cursor position, have to be expressed in normalized values (0 to 1 range).

See the entry for *hvs2* for an example of usage of *FLhvsBoxSetValue*.

See Also

hvs2, *FLhvsBox*

Credits

Author: Gabriel Maldonado

New in version 5.06

FLjoy

FLjoy — A FLTK opcode that acts like a joystick.

Description

FLjoy is a squared area that allows the user to modify two output values at the same time. It acts like a joystick.

Syntax

```
koutx, kouty, ihandlex, ihandley FLjoy "label", iminx, imaxx, iminy, \  
imaxy, iexpx, iexpy, idispx, idispy, iwidth, iheight, ix, iy
```

Initialization

ihandlex -- a handle value (an integer number) that unequivocally references a corresponding widget. Used by further opcodes that changes some valuator's properties. It is automatically set by the corresponding valuator.

ihandley -- a handle value (an integer number) that unequivocally references a corresponding widget. Used by further opcodes that changes some valuator's properties. It is automatically set by the corresponding valuator.

"label" -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

iminx -- minimum x value of output range

imaxx -- maximum x value of output range

iminy -- minimum y value of output range

imaxy -- maximum y value of output range

iwidth -- width of widget.

idispx -- a handle value that was output from a previous instance of the *FLvalue* opcode to display the current value of the current valuator in the *FLvalue* widget itself. If the user doesn't want to use this feature that displays current values, it must be set to a negative number by the user.

idispy -- a handle value that was output from a previous instance of the *FLvalue* opcode to display the current value of the current valuator in the *FLvalue* widget itself. If the user doesn't want to use this feature that displays current values, it must be set to a negative number by the user.

iexpx -- an integer number denoting the behaviour of valuator:

- 0 = valuator output is linear
- -1 = valuator output is exponential

All other positive numbers for *iexpx* indicate the number of an existing table that is used for indexing. Linear interpolation is provided in table indexing. A negative table number suppresses interpolation.

iexpy -- an integer number denoting the behaviour of valuator:

- 0 = valuator output is linear
- -1 = valuator output is exponential

All other positive numbers for *iexpy* indicate the number of an existing table that is used for indexing. Linear interpolation is provided in table indexing. A negative table number suppresses interpolation.



IMPORTANT!

Notice that the tables used by valuator must be created with the *ftgen* opcode and placed in the orchestra before the corresponding valuator. They can not placed in the score. In fact, tables placed in the score are created later than the initialization of the opcodes placed in the header section of the orchestra.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

Performance

koutx -- x output value

kouty -- y output value

Examples

Here is an example of the FLjoy opcode. It uses the file *FLjoy.csd* [examples/FLjoy.csd].

Example 158. Example of the FLjoy opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o FLjoy.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Demonstration of the flpanel opcode
; Horizontal click-dragging controls the frequency of the oscillator
; Vertical click-dragging controls the amplitude of the oscillator
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "X Y Panel", 900, 400, 50, 50
; Minimum value output by x movement (frequency)
iminx = 200
; Maximum value output by x movement (frequency)
imaxx = 5000
; Minimum value output by y movement (amplitude)
iminy = 0
; Maximum value output by y movement (amplitude)
imaxy = 15000
```

```
; Logarithmic change in x direction
iexpx = -1
; Linear change in y direction
iexpy = 0
; Display handle x direction (-1=not used)
idispx = -1
; Display handle y direction (-1=not used)
idispy = -1
; Width of the x y panel in pixels
iwidth = 800
; Height of the x y panel in pixels
iheight = 300
; Distance of the left edge of the x y panel from
; the left edge of the panel
ix = 50
; Distance of the top edge of the x y
; panel from the top edge of the panel
iy = 50

gkfreqx, gkampy, ihandlex, ihandley FLjoy "X - Frequency Y - Amplitude", iminx, imaxx, iminy, i
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
    ifn = 1
    asig oscili gkampy, gkfreqx, ifn
    out asig
endin

</CsInstruments>
<CsScore>

; Function table that defines a single cycle
; of a sine wave.
f 1 0 1024 10 1

; Instrument 1 will play a note for 1 hour.
i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLcount, FLknob, FLroller, FLslider, FLtext

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLkeyIn

FLkeyIn — Reports keys pressed (on alphanumeric keyboard) when an FLTK panel has focus.

Description

FLkeyIn informs about the status of a key pressed by the user on the alphanumeric keyboard when an FLTK panel has got the focus.

Syntax

```
kascii FLkeyIn [ifn]
```

Initialization

ifn – (optional, default value is zero) set the behavior of FLkeyIn (see below).

Performance

kascii - the ascii value of last pressed key. If the key is pressed, the value is positive, when the key is released the value is negative.

FLkeyIn is useful to know whether a key has been pressed on the computer keyboard. The behavior of this opcode depends on the optional *ifn* argument.

If *ifn* = 0 (default), *FLkeyIn* outputs the ascii code of the last pressed key. If it is a special key (ctrl, shift, alt, f1-f12 etc.), a value of 256 is added to the output value in order to distinguish it from normal keys. The output will continue to output the last key value, until a new key is pressed or released. Notice that the output will be negative when a key is depressed.

If *ifn* is set to the number of an already-allocated table having at least 512 elements, then the table element having index equal to the ascii code of the key pressed is set to 1, all other table elements are set to 0. This allows to check the state of a certain key or set of keys.

Be aware that you must set the *ikbdcapture* parameter to something other than 0 on a designated *FLpanel* for *FLkeyIn* to capture keyboard events from that panel.



Note

FLkeyIn works internally at k-rate, so it can't be used in the header as other FLTK opcodes. It must be used inside an instrument.

Examples

Here is an example of the FLkeyIn opcode. It uses the file *FLkeyIn.csd* [examples/FLkeyIn.csd].

Example 159. Example of the FLkeyIn opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
```

```
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

;Example by Andres Cabrera 2007

FLpanel "FLkeyIn", 400, 300, -1, -1, 5, 1, 1
FLpanelEnd

FLrun

Odbfs = 1

instr 1
kascii   FLkeyIn
ktrig changed kascii
if (kascii > 0) then
  printf "Key Down: %i\n", ktrig, kascii
else
  printf "Key Up: %i\n", ktrig, -kascii
endif
endin

</CsInstruments>
<CsScore>
i 1 0 120
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Gabriel Maldonado

New in version 5.06

FLknob

FLknob — A FLTK widget opcode that creates a knob.

Description

A FLTK widget opcode that creates a knob.

Syntax

```
kout, ihandle FLknob "label", imin, imax, iexp, itype, idisp, iwidth, \  
ix, iy [, icursorsize]
```

Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLknob* and must not be set by the user label. (The user label is a double-quoted string containing some user-provided text placed near the widget.)

"label" -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

imin -- minimum value of output range.

imax -- maximum value of output range.

iexp -- an integer number denoting the behaviour of valuator:

- 0 = valuator output is linear
- -1 = valuator output is exponential

All other positive numbers for *iexp* indicate the number of an existing table that is used for indexing. Linear interpolation is provided in table indexing. A negative table number suppresses interpolation.



IMPORTANT!

Notice that the tables used by valuator must be created with the *ftgen* opcode and placed in the orchestra before the corresponding valuator. They can not placed in the score. In fact, tables placed in the score are created later than the initialization of the opcodes placed in the header section of the orchestra.

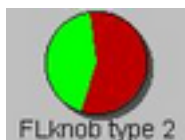
itype -- an integer number denoting the appearance of the valuator.

The *itype* argument can be set to the following values:

- 1 - a 3-D knob
- 2 - a pie-like knob
- 3 - a clock-like knob
- 4 - a flat knob



A 3-D knob.



A pie knob.



A clock knob.



A flat knob.

idisp -- a handle value that was output from a previous instance of the *FLvalue* opcode to display the current value of the current valuator in the *FLvalue* widget itself. If the user doesn't want to use this feature that displays current values, it must be set to a negative number by the user.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

icursorsize (optional) -- If *FLknob's* *itype* is set to 1 (3D knob), this parameter controls the size of knob cursor.

Performance

kout -- output value

FLknob puts a knob in the corresponding container.

Examples

Here is an example of the *FLknob* opcode. It uses the file *FLknob.csd* [examples/FLknob.csd].

Example 160. Example of the *FLknob* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

<CsoundSynthesizer>


```
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLknob.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; A sine with oscillator with flknob controlled frequency
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "Frequency Knob", 900, 400, 50, 50
; Minimum value output by the knob
imin = 200
; Maximum value output by the knob
imax = 5000
; Logarithmic type knob selected
iexp = -1
; Knob graphic type (1=3D knob)
itype = 1
; Display handle (-1=not used)
idisp = -1
; Width of the knob in pixels
iwidth = 70
; Distance of the left edge of the knob
; from the left edge of the panel
ix = 70
; Distance of the top edge of the knob
; from the top of the panel
iy = 125

gkfreq, ihandle FLknob "Frequency", imin, imax, iexp, itype, idisp, iwidth, ix, iy
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

; Set the widget's initial value
FLsetVal_i 300, ihandle

instr 1
iamp = 15000
ifn = 1
asig oscili iamp, gkfreq, ifn
out asig
endin

</CsInstruments>
<CsScore>

; Function table that defines a single cycle
; of a sine wave.
f 1 0 1024 10 1

; Instrument 1 will play a note for 1 hour.
i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the FLknob opcode, showing the different styles of knobs and the usage of FLvalue to display a knob's value. It uses the file *FLknob-2.csd* [examples/FLknob-2.csd].

Example 161. More complex example of the FLknob opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLknob.wav -W ;; for file output any platform
</CsOptions>
```

```
<CsInstruments>

sr = 44100
kr = 441
ksmps = 100
nchnls = 1

;By Andres Cabrera 2007
FLpanel "Knob Types", 330, 230, 50, 50
; Distance of the left edge of the knob
; from the left edge of the panel
ix = 20
; Distance of the top edge of the knob
; from the top of the panel
iy = 20

;Create boxes that display a widget's value
ihandleA FLvalue "A", 60, 20, ix + 130, iy + 110
ihandleB FLvalue "B", 60, 20, ix + 220, iy + 110
ihandleC FLvalue "C", 60, 20, ix + 130, iy + 160
ihandleD FLvalue "D", 60, 20, ix + 220, iy + 160

; The four types of FLknobs
gkdummy1, ihandle1 FLknob "Type 1", 200, 5000, -1, 1, ihandleA, 70, ix, iy, 90
gkdummy2, ihandle2 FLknob "Type 2", 200, 5000, -1, 2, ihandleB, 70, ix + 100, iy
gkdummy3, ihandle3 FLknob "Type 3", 200, 5000, -1, 3, ihandleC, 70, ix + 200, iy
gkdummy4, ihandle4 FLknob "Type 4", 200, 5000, -1, 4, ihandleD, 70, ix, iy + 100
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

; Set the color of widgets
FLsetColor 20, 23, 100, ihandle1
FLsetColor 0, 123, 100, ihandle2
FLsetColor 180, 23, 12, ihandle3
FLsetColor 10, 230, 0, ihandle4

FLsetColor2 200, 230, 0, ihandle1
FLsetColor2 200, 0, 123, ihandle2
FLsetColor2 180, 180, 100, ihandle3
FLsetColor2 180, 23, 12, ihandle4

; Set the initial value of the widget
FLsetVal_i 300, ihandle1
FLsetVal_i 1000, ihandle2

instr 1
; Nothing here for now
endin

</CsInstruments>
<CsScore>

f 0 3600 ;Dumy table to make csound wait for realtime events

e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLcount, FLjoy, FLroller, FLslider, FLtext

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLlabel

FLlabel — A FLTK opcode that modifies the appearance of a text label.

Description

Modifies a set of parameters related to the text label appearance of a widget (i.e. size, font, alignment and color of corresponding text).

Syntax

```
FLlabel isize, ifont, ialign, ired, igreen, iblue
```

Initialization

isize -- size of the font of the target widget. Normal values are in the order of 15. Greater numbers enlarge font size, while smaller numbers reduce it.

ifont -- sets the the font type of the label of a widget.

Legal values for ifont argument are:

- 1 - Helvetica (same as Arial under Windows)
- 2 - Helvetica Bold
- 3 - Helvetica Italic
- 4 - Helvetica Bold Italic
- 5 - Courier
- 6 - Courier Bold
- 7 - Courier Italic
- 8 - Courier Bold Italic
- 9 - Times
- 10 - Times Bold
- 11 - Times Italic
- 12 - Times Bold Italic
- 13 - Symbol
- 14 - Screen
- 15 - Screen Bold
- 16 - Dingbats

ialign -- sets the alignment of the label text of the widget.

Legal values for ialign argument are:

- 1 - align center
- 2 - align top
- 3 - align bottom
- 4 - align left
- 5 - align right
- 6 - align top-left
- 7 - align top-right
- 8 - align bottom-left
- 9 - align bottom-right

ired -- The red color of the target widget. The range for each RGB component is 0-255

igreen -- The green color of the target widget. The range for each RGB component is 0-255

ibblue -- The blue color of the target widget. The range for each RGB component is 0-255

Performance

FLlabel modifies a set of parameters related to the text label appearance of a widget, i.e. size, font, alignment and color of corresponding text. This opcode affects (almost) all widgets defined next its location. A user can put several instances of *FLlabel* in front of each widget he intends to modify. However, to modify a particular widget, it is better to use the opcode belonging to the second type (i.e. those containing the *ihandle* argument).

The influence of *FLlabel* on the next widget can be turned off by using -1 as its only argument. *FLlabel* is designed to modify text attributes of a group of related widgets.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLloadsnap

FLloadsnap — Loads all snapshots into the memory bank of the current orchestra.

Description

FLloadsnap loads all the snapshots contained in a file into the memory bank of the current orchestra.

Syntax

```
FLloadsnap "filename" [, igroup]
```

Initialization

"filename" -- a double-quoted string corresponding to a file to load a bank of snapshots.

igroup -- (optional) an integer number referring to a snapshot-related group of widget. It allows to get/set, or to load/save the state of a subset of valuator. Default value is zero that refers to the first group. The group number is determined by the opcode *FLsetSnapGroup*.



Note

The *igroup* parameter has not been yet fully implemented in the current version of csound. Please do not rely on it yet.

Performance

FLloadsnap loads all snapshots contained in filename into the memory bank of current orchestra.

For purposes of snapshot saving, widgets can be grouped, so that snapshots affect only a defined group of widgets. The opcode *FLsetSnapGroup* is used to specify the group for all widgets declared after it, until the next *FLsetSnapGroup* statement.

See Also

FLgetsnap, *FLrun*, *FLsetSnapGroup*, *FLsavesnap*, *FLsetsnap*, *FLupdate*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLmouse

FLmouse — Returns the mouse position and the state of the three mouse buttons.

Description

FLmouse returns the coordinates of the mouse position within an FLTK panel and the state of the three mouse buttons.

Syntax

```
kx, ky, kb1, kb2, kb3 FLmouse [, imode]
```

Initialization

imode – (optional, default = 0) Determines the mode for mouse location reporting.

- 0 - Absolute position normalized to range 0-1
- 1 - Absolute raw pixel position
- 2 - Raw pixel position, relative to FLTK panel

Performance

kx, ky – the mouse coordinates, whose range depends on the *iflag* argument (see above).

kb1, kb2, kb3 – the states of the mouse buttons, 1 when corresponding button is pressed, 0 when the button is not pressed.

FLmouse returns the coordinates of the mouse position and the state of the three mouse buttons. The coordinates can be retrieved in three modes depending on the *imode* argument value (see above). Modes 0 and 1 report mouse position in relation to the complete screen (Absolute mode), while mode 2, reports the pixel position within an FLTK panel. Notice that FLmouse is only active when the mouse cursor passes on an *FLpanel* area.

Examples

Here is an example of the FLmouse opcode. It uses the file *FLmouse.csd* [examples/FLmouse.csd].

Example 162. Example of the FLmouse opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2
```

```
;Example by Andres Cabrera 2007
giwidth = 400
giheight = 300
FLpanel "FLmouse", giwidth, giheight, 10, 10
FLpanelEnd

FLrun

Odbfs = 1

instr 1
  kx, ky, kb1, kb2, kb3      FLmouse 2
  ktrig changed kx, ky ;Print only if coordinates have changed
  printf "kx = %f   ky = %f \n", ktrig, kx, ky
  kfreq = ((giwidth - ky)*1000/giwidth) + 300

  ; y coordinate determines frequency, x coordinate determines amplitude
  ; Left mouse button (kb1) doubles the frequency
  ; Right mouse button (kb3) activates sound on channel 2
  aout oscil kx /giwidth , kfreq * (kb1 + 1), 1
  outs aout, aout * kb3
endin

</CsInstruments>
<CsScore>
f 1 0 1024 10 1

i 1 0 120
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Gabriel Maldonado

New in version 5.06

flooper

flooper — Function-table-based crossfading looper.

Description

This opcode reads audio from a function table and plays it back in a loop with user-defined start time, duration and crossfade time. It also allows the pitch of the loop to be controlled, including reversed playback. It accepts non-power-of-two tables, such as deferred-allocation GEN01 tables.

Syntax

```
asig flooper kamp, kpitch, istart, idur, ifad, ifn
```

Initialisation

istart -- loop start pos in seconds

idur -- loop duration in seconds

ifad -- crossfade duration in seconds

ifn -- function table number, generally created using GEN01

Performance

asig -- output sig

kon -- amplitude control

kpitch -- pitch control (transposition ratio); negative values play the loop back in reverse

Examples

Example 163. Example

```
aout flooper 16000, 1, 1, 4, 0.05, 1 ; loop starts at 1 sec, for 4 secs 0.05 crossfade
out aout
```

The example above shows the basic operation of flooper. Pitch can be controlled at the k-rate, as well as amplitude. The example assumes table 1 to contain at least 5.05 seconds of audio (4 secs loop duration, starting 1 sec into the table, using 0.05 secs after the loop end for the crossfade).

Credits

Author: Victor Lazzarini;
April 2005

New plugin in version 5

April 2005.

flooper2

flooper2 — Function-table-based crossfading looper.

Description

This opcode implements a crossfading looper with variable loop parameters and three looping modes, optionally using a table for its crossfade shape. It accepts non-power-of-two tables for its source sounds, such as deferred-allocation GEN01 tables.

Syntax

```
asig flooper2 kamp, kpitch, kloopstart, kloopend, kcrossfade, ifn \  
    [, istart, imode, ifenv, iskip]
```

Initialisation

ifn -- sound source function table number, generally created using GEN01

istart -- playback start pos in seconds

imode -- loop modes: 0 forward, 1 backward, 2 back-and-forth [def: 0]

ifenv -- if non-zero, crossfade envelope shape table number. The default, 0, sets the crossfade to linear.

iskip -- if 1, the opcode initialisation is skipped, for tied notes, performance continues from the position in the loop where the previous note stopped. The default, 0, does not skip initialisation

Performance

asig -- output sig

kamp -- amplitude control

kpitch -- pitch control (transposition ratio); negative values are not allowed.

kloopstart -- loop start point (secs). Note that although k-rate, loop parameters such as this are only updated once per loop cycle.

kloopend -- loop end point (secs), updated once per loop cycle.

kcrossfade -- crossfade length (secs), updated once per loop cycle and limited to loop length.

Examples

Example 164. Example

```
aout flooper2 16000, 1, 1, 5, 0.05, 1 ; loop starts at 1 sec, for 4 secs 0.05 crossfade  
out aout
```

The example above shows the basic operation of flooper. Pitch can be controlled at the k-rate, as

well as amplitude and loop parameters. The example assumes table 1 to contain at least 5.05 seconds of audio (4 secs loop duration, starting 1 sec into the table, using 0.05 secs after the loop end for the crossfade). Looping is in mode 0 (normal forward loop).

Credits

Author: Victor Lazzarini;
July 2006

New plugin in version 5

July 2006.

floor

floor — Returns the largest integer not greater than x

Description

Returns the largest integer not greater than x

Syntax

floor(x) (init-, control-, or audio-rate arg allowed)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

See Also

abs, exp, int, log, log10, i, sqrt

Credits

Author: Istvan Varga
New in Csound 5
2005

FLpack

FLpack — Provides the functionality of compressing and aligning FLTK widgets.

Description

FLpack provides the functionality of compressing and aligning widgets.

Syntax

FLpack *iwidth, iheight, ix, iy, itype, ispace, iborder*

Initialization

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

itype -- an integer number that modifies the appearance of the target widget.

The *itype* argument expresses the type of packing:

- 0 - vertical
- 1 - horizontal

ispace -- sets the space between the widgets.

iborder -- border type of the container. It is expressed by means of an integer number chosen from the following:

- 0 - no border
- 1 - down box border
- 2 - up box border
- 3 - engraved border
- 4 - embossed border
- 5 - black line border
- 6 - thin down border
- 7 - thin up border

Performance

FLpack provides the functionality of compressing and aligning widgets.

Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuator or other kinds of widgets.

There are no k-rate arguments in containers.

Examples

The following example:

```
FLpanel "Panel1", 450, 300, 100, 100
FLpack 400, 300, 10, 40, 0, 15, 3
gk1, ihs1    FLslider    "FLslider 1", 500, 1000, 2, 1, -1, 300, 15, 20, 50
gk2, ihs2    FLslider    "FLslider 2", 300, 5000, 2, 3, -1, 300, 15, 20, 100
gk3, ihs3    FLslider    "FLslider 3", 350, 1000, 2, 5, -1, 300, 15, 20, 150
gk4, ihs4    FLslider    "FLslider 4", 250, 5000, 1, 11, -1, 300, 30, 20, 200
gk5, ihs5    FLslider    "FLslider 5", 220, 8000, 2, 1, -1, 300, 15, 20, 250
gk6, ihs6    FLslider    "FLslider 6", 1, 5000, 1, 13, -1, 300, 15, 20, 300
gk7, ihs7    FLslider    "FLslider 7", 870, 5000, 1, 15, -1, 300, 30, 20, 350
FLpackEnd
FLpanelEnd
```

...will produce this result, when resizing the window:



FLpack.

See Also

FLgroup, *FLgroupEnd*, *FLpackEnd*, *FLpanel*, *FLpanelEnd*, *FLscroll*, *FLscrollEnd*, *FLtabs*, *FLtabsEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLpackEnd

FLpackEnd — Marks the end of a group of compressed or aligned FLTK widgets.

Description

Marks the end of a group of compressed or aligned FLTK widgets.

Syntax

`FLpackEnd`

Performance

Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuator or other kinds of widgets.

There are no k-rate arguments in containers.

See Also

FLgroup, *FLgroupEnd*, *FLpack*, *FLpanel*, *FLpanelEnd*, *FLscroll*, *FLscrollEnd*, *FLtabs*, *FLtabsEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLpack_end

FLpack_End — Marks the end of a group of compressed or aligned FLTK widgets.

Description

Marks the end of a group of compressed or aligned FLTK widgets. This is another name for **FLpanelEnd** provided for compatibility. See *FLpanel_end*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLpanel

FLpanel — Creates a window that contains FLTK widgets.

Description

Creates a window that contains FLTK widgets.

Syntax

```
FLpanel "label", iwidth, iheight [, ix] [, iy] [, iborder] [, ikbdcapture] [, iclose]
```

Initialization

"label" -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

iwidth -- width of widget.

iheight -- height of widget.

ix (optional) -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy (optional) -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iborder (optional) -- border type of the container. It is expressed by means of an integer number chosen from the following:

- 0 - no border
- 1 - down box border
- 2 - up box border
- 3 - engraved border
- 4 - embossed border
- 5 - black line border
- 6 - thin down border
- 7 - thin up border

ikbdcapture (default = 0) -- If this flag is set to 1, keyboard events are captured by the window (for use with *sensekey* and *FLkeyIn*)

iclose (default = 0) -- If this flag is set to anything other than 0, the close button of the window is disabled, and the window cannot be closed by the user directly. It will close when *csound* exits.

Performance

Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuator or

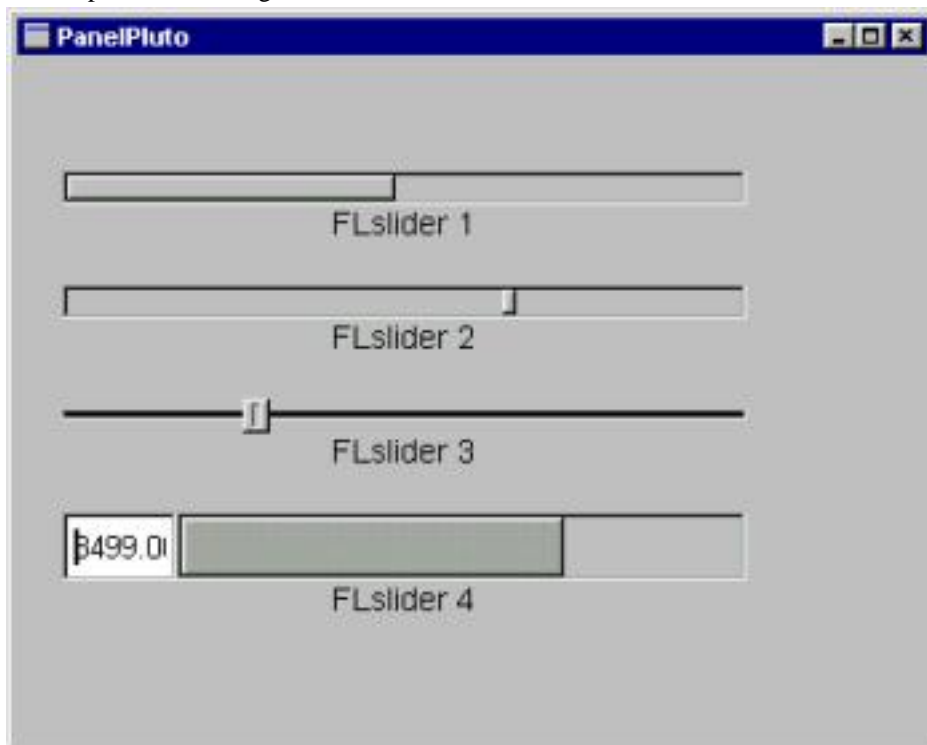
other kinds of widgets.

There are no k-rate arguments in containers.

FLpanel creates a window. It must be followed by the opcode *FLpanelEnd* when all widgets internal to it are declared. For example:

```
FLpanel    "PanelPluto",450,550,100,100 ;***** start of container
gk1,ih1 FLslider  "FLslider 1", 500, 1000, 2 ,1, -1, 300,15, 20,50
gk2,ih2 FLslider  "FLslider 2", 300, 5000, 2 ,3, -1, 300,15, 20,100
gk3,ih3 FLslider  "FLslider 3", 350, 1000, 2 ,5, -1, 300,15, 20,150
gk4,ih4 FLslider  "FLslider 4", 250, 5000, 1 ,11,-1, 300,30, 20,200
FLpanelEnd ;***** end of container
```

will output the following result:



FLpanel.

If the *ikbdcapture* flag is set, the window captures keyboard events, and sends them to all *sensekey*. This flag modifies the behavior of *sensekey*, and makes it receive events from the FLTK window instead of stdin.

Examples

Here is an example of the *FLpanel* opcode. It uses the file *FLpanel.csd* [examples/FLpanel.csd].

Example 165. Example of the *FLpanel* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
```

```
-odac          -iadc      -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o FLpanel.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Creates an empty window panel
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

; Panel height in pixels
ipanelheight = 900
; Panel width in pixels
ipanelwidth = 400
; Horizontal position of the panel on screen in pixels
ix = 50
; Vertical position of the panel on screen in pixels
iy = 50

FLpanel "A Window Panel", ipanelheight, ipanelwidth, ix, iy
; End of panel contents
FLpanelEnd

;Run the widget thread!
FLrun

instr 1
endin

</CsInstruments>
<CsScore>

; 'Dummy' score event of 1 hour.
f 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLgroup, FLgroupEnd, FLpack, FLpackEnd, FLpanelEnd, FLscroll, FLscrollEnd, FLtabs, FLtabsEnd, sensekey

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLpanelEnd

FLpanelEnd — Marks the end of a group of FLTK widgets contained inside of a window (panel).

Description

Marks the end of a group of FLTK widgets contained inside of a window (panel).

Syntax

`FLpanelEnd`

Performance

Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuator or other kinds of widgets.

There are no k-rate arguments in containers.

See Also

FLgroup, *FLgroupEnd*, *FLpack*, *FLpackEnd*, *FLpanel*, *FLscroll*, *FLscrollEnd*, *FLtabs*, *FLtabsEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLpanel_end

FLpanel_end — Marks the end of a group of FLTK widgets contained inside of a window (panel).

Description

Marks the end of a group of FLTK widgets contained inside of a window (panel). This is another name for **FLpanelEnd** provided for compatibility. See *FLpanelEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLprintk

FLprintk — A FLTK opcode that prints a k-rate value at specified intervals.

Description

FLprintk is similar to *printk* but shows values of a k-rate signal in a text field instead of on the console.

Syntax

```
FLprintk itime, kval, idisp
```

Initialization

itime -- how much time in seconds is to elapse between updated displays.

idisp -- a handle value that was output from a previous instance of the *FLvalue* opcode to display the current value of the current valuator in the *FLvalue* widget itself. If the user doesn't want to use this feature that displays current values, it must be set to a negative number by the user.

Performance

kval -- k-rate signal to be displayed.

FLprintk is similar to *printk*, but shows values of a k-rate signal in a text field instead of showing it in the console. The *idisp* argument must be filled with the *ihandle* return value of a previous *FLvalue* opcode. While *FLvalue* should be placed in the header section of an orchestra inside an *FLpanel/FLpanelEnd* block, *FLprintk* must be placed inside an instrument to operate correctly. For this reason, it slows down performance and should be used for debugging purposes only.

See Also

FLbox, *FLbutBank*, *FLbutton*, *FLprintk2*, *FLvalue*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLprintk2

FLprintk2 — A FLTK opcode that prints a new value every time a control-rate variable changes.

Description

FLprintk2 is similar to *FLprintk* but shows a k-rate variable's value only when it changes.

Syntax

```
FLprintk2 kval, idisp
```

Initialization

idisp -- a handle value that was output from a previous instance of the *FLvalue* opcode to display the current value of the current valuator in the *FLvalue* widget itself. If the user doesn't want to use this feature that displays current values, it must be set to a negative number by the user.

Performance

kval -- k-rate signal to be displayed.

FLprintk2 is similar to *FLprintk*, but shows the k-rate variable's value only each time it changes. Useful for monitoring MIDI control changes when using sliders. It should be used for debugging purposes only, since it slows-down performance.

See Also

FLbox, *FLbutBank*, *FLbutton*, *FLprintk*, *FLvalue*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLroller

FLroller — A FLTK widget that creates a transversal knob.

Description

FLroller is a sort of knob, but put transversally.

Syntax

```
kout, ihandle FLroller "label", imin, imax, istep, iexp, itype, idisp, \  
      iwidth, iheight, ix, iy
```

Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLroller* and must not be set by the user label. (The user label is a double-quoted string containing some user-provided text placed near the widget.)

"label" -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

imin -- minimum value of output range.

imax -- maximum value of output range.

istep -- a floating-point number indicating the increment of valuator value corresponding to of each mouse click. The *istep* argument allows the user to arbitrarily slow roller's motion, enabling arbitrary precision.

iexp -- an integer number denoting the behaviour of valuator:

- 0 = valuator output is linear
- -1 = valuator output is exponential

All other positive numbers for *iexp* indicate the number of an existing table that is used for indexing. Linear interpolation is provided in table indexing. A negative table number suppresses interpolation.



IMPORTANT!

Notice that the tables used by valuator must be created with the *ftgen* opcode and placed in the orchestra before the corresponding valuator. They can not placed in the score. In fact, tables placed in the score are created later than the initialization of the opcodes placed in the header section of the orchestra.

itype -- an integer number denoting the appearance of the valuator.

The *itype* argument can be set to the following values:

- 1 - horizontal roller
- 2 - vertical roller

idisp -- a handle value that was output from a previous instance of the *FLvalue* opcode to display the current value of the current valuator in the *FLvalue* widget itself. If the user doesn't want to use this feature that displays current values, it must be set to a negative number by the user.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

Performance

kout -- output value

FLroller is a sort of knob, but put transversally:



FLroller.

Examples

Here is an example of the FLroller opcode. It uses the file *FLroller.csd* [examples/FLroller.csd].

Example 166. Example of the FLroller opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o FLroller.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; A sine with oscillator with flroller controlled frequency
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "Frequency Roller", 900, 400, 50, 50
; Minimum value output by the roller
imin = 200
; Maximum value output by the roller
imax = 5000
; Increment with each pixel
istep = 1
; Logarithmic type roller selected
iexp = -1
; Roller graphic type (1=horizontal)
itype = 1
; Display handle (-1=not used)
idisp = -1
; Width of the roller in pixels
iwidth = 300
; Height of the roller in pixels
iheight = 50
; Distance of the left edge of the knob
; from the left edge of the panel
```

```
ix = 300
; Distance of the top edge of the knob
; from the top edge of the panel
iy = 50

gkfreq, ihandle FLroller "Frequency", imin, imax, istep, iexp, itype, idisp, iwidth, iheight, i
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
  iamp = 15000
  ifn = 1
  asig oscili iamp, gkfreq, ifn
  out asig
endin

</CsInstruments>
<CsScore>

; Function table that defines a single cycle
; of a sine wave.
f 1 0 1024 10 1

; Instrument 1 will play a note for 1 hour.
i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLcount, FLjoy, FLknob, FLslider, FLtext

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLrun

FLrun — Starts the FLTK widget thread.

Description

Starts the FLTK widget thread.

Syntax

`FLrun`

Performance

This opcode must be located at the end of all widget declarations. It has no arguments, and its purpose is to start the thread related to widgets. Widgets would not operate if *FLrun* is missing.

See Also

FLgetsnap, FLloadsnap, FLsavesnap, FLsetsnap, FLupdate

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsavesnap

FLsavesnap — Saves all snapshots currently created into a file.

Description

FLsavesnap saves all snapshots currently created (i.e. the entire memory bank) into a file.

Syntax

```
FLsavesnap "filename" [, igroup]
```

Initialization

"filename" -- a double-quoted string corresponding to a file to store a bank of snapshots.

igroup -- (optional) an integer number referring to a snapshot-related group of widget. It allows to get/set, or to load/save the state of a subset of valuator. Default value is zero that refers to the first group. The group number is determined by the opcode *FLsetSnapGroup*.



Note

The *igroup* parameter has not been yet fully implemented in the current version of csound. Please do not rely on it yet.

Performance

FLsavesnap saves all snapshots currently created (i.e. the entire memory bank) into a file whose name is *filename*. Since the file is a text file, snapshot values can also be edited manually by means of a text editor. The format of the data stored in the file is the following (at present time, this could be changed in next Csound version):

```
----- 0 -----
FLvalue 0 0 1 0 ""
FLvalue 0 0 1 0 ""
FLvalue 0 0 1 0 ""
FLslider 331.946 80 5000 -1 "frequency of the first oscillator"
FLslider 385.923 80 5000 -1 "frequency of the second oscillator"
FLslider 80 80 5000 -1 "frequency of the third oscillator"
FLcount 0 0 10 0 "this index must point to the location number where snapshot is stored"
FLbutton 0 0 1 0 "Store snapshot to current index"
FLbutton 0 0 1 0 "Save snapshot bank to disk"
FLbutton 0 0 1 0 "Load snapshot bank from disk"
FLbox 0 0 1 0 ""
----- 1 -----
FLvalue 0 0 1 0 ""
FLvalue 0 0 1 0 ""
FLvalue 0 0 1 0 ""
FLslider 819.72 80 5000 -1 "frequency of the first oscillator"
FLslider 385.923 80 5000 -1 "frequency of the second oscillator"
FLslider 80 80 5000 -1 "frequency of the third oscillator"
FLcount 1 0 10 0 "this index must point to the location number where snapshot is stored"
FLbutton 0 0 1 0 "Store snapshot to current index"
FLbutton 0 0 1 0 "Save snapshot bank to disk"
FLbutton 0 0 1 0 "Load snapshot bank from disk"
FLbox 0 0 1 0 ""
----- 2 -----
..... etC...
----- 3 -----
..... etc...
-----
```

As you can see, each snapshot contain several lines. Each snapshot is separated from previous and next snapshot by a line of this kind:

```
"----- snapshot Num -----"
```

Then there are several lines containing data. Each of these lines corresponds to a widget.

The first field of each line is an unquoted string containing opcode name corresponding to that widget. Second field is a number that expresses current value of a snapshot. In current version, this is the only field that can be modified manually. The third and fourth fields shows minimum and maximum values allowed for that valuator. The fifth field is a special number that indicates if the valuator is linear (value 0), exponential (value -1), or is indexed by a table interpolating values (negative table numbers) or non-interpolating (positive table numbers). The last field is a quoted string with the label of the widget. Last line of the file is always

```
"-----"
```

.

Note that *FLvalue* and *FLbox* are not valutors and their values are fixed, so they cannot be modified.

For purposes of snapshot saving, widgets can be grouped, so that snapshots affect only a defined group of widgets. The opcode *FLsetSnapGroup* is used to specify the group for all widgets declared after it, until the next *FLsetSnapGroup* statement.

Examples

Here is a simple example of the FLTK snapshot saving. It uses the file *FLsavesnap_simple.csd* [examples/FLsavesnap_simple.csd].

Example 167. Example of FLTK snapshot saving.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr=48000
ksmps=128
nchnls=2

; Example by Hector Centeno and Andres Cabrera 2007

; giSWMtab4 ftgen 0, 0, 513, 21, 10, 1, .3
; giSWMtab4M ftgen 0, 0, 64, 7, 1, 50, 1

FLpanel "Snapshots", 530, 190, 40, 410, 3
  FLcolor 100, 118, 140
  ivalSM1          FLvalue  "", 70, 20, 270, 20
  gksliderA, gislidSM1      FLslider "Slider", -4, 4, 0, 3, ivalSM1, 250, 20, 20, 20
  itext1          FLbox  "store", 1, 1, 14, 50, 25, 355, 15
  itext2          FLbox  "load", 1, 1, 14, 50, 25, 415, 15
  gksnap, ibuttn1  FLbutton "1", 1, 0, 11, 25, 25, 364, 45, 0, 3, 0, 3, 1
  gksnap, ibuttn2  FLbutton "2", 1, 0, 11, 25, 25, 364, 75, 0, 3, 0, 3, 2
  gksnap, ibuttn3  FLbutton "3", 1, 0, 11, 25, 25, 364, 105, 0, 3, 0, 3, 3
  gksnap, ibuttn4  FLbutton "4", 1, 0, 11, 25, 25, 364, 135, 0, 3, 0, 3, 4
```

```
gkload, ibuttn1 FLbutton "1", 1, 0, 11, 25, 25, 424, 45, 0, 4, 0, 3, 1
gkload, ibuttn2 FLbutton "2", 1, 0, 11, 25, 25, 424, 75, 0, 4, 0, 3, 2
gkload, ibuttn3 FLbutton "3", 1, 0, 11, 25, 25, 424, 105, 0, 4, 0, 3, 3
gkload, ibuttn4 FLbutton "4", 1, 0, 11, 25, 25, 424, 135, 0, 4, 0, 3, 4

ivalSM2 FLvalue "", 70, 20, 270, 80
gkknobA, gislidSM2 FLknob "Knob", -4, 4, 0, 3, ivalSM2, 60, 120, 60
FLpanelEnd
FLsetVal_i 1, gislidSM1
FLsetVal_i 1, gislidSM2
FLrun

instr 1

endin

instr 3 ; Save snapshot
index init 0
ipstno = p4
Sfile sprintf "snapshot_simple.%d.snap", ipstno

inumsnap, inumval FLsetsnap index ;, -1, igroup
FLsavesnap Sfile

endin

instr 4 ;Load snapshot
index init 0
ipstno = p4
Sfile sprintf "snapshot_simple.%d.snap", ipstno

FLloadsnap Sfile
inumload FLgetsnap index ;, igroup

endin

</CsInstruments>
<CsScore>
f 0 3600

e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of FLTK snapshot saving using snapshot groups. It uses the file *FLsavesnap.csd* [examples/FLsavesnap.csd].

Example 168. Example of FLTK snapshot saving using snapshot groups.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac -iadc ;;;RT audio I/O
</CsOptions>
<CsInstruments>

sr=48000
ksmps=128
nchnls=2

; Example by Hector Centeno and Andres Cabrera 2007

; giSMMtab4 ftgen 0, 0, 513, 21, 10, 1, .3
; giSMMtab4M ftgen 0, 0, 64, 7, 1, 50, 1

FLpanel "Snapshots", 530, 350, 40, 410, 3
FLcolor 100, 118, 140
FLsetSnapGroup 0
ivalSM1 FLvalue "", 70, 20, 270, 20
ivalSM2 FLvalue "", 70, 20, 270, 60
ivalSM3 FLvalue "", 70, 20, 270, 100
```

```

ivalSM4          FLvalue  "", 70, 20, 270, 140
gksliderA, gislidSM1  FLslider "Slider A", -4, 4, 0, 3, ivalSM1, 250, 20, 20, 20
gksliderB, gislidSM2  FLslider "Slider B", 1, 10, 0, 3, ivalSM2, 250, 20, 20, 60
gksliderC, gislidSM3  FLslider "Slider C", 0, 1, 0, 3, ivalSM3, 250, 20, 20, 100
gksliderD, gislidSM4  FLslider "Slider D", 0, 1, 0, 3, ivalSM4, 250, 20, 20, 140
itext1          FLbox    "store", 1, 1, 14, 50, 25, 355, 15
itext2          FLbox    "load", 1, 1, 14, 50, 25, 415, 15
itext3          FLbox    "G\nr\no\nu\np\n\n\n1", 1, 1, 14, 30, 145, 485, 15
gksnap, ibuttn1  FLbutton "1", 1, 0, 11, 25, 25, 364, 45, 0, 3, 0, 3, 1
gksnap, ibuttn2  FLbutton "2", 1, 0, 11, 25, 25, 364, 75, 0, 3, 0, 3, 2
gksnap, ibuttn3  FLbutton "3", 1, 0, 11, 25, 25, 364, 105, 0, 3, 0, 3, 3
gksnap, ibuttn4  FLbutton "4", 1, 0, 11, 25, 25, 364, 135, 0, 3, 0, 3, 4
gkload, ibuttn1  FLbutton "1", 1, 0, 11, 25, 25, 424, 45, 0, 4, 0, 3, 1
gkload, ibuttn2  FLbutton "2", 1, 0, 11, 25, 25, 424, 75, 0, 4, 0, 3, 2
gkload, ibuttn3  FLbutton "3", 1, 0, 11, 25, 25, 424, 105, 0, 4, 0, 3, 3
gkload, ibuttn4  FLbutton "4", 1, 0, 11, 25, 25, 424, 135, 0, 4, 0, 3, 4

FLcolor 100, 140, 118
FLsetSnapGroup 1
ivalSM5          FLvalue  "", 70, 20, 270, 190
ivalSM6          FLvalue  "", 70, 20, 270, 230
ivalSM7          FLvalue  "", 70, 20, 270, 270
ivalSM8          FLvalue  "", 70, 20, 270, 310
gkknobA, gislidSM5  FLknob  "Knob A", -4, 4, 0, 3, ivalSM5, 45, 10, 230
gkknobB, gislidSM6  FLknob  "Knob B", 1, 10, 0, 3, ivalSM6, 45, 75, 230
gkknobC, gislidSM7  FLknob  "Knob C", 0, 1, 0, 3, ivalSM7, 45, 140, 230
gkknobD, gislidSM8  FLknob  "Knob D", 0, 1, 0, 3, ivalSM8, 45, 205, 230
itext4          FLbox    "store", 1, 1, 14, 50, 25, 355, 185
itext5          FLbox    "load", 1, 1, 14, 50, 25, 415, 185
itext6          FLbox    "G\nr\no\nu\np\n\n\n2", 1, 1, 14, 30, 145, 485, 185
gksnap, ibuttn1  FLbutton "5", 1, 0, 11, 25, 25, 364, 215, 0, 3, 0, 3, 5
gksnap, ibuttn2  FLbutton "6", 1, 0, 11, 25, 25, 364, 245, 0, 3, 0, 3, 6
gksnap, ibuttn3  FLbutton "7", 1, 0, 11, 25, 25, 364, 275, 0, 3, 0, 3, 7
gksnap, ibuttn4  FLbutton "8", 1, 0, 11, 25, 25, 364, 305, 0, 3, 0, 3, 8
gkload, ibuttn1  FLbutton "5", 1, 0, 11, 25, 25, 424, 215, 0, 4, 0, 3, 5
gkload, ibuttn2  FLbutton "6", 1, 0, 11, 25, 25, 424, 245, 0, 4, 0, 3, 6
gkload, ibuttn3  FLbutton "7", 1, 0, 11, 25, 25, 424, 275, 0, 4, 0, 3, 7
gkload, ibuttn4  FLbutton "8", 1, 0, 11, 25, 25, 424, 305, 0, 4, 0, 3, 8

FLpanelEnd
FLsetVal_i 1, gislidSM1
FLsetVal_i 1, gislidSM2
FLsetVal_i 0, gislidSM3
FLsetVal_i 0, gislidSM4
FLsetVal_i 1, gislidSM5
FLsetVal_i 1, gislidSM6
FLsetVal_i 0, gislidSM7
FLsetVal_i 0, gislidSM8
FLrun

instr 1

endin

instr 3 ; Save snapshot
index init 0
ipstno = p4
igroup = 0
Sfile sprintf "PVCsynth.%d.snap", ipstno
if ipstno > 4 then
  igroup = 1
endif

inumsnap, inumval FLsetsnap index, -1, igroup
FLsavesnap Sfile

endin

instr 4 ; Load snapshot
index init 0
ipstno = p4
igroup = 0
Sfile sprintf "PVCsynth.%d.snap", ipstno
if ipstno > 4 then
  igroup = 1
endif

FLloadsnap Sfile
inumload FLgetsnap index, igroup

endin

</CsInstruments>
<CsScore>
;Dummy table for FLgetsnap
; f 1 0 1024 10 1

```



```
f 0 3600
```

```
e
```

```
</CsScore>
```

```
</CsoundSynthesizer>
```

See Also

FLgetsnap, FLloadsnap, FLsetSnapGroup, FLrun, FLsetsnap, FLupdate

Credits

Author: Gabriel Maldonado

New in version 4.22

FLscroll

FLscroll — A FLTK opcode that adds scroll bars to an area.

Description

FLscroll adds scroll bars to an area.

Syntax

```
FLscroll iwidth, iheight [, ix] [, iy]
```

Initialization

iwidth -- width of widget.

iheight -- height of widget.

ix (optional) -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy (optional) -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

Performance

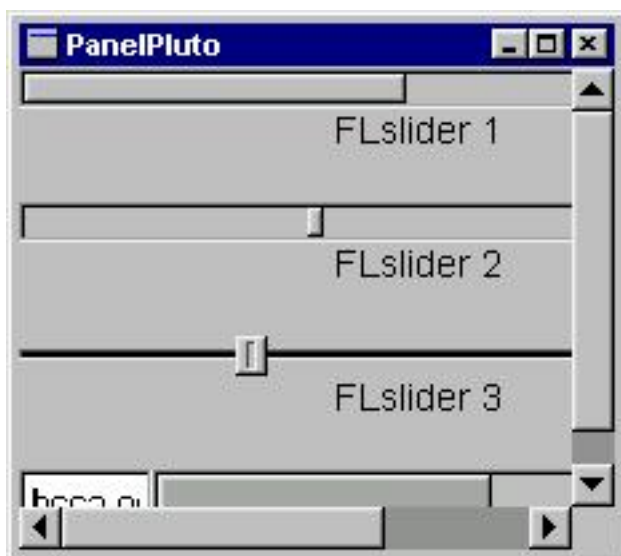
Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuator or other kinds of widgets.

There are no k-rate arguments in containers.

FLscroll adds scroll bars to an area. Normally you must set arguments *iwidth* and *iheight* equal to that of the parent window or other parent container. *ix* and *iy* are optional since they normally are set to zero. For example the following code:

```
FLpanel    "PanelPluto",400,300,100,100
FLscroll   400,300
gk1,ih1 FLslider "FLslider 1", 500, 1000, 2 ,1, -1, 300,15, 20,50
gk2,ih2 FLslider "FLslider 2", 300, 5000, 2 ,3, -1, 300,15, 20,100
gk3,ih3 FLslider "FLslider 3", 350, 1000, 2 ,5, -1, 300,15, 20,150
gk4,ih4 FLslider "FLslider 4", 250, 5000, 1 ,11,-1, 300,30, 20,200
FLscrollEnd
FLpanelEnd
```

will show scroll bars, when the main window size is reduced:



FLscroll.

Examples

Here is an example of the FLscroll opcode. It uses the file *FLscroll.csd* [examples/FLscroll.csd].

Example 169. Example of the FLscroll opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLscroll.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Demonstration of the flscroll opcode which enables
; the use of widget sizes and placings beyond the
; dimensions of the containing panel
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "Text Box", 420, 200, 50, 50
  iwidth = 420
  iheight = 200
  ix = 0
  iy = 0
  FLscroll iwidth, iheight, ix, iy
  ih3 FLbox "DRAG THE SCROLL BAR TO THE RIGHT IN ORDER TO READ THE REST OF THIS TEXT!", 1, 10, 20
  FLscrollEnd
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
endin

</CsInstruments>
<CsScore>

; 'Dummy' score event of 1 hour.
f 0 3600
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

FLgroup, FLgroupEnd, FLpack, FLpackEnd, FLpanel, FLpanelEnd, FLscrollEnd, FLtabs, FLtabsEnd

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLscrollEnd

FLscrollEnd — A FLTK opcode that marks the end of an area with scrollbars.

Description

A FLTK opcode that marks the end of an area with scrollbars.

Syntax

`FLscrollEnd`

Performance

Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuator or other kinds of widgets.

There are no k-rate arguments in containers.

See Also

FLgroup, *FLgroupEnd*, *FLpack*, *FLpackEnd*, *FLpanel*, *FLpanelEnd*, *FLscroll*, *FLtabs*, *FLtabsEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLscroll_end

FLscroll_end — A FLTK opcode that marks the end of an area with scrollbars.

Description

A FLTK opcode that marks the end of an area with scrollbars. This is another name for **FLscrollEnd** provided for compatibility. See *FLscrollEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetAlign

FLsetAlign — Sets the text alignment of a label of a FLTK widget.

Description

FLsetAlign sets the text alignment of the label of the target widget.

Syntax

```
FLsetAlign ialign, ihandle
```

Initialization

ialign -- sets the alignment of the label text of widgets.

The legal values for the *ialign* argument are:

- 1 - align center
- 2 - align top
- 3 - align bottom
- 4 - align left
- 5 - align right
- 6 - align top-left
- 7 - align top-right
- 8 - align bottom-left
- 9 - align bottom-right

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetBox

FLsetBox — Sets the appearance of a box surrounding a FLTK widget.

Description

FLsetBox sets the appearance of a box surrounding the target widget.

Syntax

```
FLsetBox itype, ihandle
```

Initialization

itype -- an integer number that modify the appearance of the target widget.

Legal values for the *itype* argument are:

- 1 - flat box
- 2 - up box
- 3 - down box
- 4 - thin up box
- 5 - thin down box
- 6 - engraved box
- 7 - embossed box
- 8 - border box
- 9 - shadow box
- 10 - rounded box
- 11 - rounded box with shadow
- 12 - rounded flat box
- 13 - rounded up box
- 14 - rounded down box
- 15 - diamond up box
- 16 - diamond down box
- 17 - oval box
- 18 - oval shadow box
- 19 - oval flat box

ihandle -- an integer number (used as unique identifier) taken from the output of a previously loc-

ated widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetColor

FLsetColor — Sets the primary color of a FLTK widget.

Description

FLsetColor sets the primary color of the target widget.

Syntax

```
FLsetColor ired, igreen, iblue, ihandle
```

Initialization

ired -- The red color of the target widget. The range for each RGB component is 0-255

igreen -- The green color of the target widget. The range for each RGB component is 0-255

iblue -- The blue color of the target widget. The range for each RGB component is 0-255

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

Examples

Here is an example of the FLsetcolor opcode. It uses the file *FLsetcolor.csd* [examples/FLsetcolor.csd].

Example 170. Example of the FLsetcolor opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLsetcolor.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Using the opcode flsetcolor to change from the
; default colours for widgets
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "Coloured Sliders", 900, 360, 50, 50
gkfreq, ihandle FLslider "A Red Slider", 200, 5000, -1, 5, -1, 750, 30, 85, 50
iredl = 255
igreenl = 0
ibluel = 0
FLsetColor iredl, igreenl, ibluel, ihandle

gkfreq, ihandle FLslider "A Green Slider", 200, 5000, -1, 5, -1, 750, 30, 85, 150
iredl = 0
igreenl = 255
ibluel = 0
```

```
    FLsetColor ired1, igreen1, iblue1, ihandle

    gkfreq, ihandle FLslider "A Blue Slider", 200, 5000, -1, 5, -1, 750, 30, 85, 250
    ired1 = 0
    igreen1 = 0
    iblue1 = 255
    FLsetColor ired1, igreen1, iblue1, ihandle
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
endin

</CsInstruments>
<CsScore>

; 'Dummy' score event for 1 hour.
f 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLcolor2, FLhide, FLlabel, FLsetAlign, FLsetBox, FLsetColor, FLsetColor2, FLsetFont, FLsetPosition, FLsetSize, FLsetText, FLsetTextColor, FLsetTextSize, FLsetTextType, FLsetVal_i, FLsetVal, FLshow

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLsetColor2

FLsetColor2 — Sets the secondary (or selection) color of a FLTK widget.

Description

FLsetColor2 sets the secondary (or selection) color of the target widget.

Syntax

```
FLsetColor2 ired, igrreen, iblue, ihandle
```

Initialization

ired -- The red color of the target widget. The range for each RGB component is 0-255

igrreen -- The green color of the target widget. The range for each RGB component is 0-255

iblue -- The blue color of the target widget. The range for each RGB component is 0-255

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetFont

FLsetFont — Sets the font type of a FLTK widget.

Description

FLsetFont sets the font type of the target widget.

Syntax

```
FLsetFont ifont, ihandle
```

Initialization

ifont -- sets the the font type of the label of a widget.

Legal values for ifont argument are:

- 1 - Helvetica (same as Arial under Windows)
- 2 - Helvetica Bold
- 3 - Helvetica Italic
- 4 - Helvetica Bold Italic
- 5 - Courier
- 6 - Courier Bold
- 7 - Courier Italic
- 8 - Courier Bold Italic
- 9 - Times
- 10 - Times Bold
- 11 - Times Italic
- 12 - Times Bold Italic
- 13 - Symbol
- 14 - Screen
- 15 - Screen Bold
- 16 - Dingbats

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, FLhide, FLlabel, FLsetAlign, FLsetBox, FLsetColor, FLsetColor2, FLsetFont, FLsetPosition, FLsetSize, FLsetText, FLsetTextColor, FLsetTextSize, FLsetTextType, FLsetVal_i, FLsetVal, FLshow

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetPosition

FLsetPosition — Sets the position of a FLTK widget.

Description

FLsetPosition sets the position of the target widget according to the *ix* and *iy* arguments.

Syntax

```
FLsetPosition ix, iy, ihandle
```

Initialization

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetSize

FLsetSize — Resizes a FLTK widget.

Description

FLsetSize resizes the target widget (not the size of its text) according to the *iwidth* and *iheight* arguments.

Syntax

```
FLsetSize iwidth, iheight, ihandle
```

Initialization

iwidth -- width of widget.

iheight -- height of widget.

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetsnap

FLsetsnap — Stores the current status of all FLTK valutors into a snapshot location.

Description

FLsetsnap stores the current status of all valutors present in the orchestra into a snapshot location (in memory).

Syntax

```
inumsnap, inumval FLsetsnap index [, ifn, igroup]
```

Initialization

inumsnap -- current number of snapshots.

inumval -- number of valutors (whose value is stored in a snapshot) present in current orchestra.

index -- a number referring unequivocally to a snapshot. Several snapshots can be stored in the same bank.

ifn (optional) -- optional argument referring to an already allocated table, to store values of a snapshot.

igroup -- (optional) an integer number referring to a snapshot-related group of widget. It allows to get/set, or to load/save the state of a subset of valutors. Default value is zero that refers to the first group. The group number is determined by the opcode *FLsetSnapGroup*.



Note

The *igroup* parameter has not been yet fully implemented in the current version of csound. Please do not rely on it yet.

Performance

The *FLsetsnap* opcode stores current status of all valutors present in the orchestra into a snapshot location (in memory). Any number of snapshots can be stored in the current bank. Banks are structures that only exist in memory, there are no other reference to them other that they can be accessed by *FLsetsnap*, *FLsavesnap*, *FLloadsnap* and *FLgetsnap* opcodes. Only a single bank can be present in memory.

If the optional *ifn* argument refers to an already allocated and valid table, the snapshot will be stored in the table instead of in the bank. So that table can be accessed from other Csound opcodes.

The *index* argument unequivocally refers to a determinate snapshot. If the value of *index* refers to a previously stored snapshot, all its old values will be replaced with current ones. If *index* refers to a snapshot that doesn't exist, a new snapshot will be created. If the *index* value is not adjacent with that of a previously created snapshot, some empty snapshots will be created. For example, if a location with *index* 0 contains the only and unique snapshot present in a bank and the user stores a new snapshot using *index* 5, all locations between 1 and 4 will automatically contain empty snapshots. Empty snapshots don't contain any data and are neutral.

FLsetsnap outputs the current number of snapshots (the *inumsnap* argument) and the total number of values stored in each snapshot (*inumval*). *inumval* is equal to the number of valutors present in the orchestra.

For purposes of snapshot saving, widgets can be grouped, so that snapshots affect only a defined group of widgets. The opcode *FLsetSnapGroup* is used to specify the group for all widgets declared after it, until the next *FLsetSnapGroup* statement.

See Also

FLgetsnap, *FLloadsnap*, *FLsetSnapGroup*, *FLrun*, *FLsavesnap*, *FLupdate*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetSnapGroup

FLsetSnapGroup — Determines the snapshot group for FL valuator.

Description

FLsetSnapGroup determines the snapshot group of valuator declared after it.

Syntax

FLsetSnapGroup *igroup*

Initialization

igroup -- (optional) an integer number referring to a snapshot-related group of widget. It allows to get/set, or to load/save the state of a subset of valuator.



Note

The *igroup* parameter has not been yet fully implemented in the current version of csound. Please do not rely on it yet.

Performance

For purposes of snapshot saving, widgets can be grouped, so that snapshots affect only a defined group of widgets. The opcode *FLsetSnapGroup* is used to specify the group for all widgets declared after it, until the next *FLsetSnapGroup* statement.

FLsetSnapGroup determines the snapshot group of a declared valuator. To make a valuator belong to a stated group, you have to place *FLsetSnapGroup* just before the declaration of the widget itself. The group stated by *FLsetSnapGroup* lasts for all valuator declared after it, until a new *FLsetSnapGroup* statement with a different group is encountered. If no *FLsetSnapGroup* statement are present in an orchestra, the default group for all widgets will be group zero.

See Also

FLgetsnap, *FLsetsnap*, *FLloadsnap*, *FLsavesnap*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetText

FLsetText — Sets the label of a FLTK widget.

Description

FLsetText sets the label of the target widget to the double-quoted text string provided with the *itext* argument.

Syntax

```
FLsetText "itext", ihandle
```

Initialization

“itext” -- a double-quoted string denoting the text of the label of the widget.

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

Examples

Here is an example of the FLsetText opcode. It uses the file *FLsetText.csd* [examples/FLsetText.csd].

Example 171. Example of the FLsetText opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>

; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o FLsetText.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 128
nchnls = 2

; Example by Giorgio Zucco and Andres Cabrera 2007

FLpanel "FLsetText",250,100,50,50

gk1,giha FLcount "", 1, 20, 1, 20, 1, 200, 40, 20, 20, 0, 1, 0, 1

FLpanelEnd
FLrun

instr 1
; This instrument is triggered by FLcount above each time
; its value changes
iname = i(gk1)
print iname
; Must use FLsetText on the init pass!
if (iname == 1) igoto text1
```

```
if (iname == 2) igoto text2
if (iname == 3) igoto text3

igoto end

text1:
FLsetText "FM",giha
igoto end

text2:
FLsetText "GRANUL",giha
igoto end

text3:
FLsetText "PLUCK",giha
igoto end

end:
    endin

</CsInstruments>
<CsScore>

f 0 3600

</CsScore>
</CsoundSynthesizer>
```

See Also

FLcolor2, FLhide, FLlabel, FLsetAlign, FLsetBox, FLsetColor, FLsetColor2, FLsetFont, FLsetPosition, FLsetSize, FLsetText, FLsetTextColor, FLsetTextSize, FLsetTextType, FLsetVal_i, FLsetVal, FLshow

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetTextColor

FLsetTextColor — Sets the color of the text label of a FLTK widget.

Description

FLsetTextColor sets the color of the text label of the target widget.

Syntax

```
FLsetTextColor ired, iblue, igreen, ihandle
```

Initialization

ired -- The red color of the target widget. The range for each RGB component is 0-255

igreen -- The green color of the target widget. The range for each RGB component is 0-255

iblue -- The blue color of the target widget. The range for each RGB component is 0-255

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetTextSize

FLsetTextSize — Sets the size of the text label of a FLTK widget.

Description

FLsetTextSize sets the size of the text label of the target widget.

Syntax

```
FLsetTextSize isize, ihandle
```

Initialization

isize -- size of the font of the target widget. Normal values are in the order of 15. Greater numbers enlarge font size, while smaller numbers reduce it.

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetTextType

FLsetTextType — Sets some font attributes of the text label of a FLTK widget.

Description

FLsetTextType sets some attributes related to the fonts of the text label of the target widget.

Syntax

```
FLsetTextType itype, ihandle
```

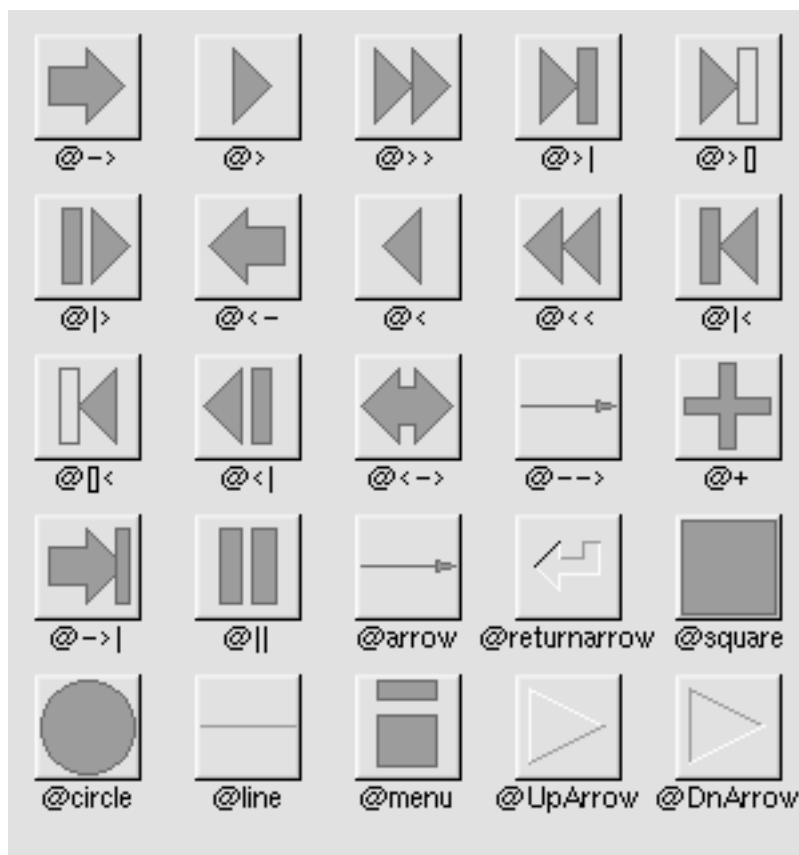
Initialization

itype -- an integer number that modify the appearance of the target widget.

The legal values of *itype* are:

- 0 - normal label
- 1 - no label (hides the text)
- 2 - symbol label (see below)
- 3 - shadow label
- 4 - engraved label
- 5- embossed label
- 6- bitmap label (not implemented yet)
- 7- pixmap label (not implemented yet)
- 8- image label (not implemented yet)
- 9- multi label (not implemented yet)
- 10- free-type label (not implemented yet)

When using *itype*=3 (symbol label), it is possible to assign a graphical symbol instead of the text label of the target widget. In this case, the string of the target label must always start with “@”. If it starts with something else (or the symbol is not found), the label is drawn normally. The following symbols are supported:



FLTK label supported symbols.

The @ sign may be followed by the following optional “formatting” characters, in this order:

1. “#” forces square scaling rather than distortion to the widget’s shape.
2. +[1-9] or -[1-9] tweaks the scaling a little bigger or smaller.
3. [1-9] rotates by a multiple of 45 degrees. “6” does nothing, the others point in the direction of that key on a numeric keypad.

Notice that with *FLbox* and *FLbutton*, it is not necessary to call *FLsetTextType* opcode at all in order to use a symbol. In this case, it is sufficient to set a label starting with “@” followed by the proper formatting string.

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetVal_i

FLsetVal_i — Sets the value of a FLTK valuator to a number provided by the user.

Description

FLsetVal_i forces the value of a valuator to a number provided by the user.

Syntax

```
FLsetVal_i ivalue, ihandle
```

Initialization

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

Performance

ivalue -- Value to set the widget to.



Note

FLsetVal is not fully implemented yet, and may crash in certain cases (e.g. when setting the value of a widget connected to a *FLvalue* widget- in this case use two separate *FLsetVal_i*).

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLsetVal

FLsetVal — Sets the value of a FLTK valuator at control-rate.

Description

FLsetVal is almost identical to *FLsetVal_i*. Except it operates at k-rate and it affects the target valuator only when *ktrig* is set to a non-zero value.

Syntax

```
FLsetVal ktrig, kvalue, ihandle
```

Initialization

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

Performance

ktrig -- triggers the opcode when different than 0.

kvalue -- Value to set the widget to.



Note

FLsetVal is not fully implemented yet, and may crash in certain cases (e.g. when setting the value of a widget connected to a *FLvalue* widget- in this case use two separate *FLsetVal*)

See Also

FLcolor, *FLcolor2*, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLshow

FLshow — Restores the visibility of a previously hidden FLTK widget.

Description

FLshow restores the visibility of a previously hidden widget.

Syntax

```
FLshow ihandle
```

Initialization

ihandle -- an integer number (used as unique identifier) taken from the output of a previously located widget opcode (which corresponds to the target widget). It is used to unequivocally identify the widget when modifying its appearance with this class of opcodes. The user must not set the *ihandle* value directly, otherwise a Csound crash will occur.

See Also

FLcolor2, *FLhide*, *FLlabel*, *FLsetAlign*, *FLsetBox*, *FLsetColor*, *FLsetColor2*, *FLsetFont*, *FLsetPosition*, *FLsetSize*, *FLsetText*, *FLsetTextColor*, *FLsetTextSize*, *FLsetTextType*, *FLsetVal_i*, *FLsetVal*, *FLshow*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLslidBnk

FLslidBnk — A FLTK widget containing a bank of horizontal sliders.

Description

FLslidBnk is a widget containing a bank of horizontal sliders.

Syntax

```
FLslidBnk "names", inumsliders [, ioutable] [, iwidth] [, iheight] [, ix] \  
[, iy] [, itypetable] [, iexptable] [, istart_index] [, iminmaxtable]
```

Initialization

"names" -- a double-quoted string containing the names of each slider. Each slider can have a different name. Separate each name with "@" character, for example: "frequency@amplitude@cutoff". It is possible to not provide any name by giving a single space " ". In this case, the opcode will automatically assign a progressive number as a label for each slider.

inumsliders -- the number of sliders.

ioutable (optional, default=0) -- number of a previously-allocated table in which to store output values of each slider. The user must be sure that table size is large enough to contain all output cells, otherwise a segfault will crash Csound. By assigning zero to this argument, the output will be directed to the zak space in the k-rate zone. In this case, the zak space must be previously allocated with the *zakinit* opcode and the user must be sure that the allocation size is big enough to cover all sliders. The default value is zero (i.e. store output in zak space).

istart_index (optional, default=0) -- an integer number referring to a starting offset of output cell locations. It can be positive to allow multiple banks of sliders to output in the same table or in the zak space. The default value is zero (no offset).

iminmaxtable (optional, default=0) -- number of a previously-defined table containing a list of min-max pairs, referred to each slider. A zero value defaults to the 0 to 1 range for all sliders without necessity to provide a table. The default value is zero.

iexptable (optional, default=0) -- number of a previously-defined table containing a list of identifiers (i.e. integer numbers) provided to modify the behaviour of each slider independently. Identifiers can assume the following values:

- -1 -- exponential curve response
- 0 -- linear response
- number > than 0 -- follow the curve of a previously-defined table to shape the response of the corresponding slider. In this case, the number corresponds to table number.

You can assume that all sliders of the bank have the same response curve (exponential or linear). In this case, you can assign -1 or 0 to *iexptable* without worrying about previously defining any table. The default value is zero (all sliders have a linear response, without having to provide a table).

ityetable (optional, default=0) -- number of a previously-defined table containing a list of identifiers (i.e. integer numbers) provided to modify the aspect of each individual slider independently. Identifiers can assume the following values:

- 0 = Nice slider

- 1 = Fill slider
- 3 = Normal slider
- 5 = Nice slider
- 7 = Nice slider with down-box

You can assume that all sliders of the bank have the same aspect. In this case, you can assign a negative number to *ityetable* without worrying about previously defining any table. Negative numbers have the same meaning of the corresponding positive identifiers with the difference that the same aspect is assigned to all sliders. You can also assign a random aspect to each slider by setting *ityetable* to a negative number lower than -7. The default value is zero (all sliders have the aspect of nice sliders, without having to provide a table).

You can add 20 to a value inside the table to make the slider "plastic", or subtract 20 if you want to set the value for all widgets without defining a table (e.g. -21 to set all sliders types to Plastic Fill slider).

iwidth (optional) -- width of the rectangular area containing all sliders of the bank, excluding text labels, that are placed to the left of that area.

iheight (optional) -- height of the rectangular area containing all sliders of the bank, excluding text labels, that are placed to the left of that area.

ix (optional) -- horizontal position of the upper left corner of the rectangular area containing all sliders belonging to the bank. You have to leave enough space, at the left of that rectangle, in order to make sure labels of sliders to be visible. This is because the labels themselves are external to the rectangular area.

iy (optional) -- vertical position of the upper left corner of the rectangular area containing all sliders belonging to the bank. You have to leave enough space, at the left of that rectangle, in order to make sure labels of sliders to be visible. This is because the labels themselves are external to the rectangular area.

Performance

There are no k-rate arguments, even if cells of the output table (or the zak space) are updated at k-rate.

FLslidBnk is a widget containing a bank of horizontal sliders. Any number of sliders can be placed into the bank (*inumsliders* argument). The output of all sliders is stored into a previously allocated table or into the zak space (*ioutable* argument). It is possible to determine the first location of the table (or of the zak space) in which to store the output of the first slider by means of *istart_index* argument.

Each slider can have an individual label that is placed to the left of it. Labels are defined by the "names" argument. The output range of each slider can be individually set by means of an external table (*iminmaxtable* argument). The curve response of each slider can be set individually, by means of a list of identifiers placed in a table (*ixptable* argument). It is possible to define the aspect of each slider independently or to make all sliders have the same aspect (*ityetable* argument).

The *iwidth*, *iheight*, *ix*, and *iy* arguments determine width, height, horizontal and vertical position of the rectangular area containing sliders. Notice that the label of each slider is placed to the left of them and is not included in the rectangular area containing sliders. So the user should leave enough space to the left of the bank by assigning a proper *ix* value in order to leave labels visible.



IMPORTANT!

Notice that the tables used by *FLslidBnk* must be created with the *figen* opcode and placed in the orchestra before the corresponding valuator. They can not placed in the score. This is because tables placed in the score are created later than the initialization

of the opcodes placed in the header section of the orchestra.

Examples

Here is an example of the FLslidBnk opcode. It uses the file *FLslidBnk.csd* [examples/FLslidBnk.csd].

Example 172. Example of the FLslidBnk opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLslidBnk.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 441
ksmps = 100
nchnls = 1

gitypetable ftgen 0, 0, 8, -2, 1, 1, 3, 3, 5, 5, 7, 7
giouttable ftgen 0, 0, 8, -2, 0, 0.2, 0.3, 0.4, 0.5, 0.6, 0.8, 1

FLpanel "Slider Bank", 400, 380, 50, 50
;Number of sliders
inum = 8
; Table to store output
iouttable = giouttable
; Width of the slider bank in pixels
iwidth = 350
; Height of the slider in pixels
iheight = 160
; Distance of the left edge of the slider
; from the left edge of the panel
ix = 30
; Distance of the top edge of the slider
; from the top edge of the panel
iy = 10
; Table containing fader types
itypetable = gitypetable
FLslidBnk "1@2@3@4@5@6@7@8", inum , iouttable , iwidth , iheight , ix \
, iy , itypetable
FLslidBnk "1@2@3@4@5@6@7@8", inum , iouttable , iwidth , iheight , ix \
, iy + 200 , -23
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
;Dummy instrument
endin

</CsInstruments>
<CsScore>

; Instrument 1 will play a note for 1 hour.
i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```


See Also

FLslider, FLslidBnk2, FLvslidBnk, FLvslidBnk2

Credits

Author: Gabriel Maldonado

New in version 4.22

FLslidBnk2

FLslidBnk2 — A FLTK widget containing a bank of horizontal sliders.

Description

FLslidBnk2 is a widget containing a bank of horizontal sliders.

Syntax

```
FLslidBnk2 "names", inumsliders, ioutable, iconfigtable [,iwidth, iheight, ix, iy, istart_index]
```

Initialization

"names" -- a double-quoted string containing the names of each slider. Each slider can have a different name. Separate each name with "@" character, for example: "frequency@amplitude@cutoff". It is possible to not provide any name by giving a single space ". In this case, the opcode will automatically assign a progressive number as a label for each slider.

inumsliders -- the number of sliders.

ioutable (optional, default=0) -- number of a previously-allocated table in which to store output values of each slider. The user must be sure that table size is large enough to contain all output cells, otherwise a segfault will crash Csound. By assigning zero to this argument, the output will be directed to the zak space in the k-rate zone. In this case, the zak space must be previously allocated with the *zakinit* opcode and the user must be sure that the allocation size is big enough to cover all sliders. The default value is zero (i.e. store output in zak space).

iconfigtable -- in the *FLslidBnk2* and *FLvslidBnk2* opcodes, this table replaces *iminmaxtable*, *iexpstable* and *istyletable*, all these parameters being placed into a single table. This table has to be filled with a group of 5 parameters for each slider in this way:

min1, max1, exp1, style1, min2, max2, exp2, style2, min3, max3, exp3, style3 etc.

for example using GEN02 you can type:

```
inum ftgen 1,0,256, -2, 0,1,0,1, 100, 5000, -1, 3, 50, 200, -1, 5,..... [etcetera]
```

In this example the first slider will be affected by the [0,1,0,1] parameters (the range will be 0 to 1, it will have linear response, and its aspect will be a fill slider), the second slider will be affected by the [100,5000,-1,3] parameters (the range is 100 to 5000, the response is exponential and the aspect is a normal slider), the third slider will be affected by the [50,200,-1,5] parameters (the range is 50 to 200, the behavior exponential, and the aspect is a nice slider), and so on.

iwidth (optional) -- width of the rectangular area containing all sliders of the bank, excluding text labels, that are placed to the left of that area.

iheight (optional) -- height of the rectangular area containing all sliders of the bank, excluding text labels, that are placed to the left of that area.

ix (optional) -- horizontal position of the upper left corner of the rectangular area containing all sliders belonging to the bank. You have to leave enough space, at the left of that rectangle, in order to make sure labels of sliders to be visible. This is because the labels themselves are external to the rectangular area.

iy (optional) -- vertical position of the upper left corner of the rectangular area containing all sliders belonging to the bank. You have to leave enough space, at the left of that rectangle, in order to make

sure labels of sliders to be visible. This is because the labels themselves are external to the rectangular area.

istart_index (optional, default=0) -- an integer number referring to a starting offset of output cell locations. It can be positive to allow multiple banks of sliders to output in the same table or in the zak space. The default value is zero (no offset).

Performance

There are no k-rate arguments, even if cells of the output table (or the zak space) are updated at k-rate.

FLslidBnk2 is a widget containing a bank of horizontal sliders. Any number of sliders can be placed into the bank (*inumslders* argument). The output of all sliders is stored into a previously allocated table or into the zak space (*ioutable* argument). It is possible to determine the first location of the table (or of the zak space) in which to store the output of the first slider by means of *istart_index* argument.

Each slider can have an individual label that is placed to the left of it. Labels are defined by the “*names*” argument. The output range of each slider can be individually set by means of the *min* and *max* values inside the *iconfigtable* table. The curve response of each slider can be set individually, by means of a list of identifiers placed in the *iconfigtable* table (*exp* argument). It is possible to define the aspect of each slider independently or to make all sliders have the same aspect (*style* argument in the *iconfigtable* table).

The *iwidth*, *iheight*, *ix*, and *iy* arguments determine width, height, horizontal and vertical position of the rectangular area containing sliders. Notice that the label of each slider is placed to the left of them and is not included in the rectangular area containing sliders. So the user should leave enough space to the left of the bank by assigning a proper *ix* value in order to leave labels visible.



IMPORTANT!

Notice that the tables used by *FLslidBnk2* must be created with the *ftgen* opcode and placed in the orchestra before the corresponding valuator. They can not be placed in the score. This is because tables placed in the score are created later than the initialization of the opcodes placed in the header section of the orchestra.

Examples

Here is an example of the *FLslidBnk2* opcode. It uses the file *FLslidBnk2.csd* [examples/FLslidBnk2.csd].

Example 173. Example of the *FLslidBnk2* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadac          -M0 ;;RT audio I/O with MIDI in
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 100
nchnls = 2

;Example by Gabriel Maldonado

giElem init 8
giOutTab ftgen 1,0,128, 2,          0
```

```

;min1, max1, exp1, type1, min2, max2, exp2, type2, min3, max3, exp3, type3 etc.
giConfigTab ftgen 2,0,128,-2, .1, 1000, -1, 3, .1, 1000, -1, 3, .1, 1000, -1, 3,
giSine ftgen 3,0,256,10, 1 .1, 5000, -1, 5, .1, 5000, -1, 5, .1, 5000, -1,

FLpanel "This Panel contains a Slider Bank",600,600
FLslidBnk2 "mod1@mod2@mod3@amp@freq1@freq2@freq3@freqPo", giElem, giOutTab, giConfigTab, 40
FLpanel_end

FLrun

instr 1

kmodindex1 init 0
kmodindex2 init 0
kmodindex3 init 0
kamp init 0
kfreq1 init 0
kfreq2 init 0
kfreq3 init 0
kfreq4 init 0

vtablelk giOutTab, kmodindex1, kmodindex2, kmodindex3, kamp, kfreq1, kfreq2, kfreq3, kfreq4

amod1 oscili kmodindex1, kfreq1, giSine
amod2 oscili kmodindex2, kfreq2, giSine
amod3 oscili kmodindex3, kfreq3, giSine
aout oscili kamp, kfreq4+amod1+amod2+amod3, giSine

outs aout, aout
endin

</CsInstruments>
<CsScore>

i1 0 3600
f0 3600

</CsScore>
</CsoundSynthesizer>
```

See Also

FLslider, FLslidBnk, FLvslidBnk, FLvslidBnk2

Credits

Author: Gabriel Maldonado

New in version 5.06

FLslidBnkGetHandle

FLslidBnkGetHandle — gets the handle of last slider bank created.

Description

FLslidBnkGetHandle gets the handle of last slider bank created.

Syntax

`ihandle` **FLslidBnkGetHandle**

Initialization

ihandle - handle of the sliderBnk (to be used to set its values).

Performance

There are no k-rate arguments, even if cells of the output table (or the zak space) are updated at k-rate.

FLslidBnkGetHandle gets the handle of last slider bank created. This opcode must follow corresponding *FLslidBnk* (or *FLvslidBnk*, *FLslidBnk2* and *FLvslidBnk2*) immediately, in order to get its handle.

See the entry for *FLslidBnk2Setk* to see an example of usage.

See Also

FLslider, *FLslidBnk*, *FLslidBnk2*, *FLvslidBnk*, *FLvslidBnk2*, *FLslidBnk2Set*, *FLslidBnk2Setk*

Credits

Author: Gabriel Maldonado

New in version 5.06

FLslidBnkSet

FLslidBnkSet — modify the values of a slider bank.

Description

FLslidBnkSet modifies the values of a slider bank according to an array of values stored in a table.

Syntax

```
FLslidBnkSet ihandle, ifn [, istartIndex, istartSlid, inumSlid]
```

Initialization

ihandle - handle of the sliderBnk (to be used to set its values).

ifn - number of a table containing an array of values to set each slider to.

istartIndex - (optional) starting index of the table element of to be evaluated firstly. Default value is zero

istartSlid - (optional) starting slider to be evaluated. Default 0, denoting the first slider.

inumSlid - (optional) number of sliders to be updated. Default 0, denoting all sliders.

Performance

FLslidBnkSet modifies the values of a slider bank (created with *FLslidBnk* or with *FLvslidBnk*) according to an array of values stored into table *ifn*. It actually allows to update an *FLslidBnk* (or *FLvslidBnk*) bank of sliders (for instance, using the *slider8table* opcode) to a set of values located in a table. User has to set *ihandle* argument to the handle got from *FLslidBnkGetHandle* opcode. It works at init-rate only. It is possible to reset only a range of sliders, by using the optional arguments *istartIndex*, *istartSlid*, *inumSlid*

There is a k-rate version of this opcode called *FLslidBnkSetk*.

See Also

FLslider, *FLslidBnkGetHandle*, *FLslidBnk*, *FLslidBnk2*, *FLvslidBnk*, *FLvslidBnk2* *FLslidBnk2Set*, *FLslidBnkSetk*

Credits

Author: Gabriel Maldonado

New in version 5.06

FLslidBnkSetk

FLslidBnkSetk — modify the values of a slider bank.

Description

FLslidBnkSetk modifies the values of a slider bank according to an array of values stored in a table.

Syntax

```
FLslidBnkSetk ktrig, ihandle, ifn [, istartIndex, istartSlid, inumSlid]
```

Initialization

ihandle - handle of the sliderBnk (to be used to set its values).

ifn - number of a table containing an array of values to set each slider to.

istartIndex - (optional) starting index of the table element of to be evaluated firstly. Default value is zero

istartSlid - (optional) starting slider to be evaluated. Default 0, denoting the first slider.

inumSlid - (optional) number of sliders to be updated. Default 0, denoting all sliders.

Performance

ktrig – the output of *FLslidBnkSetk* consists of a trigger that informs if sliders have to be updated or not. A non-zero value forces the slider to be updated.

FLslidBnkSetk is similar to *FLslidBnkSet* but allows k-rate to modify the values of *FLslidBnk* (*FLslidBnkSetk* can also be used with *FLvslidBnk*, obtaining identical result). It also allows the slider bank to be joined with MIDI. If you are using MIDI (for instance, when using the *slider8table* opcode), *FLslidBnkSetk* changes the values of *FLslidBnk* bank of sliders to a set of values located in a table. This opcode is actually able to serve as a MIDI bridge to the *FLslidBnk* widget when used together with the *sliderXXtable* set of opcodes (see *slider8table* entry for more information). Notice, that, when you want to use table indexing as a curve response, it is not possible to do it directly in the *iconfigtable* configuration of *FLslidBnk2*, when you intend to use the *FLslidBnkSetk* opcode. In fact, corresponding *inputTable* element of *FLslidBnkSetk* must be set in linear mode and respect the 0 to 1 range. Even the corresponding elements of *sliderXXtable* must be set in linear mode and in the normalized range. You can do table indexing later, by using the *tab* and *tb* opcodes, and rescaling output according to max and min values. By the other hand, it is possible to use linear and exponential curve response directly, by setting the actual min-max range and flag both in the *iconfigtable* of corresponding *FLslidBnk2* and in *sliderXXtable*.

FLslidBnkSetk the k-rate version of *FLslidBnk2Set*.

See Also

FLslider, *FLslidBnkGetHandle*, *FLslidBnk*, *FLslidBnk2*, *FLvslidBnk*, *FLvslidBnk2*, *FLslidBnkSet*, *FLslidBnk2Set*, *slider8table*

Credits

Author: Gabriel Maldonado

New in version 5.06

FLslidBnk2Set

FLslidBnk2Set — modify the values of a slider bank.

Description

FLslidBnk2Set modifies the values of a slider bank according to an array of values stored in a table.

Syntax

```
FLslidBnk2Set ihandle, ifn [, istartIndex, istartSlid, inumSlid]
```

Initialization

ihandle - handle of the sliderBnk (to be used to set its values).

ifn - number of a table containing an array of values to set each slider to.

istartIndex - (optional) starting index of the table element of to be evaluated firstly. Default value is zero

istartSlid - (optional) starting slider to be evaluated. Default 0, denoting the first slider.

inumSlid - (optional) number of sliders to be updated. Default 0, denoting all sliders.

Performance

FLslidBnk2Set modifies the values of a slider bank (created with *FLslidBnk2* or with *FLvslidBnk2*) according to an array of values stored into table *ifn*. It actually allows to update an *FLslidBnk2* (or *FLvslidBnk2*) bank of sliders (for instance, using the *slider8table* opcode) to a set of values located in a table. User has to set *ihandle* argument to the handle got from *FLslidBnkGetHandle* opcode. It works at init-rate only. It is possible to reset only a range of sliders, by using the optional arguments *istartIndex*, *istartSlid*, *inumSlid*

FLslidBnk2Set is identical to *FLslidBnkSet*, but works on *FLslidBnk2* and *FLvslidBnk2* instead of *FLslidBnk* and *FLvslidBnk*.

There is a k-rate version of this opcode called *FLslidBnk2Setk*.

See Also

FLslider, *FLslidBnkGetHandle*, *FLslidBnk*, *FLslidBnk2*, *FLvslidBnk*, *FLvslidBnk2* *FLslidBnkSet*, *FLslidBnk2Setk*

Credits

Author: Gabriel Maldonado

New in version 5.06

FLslidBnk2Setk

FLslidBnk2Setk — modify the values of a slider bank.

Description

FLslidBnk2Setk modifies the values of a slider bank according to an array of values stored in a table.

Syntax

```
FLslidBnk2Setk ktrig, ihandle, ifn [, istartIndex, istartSlid, inumSlid]
```

Initialization

ihandle - handle of the sliderBnk (to be used to set its values).

ifn - number of a table containing an array of values to set each slider to.

istartIndex - (optional) starting index of the table element of to be evaluated firstly. Default value is zero

istartSlid - (optional) starting slider to be evaluated. Default 0, denoting the first slider.

inumSlid - (optional) number of sliders to be updated. Default 0, denoting all sliders.

Performance

ktrig – the output of *FLslidBnk2Setk* consists of a trigger that informs if sliders have to be updated or not. A non-zero value forces the slider to be updated.

FLslidBnk2Setk is similar to *FLslidBnkSet* but allows k-rate to modify the values of *FLslidBnk2* (*FLslidBnk2Setk* can also be used with *FLvslidBnk2*, obtaining identical result). It also allows the slider bank to be joined with MIDI. If you are using MIDI (for instance, when using the *slider8table* opcode), *FLslidBnk2Setk* changes the values of *FLslidBnk2* bank of sliders to a set of values located in a table. This opcode is actually able to serve as a MIDI bridge to the *FLslidBnk2* widget when used together with the *sliderXXtable* set of opcodes (see *slider8table* entry for more information). Notice, that, when you want to use table indexing as a curve response, it is not possible to do it directly in the *iconfigtable* configuration of *FLslidBnk2*, when you intend to use the *FLslidBnk2Setk* opcode. In fact, corresponding *inputTable* element of *FLslidBnk2Setk* must be set in linear mode and respect the 0 to 1 range. Even the corresponding elements of *sliderXXtable* must be set in linear mode and in the normalized range. You can do table indexing later, by using the *tab* and *tb* opcodes, and rescaling output according to max and min values. By the other hand, it is possible to use linear and exponential curve response directly, by setting the actual min-max range and flag both in the *iconfigtable* of corresponding *FLslidBnk2* and in *sliderXXtable*.

FLslidBnk2Setk the k-rate version of *FLslidBnk2Set*.

Examples

Here is an example of the *FLslidBnk2Setk* opcode. It uses the file *FLslidBnk2Setk.csd* [examples/FLslidBnk2Setk.csd].

Example 174. Example of the FLslidBnk2Setk opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using com-

mand line flags.

```

<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr      = 44100
ksmps   = 10
nchnls  = 2

;Example by Gabriel Maldonado 2007

giElem  init 8
giOutTab  ftgen 1,0,128, 2,          0
giSine   ftgen 3,0,256,10, 1
giOutTab2 ftgen 4,0,128, 2,          0

itab  ftgen 29, 0, 129, 5, .002, 128, 1      ;** exponential ascending curve for slider mapping
giExpTab  ftgen 30, 0, 129, -24, itab, 0, 1    ;** rescaled curve for slider mapping

giConfigTab  ftgen 2,0,128,-2,          1,          500, -1,          13, \
                                           1,          500, -1,          13, \
                                           1,          500, -1,          13, \
                                           1,          5000, -1,          13, \
\
                                           1,          1000,          -1,          5, \
                                           1,          1000,          -1,          5, \
                                           1,          1000,          -1,          5, \
                                           1,          5000,          -1,          5

FLpanel "Multiple FM",600,600
FLslidBnk2 "mod1@mod2@mod3@amp@freq1@freq2@freq3@freqPo", giElem, giOutTab2, giConfigTab, 4
giHandle FLslidBnkGetHandle

FLpanel_end

FLrun

instr 1

ktrig slider8table 1, giOutTab, 0,\
; ctl min max init func
27, 1,          500, 3, -1, \;1 repeat rate
28, 1,          500, 4, -1, \;2 random freq. amount
29, 1,          500, 1, -1, \;3 random amp. amount
30, 1, 5000, 1, -1, \;4 number of concurrent loop points
\
31, 1,          1000, 1, -1, \;5 kloop1
32, 1,          1000, 1, -1, \;6 kloop2
33, 1,          1000, 1, -1, \;7 kloop3
34, 1,          1000, 1, -1, \;8 kloop4

kmodindex1 init 0
kmodindex2 init 0
kmodindex3 init 0
kamp init 0
kfreq1 init 0
kfreq2 init 0
kfreq3 init 0
kfreq4 init 0

vtable1k giOutTab2, kmodindex1, kmodindex2, kmodindex3, kamp, kfreq1, kfreq2, kfreq3, kf

; *kflag, *ihandle, *ifn, *startInd, *startSlid, *numSlid;
FLslidBnk2Setk ktrig, giHandle, giOutTab, 0, 0, giElem
printk2 kmodindex1
printk2 kmodindex2,10
printk2 kmodindex3,20
printk2 kamp,30

amod1 oscili kmodindex1, kfreq1, giSine
amod2 oscili kmodindex2, kfreq2, giSine
amod3 oscili kmodindex3, kfreq3, giSine
aout oscili kamp, kfreq4+amod1+amod2+amod3, giSine
outs aout, aout

endin

```

```
</CsInstruments>
<CsScore>
i1 0 3600
</CsScore>
</CsoundSynthesizer>
```

See Also

FLslider, FLslidBnkGetHandle, FLslidBnk, FLslidBnk2, FLvslidBnk, FLvslidBnk2 FLslidBnkSet, FLslidBnk2Set, slider8table

Credits

Author: Gabriel Maldonado

New in version 5.06

FLslider

FLslider — Puts a slider into the corresponding FLTK container.

Description

FLslider puts a slider into the corresponding container.

Syntax

```
kout, ihandle FLslider "label", imin, imax, iexp, itype, idisp, iwidth, \  
iheight, ix, iy
```

Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLslider* and must not be set by the user label. (The user label is a double-quoted string containing some user-provided text placed near the widget.)

“label” -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

imin -- minimum value of output range (corresponds to the left value for horizontal sliders, and the top value for vertical sliders).

imax -- maximum value of output range (corresponds to the right value for horizontal sliders, and the bottom value for vertical sliders).

The *imin* argument may be greater than *imax* argument. This has the effect of “reversing” the object so the larger values are in the opposite direction. This also switches which end of the filled sliders is filled.

iexp -- an integer number denoting the behaviour of valuator:

- 0 = valuator output is linear
- -1 = valuator output is exponential

All other positive numbers for *iexp* indicate the number of an existing table that is used for indexing. Linear interpolation is provided in table indexing. A negative table number suppresses interpolation.



IMPORTANT!

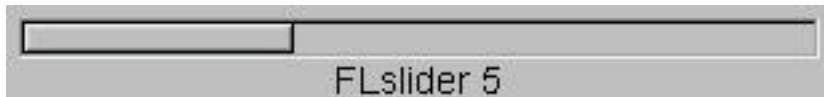
Notice that the tables used by valuator must be created with the *ftgen* opcode and placed in the orchestra before the corresponding valuator. They can not placed in the score. This is because tables placed in the score are created later than the initialization of the opcodes placed in the header section of the orchestra.

itype -- an integer number denoting the appearance of the valuator.

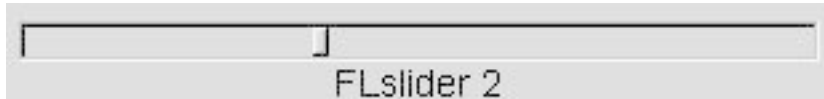
The *itype* argument can be set to the following values:

- 1 - shows a horizontal fill slider

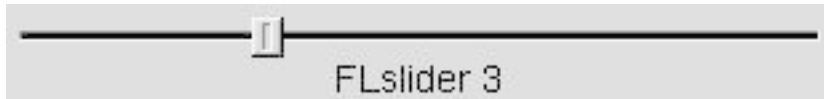
- 2 - a vertical fill slider
- 3 - a horizontal engraved slider
- 4 - a vertical engraved slider
- 5 - a horizontal nice slider
- 6 - a vertical nice slider
- 7 - a horizontal up-box nice slider
- 8 - a vertical up-box nice slider



FLslider - a horizontal fill slider (itype=1).



FLslider - a horizontal engraved slider (itype=3).



FLslider - a horizontal nice slider (itype=5).

You can also create "plastic" looking sliders by adding 20 to *itype*.

idisp -- a handle value that was output from a previous instance of the *FLvalue* opcode to display the current value of the current valuator in the *FLvalue* widget itself. If the user doesn't want to use this feature that displays current values, it must be set to a negative number by the user.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

Performance

kout -- output value

FLsliders are created with the minimum value by default in the left/at the top. If you want to reverse the slider, reverse the values. See the example below.

Examples

Here is an example of the FLslider opcode. It uses the file *FLslider.csd* [examples/FLslider.csd].

Example 175. Example of the FLslider opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using com-

mand line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLslider.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; A sine with oscillator with flslider controlled frequency
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "Frequency Slider", 900, 400, 50, 50
; Minimum value output by the slider
imin = 200
; Maximum value output by the slider
imax = 5000
; Logarithmic type slider selected
iexp = -1
; Slider graphic type (5='nice' slider)
itype = 5
; Display handle (-1=not used)
idisp = -1
; Width of the slider in pixels
iwidth = 750
; Height of the slider in pixels
iheight = 30
; Distance of the left edge of the slider
; from the left edge of the panel
ix = 125
; Distance of the top edge of the slider
; from the top edge of the panel
iy = 50

gkfreq, ihandle FLslider "Frequency", imin, imax, iexp, itype, idisp, iwidth, iheight, ix, iy
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

;Set the widget's initial value
FLsetVal_i 300, ihandle

instr 1
iamp = 15000
ifn = 1
kfreq portk gkfreq, 0.005 ;Smooth gkfreq to avoid zipper noise
asig oscili iamp, kfreq, ifn
out asig
endin

</CsInstruments>
<CsScore>

; Function table that defines a single cycle
; of a sine wave.
f 1 0 1024 10 1

; Instrument 1 will play a note for 1 hour.
i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the FLslider opcode, showing the slider types and other options. It also shows the usage of FLvalue to display a widget's contents. It uses the file *FLslider-2.csd* [examples/FLslider-2.csd].

Example 176. More complex example of the FLslider opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in      No messages
-odac            -iadc         -d            ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLslider-2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 441
ksmps = 100
nchnls = 1

;By Andres Cabrera 2007

FLpanel "Slider Types", 410, 260, 50, 50
; Distance of the left edge of the slider
; from the left edge of the panel
ix = 10
; Distance of the top edge of the slider
; from the top edge of the panel
iy = 10
; Create boxes to display widget values
givalue1 FLvalue "1", 60, 20, ix + 330, iy
givalue3 FLvalue "3", 60, 20, ix + 330, iy + 40
givalue5 FLvalue "5", 60, 20, ix + 330, iy + 80

givalue2 FLvalue "2", 60, 20, ix + 60, iy + 140
givalue4 FLvalue "4", 60, 20, ix + 195, iy + 140
givalue6 FLvalue "6", 60, 20, ix + 320, iy + 140

;Horizontal sliders
gkdummy1, gihandle1 FLslider "Type 1", 200, 5000, -1, 1, givalue1, 320, 20, ix, iy
gkdummy3, gihandle3 FLslider "Type 3", 0, 15000, 0, 3, givalue3, 320, 20, ix, iy + 40
; Reversed slider
gkdummy5, gihandle5 FLslider "Type 5", 1, 0, 0, 5, givalue5, 320, 20, ix, iy + 80

;Vertical sliders
gkdummy2, gihandle2 FLslider "Type 2", 0, 1, 0, 2, givalue2, 20, 100, ix+ 30 , iy + 120
; Reversed slider
gkdummy4, gihandle4 FLslider "Type 4", 1, 0, 0, 4, givalue4, 20, 100, ix + 165 , iy + 120
gkdummy6, gihandle6 FLslider "Type 6", 0, 1, 0, 6, givalue6, 20, 100, ix + 290 , iy + 120
FLpanelEnd

FLpanel "Plastic Slider Types", 410, 300, 150, 150
; Distance of the left edge of the slider
; from the left edge of the panel
ix = 10
; Distance of the top edge of the slider
; from the top edge of the panel
iy = 10
; Create boxes to display widget values
givalue21 FLvalue "21", 60, 20, ix + 330, iy
givalue23 FLvalue "23", 60, 20, ix + 330, iy + 40
givalue25 FLvalue "25", 60, 20, ix + 330, iy + 80

givalue22 FLvalue "22", 60, 20, ix + 60, iy + 140
givalue24 FLvalue "24", 60, 20, ix + 195, iy + 140
givalue26 FLvalue "26", 60, 20, ix + 320, iy + 140

;Horizontal sliders
gkdummy21, gihandle21 FLslider "Type 21", 200, 5000, -1, 21, givalue21, 320, 20, ix, iy
gkdummy23, gihandle23 FLslider "Type 23", 0, 15000, 0, 23, givalue23, 320, 20, ix, iy + 40
; Reversed slider
gkdummy25, gihandle25 FLslider "Type 25", 1, 0, 0, 25, givalue25, 320, 20, ix, iy + 80

;Vertical sliders
gkdummy22, gihandle22 FLslider "Type 22", 0, 1, 0, 22, givalue22, 20, 100, ix+ 30 , iy + 120
; Reversed slider
gkdummy24, gihandle24 FLslider "Type 24", 1, 0, 0, 24, givalue24, 20, 100, ix + 165 , iy + 120
gkdummy26, gihandle26 FLslider "Type 26", 0, 1, 0, 26, givalue26, 20, 100, ix + 290 , iy + 120
;Button to add color to the sliders
gkcolors, ihdummy FLbutton "Color", 1, 0, 21, 150, 30, 30, 260, 0, 10, 0, 1
FLpanelEnd
FLrun

;Set some widget's initial value
FLsetVal_i 500, gihandle1
FLsetVal_i 1000, gihandle3

instr 10
; Set the color of widgets
```



```
FLsetColor 200, 230, 0, gihandle1
FLsetColor 0, 123, 100, gihandle2
FLsetColor 180, 23, 12, gihandle3
FLsetColor 10, 230, 0, gihandle4
FLsetColor 0, 0, 0, gihandle5
FLsetColor 0, 0, 0, gihandle6

FLsetColor 200, 230, 0, givalue1
FLsetColor 0, 123, 100, givalue2
FLsetColor 180, 23, 12, givalue3
FLsetColor 10, 230, 0, givalue4
FLsetColor 255, 255, 255, givalue5
FLsetColor 255, 255, 255, givalue6

FLsetColor2 20, 23, 100, gihandle1
FLsetColor2 200,0,123, gihandle2
FLsetColor2 180, 180, 100, gihandle3
FLsetColor2 180, 23, 12, gihandle4
FLsetColor2 180, 180, 100, gihandle5
FLsetColor2 180, 23, 12, gihandle6

FLsetColor 200, 230, 0, gihandle21
FLsetColor 0, 123, 100, gihandle22
FLsetColor 180, 23, 12, gihandle23
FLsetColor 10, 230, 0, gihandle24
FLsetColor 0, 0, 0, gihandle25
FLsetColor 0, 0, 0, gihandle26

FLsetColor 200, 230, 0, givalue21
FLsetColor 0, 123, 100, givalue22
FLsetColor 180, 23, 12, givalue23
FLsetColor 10, 230, 0, givalue24
FLsetColor 255, 255, 255, givalue25
FLsetColor 255, 255, 255, givalue26

FLsetColor2 20, 23, 100, gihandle21
FLsetColor2 200,0,123, gihandle22
FLsetColor2 180, 180, 100, gihandle23
FLsetColor2 180, 23, 12, gihandle24
FLsetColor2 180, 180, 100, gihandle25
FLsetColor2 180, 23, 12, gihandle26

; Slider values must be updated for colors to change
FLsetVal_i 250, gihandle1
FLsetVal_i 0.5, gihandle2
FLsetVal_i 0, gihandle3
FLsetVal_i 0, gihandle4
FLsetVal_i 0, gihandle5
FLsetVal_i 0.5, gihandle6
FLsetVal_i 250, gihandle21
FLsetVal_i 0.5, gihandle22
FLsetVal_i 500, gihandle23
FLsetVal_i 0, gihandle24
FLsetVal_i 0, gihandle25
FLsetVal_i 0.5, gihandle26

endin

</CsInstruments>
<CsScore>
f 0 3600 ;Dummy table to make csound wait for realtime events

e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLcount, FLjoy, FLknob, FLroller, FLslidBnk, FLtext

Credits

Author: Gabriel Maldonado

New in version 4.22

February 2004. Thanks to a note from Dave Phillips, deleted the extraneous istep parameter.

Example written by Iain McCurdy, edited by Kevin Conder.

FLtabs

FLtabs — Creates a tabbed FLTK interface.

Description

FLtabs is the “file card tabs” interface that allows useful to display several areas containing widgets in the same windows, alternatively. It must be used together with *FLgroup*, another container that groups child widgets.

Syntax

```
FLtabs iwidth, iheight, ix, iy
```

Initialization

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window. Expressed in pixels.

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window. Expressed in pixels.

Performance

Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuators or other kinds of widgets.

There are no k-rate arguments in containers.

FLtabs is a “file card tabs” interface that is useful to display several alternate areas containing widgets in the same window.



FLtabs.

It must be used together with *FLgroup*, another FLTK container opcode that groups child widgets.

Examples

The following example code:

```
FLpanel "Panel1", 450, 550, 100, 100
FLscroll 450, 550, 0, 0
FLtabs 400, 550, 5, 5
FLgroup "sliders", 380, 500, 10, 40, 1
gk1, ihs FLslider "FLslider 1", 500, 1000, 2, 1, -1, 300, 15, 20, 50
```

```

gk2,ihs FLslider "FLslider 2", 300, 5000, 2 ,3, -1, 300,15, 20,100
gk3,ihs FLslider "FLslider 3", 350, 1000, 2 ,5, -1, 300,15, 20,150
gk4,ihs FLslider "FLslider 4", 250, 5000, 1 ,11, -1, 300,30, 20,200
gk5,ihs FLslider "FLslider 5", 220, 8000, 2 ,1, -1, 300,15, 20,250
gk6,ihs FLslider "FLslider 6", 1, 5000, 1 ,13, -1, 300,15, 20,300
gk7,ihs FLslider "FLslider 7", 870, 5000, 1 ,15, -1, 300,30, 20,350
gk8,ihs FLslider "FLslider 8", 20, 20000, 2 ,6, -1, 30,400, 350,50
      FLgroupEnd

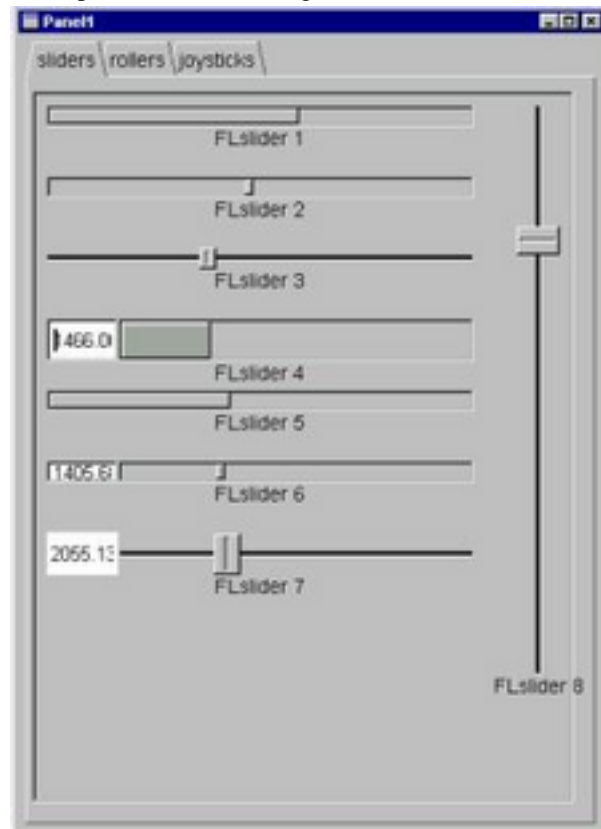
      FLgroup "rollers",380,500, 10,30,2
gk1,ihr FLroller "FLroller 1", 50, 1000,.1,2 ,1 ,-1, 200,22, 20,50
gk2,ihr FLroller "FLroller 2", 80, 5000,1,2 ,1 ,-1, 200,22, 20,100
gk3,ihr FLroller "FLroller 3", 50, 1000,.1,2 ,1 ,-1, 200,22, 20,150
gk4,ihr FLroller "FLroller 4", 80, 5000,1,2 ,1 ,-1, 200,22, 20,200
gk5,ihr FLroller "FLroller 5", 50, 1000,.1,2 ,1 ,-1, 200,22, 20,250
gk6,ihr FLroller "FLroller 6", 80, 5000,1,2 ,1 ,-1, 200,22, 20,300
gk7,ihr FLroller "FLroller 7",50, 5000,1,1 ,2 ,-1, 30,300, 280,50
      FLgroupEnd

      FLgroup "joysticks",380,500, 10,40,3
gk1,gk2,ihj1,ihj2 FLjoy "FLjoy", 50, 18000, 50, 18000,2,2,-1,-1,300,300,30,60
      FLgroupEnd

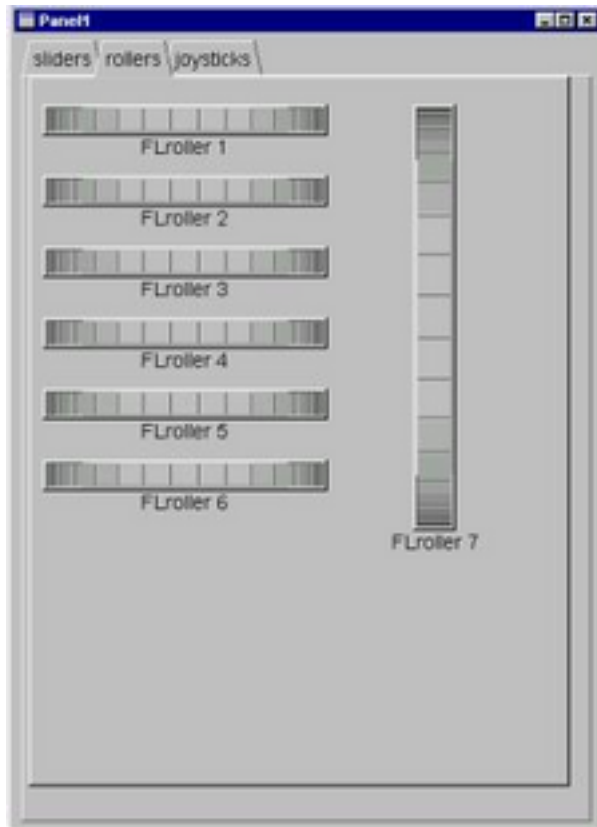
      FLtabsEnd
      FLscrollEnd
      FLpanelEnd

```

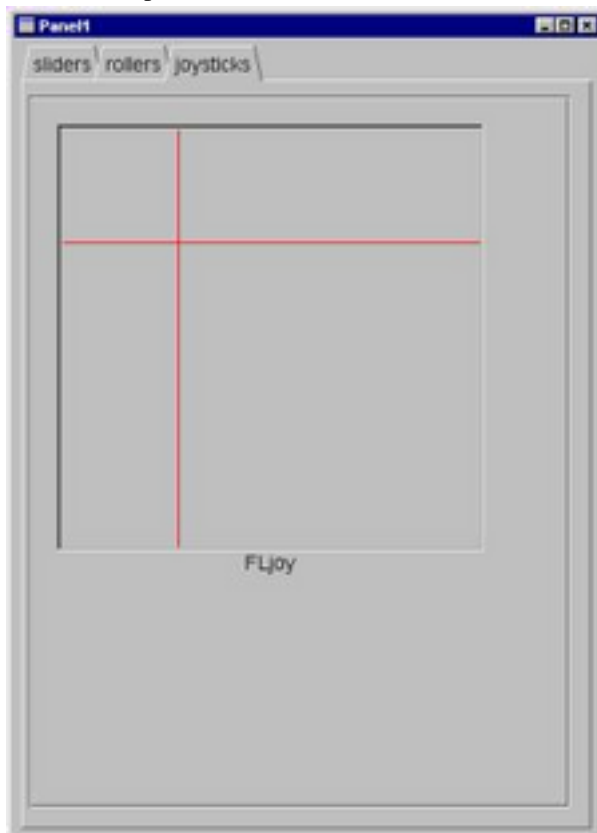
...will produce the following result:



FLtabs example, sliders tab.



FLtabs example, rollers tab.



FLtabs example, joysticks tab.
(Each picture shows a different tab selection inside the same window.)

Examples

Here is an example of the FLtabs opcode. It uses the file *FLtabs.csd* [examples/FLtabs.csd].

Example 177. Example of the FLtabs opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc       -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLtabs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; A single oscillator with frequency, amplitude and
; panning controls on separate file tab cards
sr = 44100
kr = 441
ksmps = 100
nchnls = 2

FLpanel "Tabs", 300, 350, 100, 100
itabswidth = 280
itabsheight = 330
ix = 5
iy = 5
FLtabs itabswidth,itabsheight, ix,iy

    itablwidth = 280
    itablheight = 300
    itablx = 10
    itably = 40
    FLgroup "Tab 1", itablwidth, itablheight, itablx, itably
        gkfreq, i1 FLknob "Frequency", 200, 5000, -1, 1, -1, 70, 70, 130
        FLsetVal_i 400, i1
    FLgroupEnd

    itab2width = 280
    itab2height = 300
    itab2x = 10
    itab2y = 40
    FLgroup "Tab 2", itab2width, itab2height, itab2x, itab2y
        gkamp, i2 FLknob "Amplitude", 0, 15000, 0, 1, -1, 70, 70, 130
        FLsetVal_i 15000, i2
    FLgroupEnd

    itab3width = 280
    itab3height = 300
    itab3x = 10
    itab3y = 40
    FLgroup "Tab 3", itab3width, itab3height, itab3x, itab3y
        gkpan, i3 FLknob "Pan position", 0, 1, 0, 1, -1, 70, 70, 130
        FLsetVal_i 0.5, i3
    FLgroupEnd
FLtabsEnd
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
    ifn = 1
        asig oscili gkamp, gkfreq, ifn
        outs asig*(1-gkpan), asig*gkpan
endin

</CsInstruments>
<CsScore>

; Function table that defines a single cycle
; of a sine wave.
f 1 0 1024 10 1

; Instrument 1 will play a note for 1 hour.
i 1 0 3600
```

e

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

FLgroup, FLgroupEnd, FLpack, FLpackEnd, FLpanel, FLpanelEnd, FLscroll, FLscrollEnd, FLtabsEnd

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLtabsEnd

FLtabsEnd — Marks the end of a tabbed FLTK interface.

Description

Marks the end of a tabbed FLTK interface.

Syntax

`FLtabsEnd`

Performance

Containers are useful to format the graphic appearance of the widgets. The most important container is *FLpanel*, that actually creates a window. It can be filled with other containers and/or valuators or other kinds of widgets.

There are no k-rate arguments in containers.

See Also

FLgroup, *FLgroupEnd*, *FLpack*, *FLpackEnd*, *FLpanel*, *FLpanelEnd*, *FLscroll*, *FLscrollEnd*, *FLtabs*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLtabs_end

FLtabs_end — Marks the end of a tabbed FLTK interface.

Description

Marks the end of a tabbed FLTK interface. This is another name for **FLtabsEnd** provided for compatibility. See *FLtabsEnd*

Credits

Author: Gabriel Maldonado

New in version 4.22

FLtext

FLtext — A FLTK widget opcode that creates a textbox.

Description

FLtext allows the user to modify a parameter value by directly typing it into a text field.

Syntax

```
kout, ihandle FLtext "label", imin, imax, istep, itype, iwidth, \  
         iheight, ix, iy
```

Initialization

ihandle -- a handle value (an integer number) that unequivocally references a corresponding widget. This is used by other opcodes that modify a widget's properties (see *Modifying FLTK Widget Appearance*). It is automatically output by *FLtext* and must not be set by the user label. (The user label is a double-quoted string containing some user-provided text placed near the widget.)

"label" -- a double-quoted string containing some user-provided text, placed near corresponding widget.

imin -- minimum value of output range.

imax -- maximum value of output range.

istep -- a floating-point number indicating the increment of valuator value corresponding to of each mouse click. The *istep* argument allows the user to arbitrarily slow roller's motion, enabling arbitrary precision.

itype -- an integer number denoting the appearance of the valuator.

The *itype* argument can be set to the following values:

- 1 - normal behaviour
- 2 - dragging operation is suppressed, instead it will appear two arrow buttons. A mouse-click on one of these buttons can increase/decrease the output value.
- 3 - text editing is suppressed, only mouse dragging modifies the output value.

iwidth -- width of widget.

iheight -- height of widget.

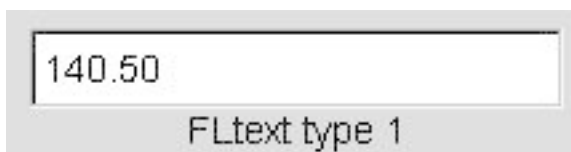
ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

Performance

kout -- output value

FLtext allows the user to modify a parameter value by directly typing it into a text field:



FLtext.

Its value can also be modified by clicking on it and dragging the mouse horizontally. The *istep* argument allows the user to arbitrarily set the response on mouse dragging.

Examples

Here is an example of the FLtext opcode. It uses the file *FLtext.csd* [examples/FLtext.csd].

Example 178. Example of the FLtext opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o FLtext.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; A sine with oscillator with fltext box controlled
; frequency either click and drag or double click and
; type to change frequency value
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "Frequency Text Box", 270, 600, 50, 50
; Minimum value output by the text box
imin = 200
; Maximum value output by the text box
imax = 5000
; Step size
istep = 1
; Text box graphic type
itype = 1
; Width of the text box in pixels
iwidth = 70
; Height of the text box in pixels
iheight = 30
; Distance of the left edge of the text box
; from the left edge of the panel
ix = 100
; Distance of the top edge of the text box
; from the top edge of the panel
iy = 300

gkfreq,ihandle FLtext "Enter the frequency", imin, imax, istep, itype, iwidth, iheight, ix, iy
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
iamp = 15000
ifn = 1
asig oscili iamp, gkfreq, ifn
out asig
endin

</CsInstruments>
<CsScore>

; Function table that defines a single cycle
; of a sine wave.
```

```
f 1 0 1024 10 1
; Instrument 1 will play a note for 1 hour.
i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLcount, FLjoy, FLknob, FLroller, FLslider

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLupdate

FLupdate — Same as the FLrun opcode.

Description

Same as the *FLrun* opcode.

Syntax

FLupdate

fluidAllOut

fluidAllOut — Collects all audio from all Fluidsynth engines in a performance

Syntax

```
aleft, aright fluidAllOut
```

Description

Collects all audio from all Fluidsynth engines in a performance

Performance

aleft -- Left channel audio output.

aright -- Right channel audio output.

Invoke fluidAllOut in an instrument definition numbered higher than any fluidcontrol instrument definitions. All SoundFonts send their audio output to this one opcode. Send a note with an indefinite duration to this instrument to turn the SoundFonts on for as long as required.

In this implementation, SoundFont effects such as chorus or reverb are used if and only if they are defaults for the preset. There is no means of turning such effects on or off, or of changing their parameters, from Csound.

Examples

Here is an example of the fluidsynth opcodes. It uses the file *fluidAllOut.orc* [examples/fluidAllOut.orc].

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
odbfs = 32767

; LOAD SOUNDFONTS
gienginenum1 fluidEngine
gienginenum2 fluidEngine
isfnum1 fluidLoad "Piano Steinway Grand Model C (21,738KB).sf2", gienginenum1, 1
; Bright Steinway, program 1, channel 1
fluidProgramSelect gienginenum1, 1, isfnum1, 0, 1
; Concert Steinway with reverb, program 2, channel 3
fluidProgramSelect gienginenum1, 3, isfnum1, 0, 2
isfnum2 fluidLoad "63.3mg The Sound Site Album Bank V1.0.SF2", gienginenum2, 1
; General MIDI, program 50, channel 2
fluidProgramSelect gienginenum2, 2, isfnum2, 0, 50

; SEND NOTES TO STEINWAY SOUNDFONT

instr 1 ; FluidSynth Steinway Rev
; INITIALIZATION
mididefault 60, p3 ; Default duration of 60 -- overridden by score.
midinoteonkey p4, p5 ; Channels MIDI input to pfields.
; Use channel assigned in fluidload.
ichannel = 1
ikey = p4
ivelocity = p5
istatus = 144
fluidControl gienginenum1, istatus, ichannel, ikey, ivelocity
endin

instr 2 ; GM soundfont
; INITIALIZATION
mididefault 60, p3 ; Default duration of 60 -- overridden by score.
midinoteonkey p4, p5 ; Channels MIDI input to pfields.
```

```
; Use channel assigned in fluidload.
ichannel = 2
ikey = p4
ivelocity = p5
istatus = 144
fluidNote gienginenum2, ichannel, ikey, ivelocity
endin

instr 3 ; FluidSynth Steinway Rev
; INITIALIZATION
mididefault 60, p3 ; Default duration of 60 -- overridden by score.
midinoteonkey p4, p5 ; Channels MIDI input to pfields.
; Use channel assigned in fluidload.
ichannel = 3
ikey = p4
ivelocity = p5
istatus = 144
fluidNote gienginenum1, ichannel, ikey, ivelocity
endin

; COLLECT AUDIO FROM ALL SOUND FONTS

instr 100 ; Fluidsynth output
; INITIALIZATION
; Normalize so iamplitude for p5 of 80 == ampdb(80).
iamplitude = ampdb(p5) * (10000.0 / 0.1)
; AUDIO
aleft, aright fluidAllOut
outs aleft * iamplitude, aright * iamplitude
endin
```

Here is another more complex example of the fluidsynth opcodes written by Istvan Varga. It uses the file *fluidcomplex.csd* [examples/fluidcomplex.csd].

```
<CsoundSynthesizer>
<CsOptions>
-d -m229 -o dac -T -F midifile.mid
</CsOptions>
<CsInstruments>
sr = 44100
ksmps = 128
nchnls = 2
0dbfs = 1

; Example by Istvan Varga

; disable triggering of instruments by MIDI events

ichn = 1
lp1:
    massign    ichn, 0
    loop_le    ichn, 1, 16, lp1
    pgmassign  0, 0

; initialize FluidSynth

gifld fluidEngine
gisf2 fluidLoad "07AcousticGuitar.sf2", gifld, 1

; k-rate version of fluidProgramSelect

opcode fluidProgramSelect_k, 0, kkkkk
    keng, kchn, ksf2, kbnk, kpre xin
    igoto      skipInit
doInit:
    fluidProgramSelect i(keng), i(kchn), i(ksf2), i(kbnk), i(kpre)
    reinit    doInit
    rireturn
skipInit:
endop

instr 1
; initialize channels
kchn init 1
if (kchn == 1) then
lp2:
    fluidControl gifld, 192, kchn - 1, 0, 0
    fluidControl gifld, 176, kchn - 1, 7, 100
    fluidControl gifld, 176, kchn - 1, 10, 64
    loop_le kchn, 1, 16, lp2
endif

; send any MIDI events received to FluidSynth
nxt:
```

```
kst, kch, kd1, kd2 midin
if (kst != 0) then
  if (kst != 192) then
    fluidControl gifld, kst, kch - 1, kd1, kd2
  else
    fluidProgramSelect_k gifld, kch - 1, gisf2, 0, kd1
  endif
  kgoto nxt
endif

; get audio output from FluidSynth
aL, aR fluidOut gifld
      outs      aL, aR
endin

</CsInstruments>
<CsScore>

i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

fluidEngine, fluidNote, fluidLoad

Credits

Opcode by Michael Gogins (gogins at pipeline dot com). Thanks to Peter Hanappe for Fluidsynth, and to Steven Yi for seeing that it is necessary to break up the Fluidsynth into several different Csound opcodes.

fluidCCi

fluidCCi — Sends a MIDI controller data message to fluid.

Syntax

```
fluidCCi iEngineNumber, iChannelNumber, iControllerNumber, iValue
```

Description

Sends a MIDI controller data (MIDI controller number and value to use) message to a fluid engine by number on the user specified MIDI channel number.

Initialization

iEngineNumber -- engine number assigned from fluidEngine

iChannelNumber -- MIDI channel number to which the Fluidsynth program is assigned: from 0 to 255. MIDI channels numbered 16 or higher are virtual channels.

iControllerNumber -- MIDI controller number to use for this message

iValue -- value to set for controller (usually 0-127)

Performance

This opcode is useful for setting controller values at init time. For continuous changes, use fluidCCk.

See Also

fluidEngine, fluidNote, fluidLoad, fluidCCk

Credits

Michael Gogins (gogins at pipeline dot com), Steven Yi. Thanks to Peter Hanappe for Fluidsynth.

fluidCCk

fluidCCk — Sends a MIDI controller data message to fluid.

Syntax

```
fluidCCk iEngineNumber, iChannelNumber, iControllerNumber, kValue
```

Description

Sends a MIDI controller data (MIDI controller number and value to use) message to a fluid engine by number on the user specified MIDI channel number.

Initialization

iEngineNumber -- engine number assigned from fluidEngine

iChannelNumber -- MIDI channel number to which the Fluidsynth program is assigned: from 0 to 255. MIDI channels numbered 16 or higher are virtual channels.

iControllerNumber -- MIDI controller number to use for this message

Performance

kValue -- value to set for controller (usually 0-127)

See Also

fluidEngine, fluidNote, fluidLoad, fluidCCi

Credits

Michael Gogins (gogins at pipeline dot com), Steven Yi. Thanks to Peter Hanappe for Fluidsynth.

fluidControl

fluidControl — Sends MIDI note on, note off, and other messages to a SoundFont preset.

Syntax

```
fluidControl ienginenum, kstatus, kchannel, kdata1, kdata2
```

Description

The fluid opcodes provide a simple Csound opcode wrapper around Peter Hanappe's Fluidsynth SoundFont2 synthesizer. This implementation accepts any MIDI note on, note off, controller, pitch bend, or program change message at k-rate. Maximum polyphony is 4096 simultaneously sounding voices. Any number of SoundFonts may be loaded and played simultaneously.

Initialization

ienginenum -- engine number assigned from fluidEngine

Performance

kstatus -- MIDI channel message status byte: 128 for note off, 144 for note on, 176 for control change, 192 for program change, or 224 for pitch bend.

kchannel -- MIDI channel number to which the Fluidsynth program is assigned: from 0 to 255. MIDI channels numbered 16 or higher are virtual channels.

kdata1 -- For note on, MIDI key number: from 0 (lowest) to 127 (highest), where 60 is middle C. For continuous controller messages, controller number.

kdata2 -- For note on, MIDI key velocity: from 0 (no sound) to 127 (loudest). For continuous controller messages, controller value.

Invoke fluidControl in instrument definitions that actually play notes and send control messages. Each instrument definition must consistently use one MIDI channel that was assigned to a Fluidsynth program using fluidLoad.

In this implementation, SoundFont effects such as chorus or reverb are used if and only if they are defaults for the preset. There is no means of turning such effects on or off, or of changing their parameters, from Csound.

See Also

fluidEngine, fluidNote, fluidLoad

Credits

Opcodes by Michael Gogins (gogins at pipeline dot com). Thanks to Peter Hanappe for Fluidsynth, and to Steven Yi for seeing that it is necessary to break up the Fluidsynth into several different Csound opcodes.

New in Csound5.00

fluidEngine

fluidEngine — Instantiates a fluidsynth engine.

Syntax

```
ienginenum fluidEngine [iReverbEnabled] [, iChorusEnabled] [,iNumChannels] [, iPolyphony]
```

Description

Instantiates a fluidsynth engine, and returns *ienginenum* to identify the engine. *ienginenum* is passed to other other opcodes for loading and playing SoundFonts and gathering the generated sound.

Initialization

ienginenum -- engine number assigned from fluidEngine.

iReverbEnabled -- optionally set to 0 to disable any reverb effect in the loaded SoundFonts.

iChorusEnabled -- optionally set to 0 to disable any chorus effect in the loaded SoundFonts.

iNumChannels -- number of channels to use; range is 16-256 and Csound default is 256 (Fluidsynth's default is 16).

iPolyphony -- number of voices to be played in parallel; range is 16-4096 and Csound default is 4096 (Fluidsynth's default is 256). Note: this is not the number of notes played at the same time as a single note may use create multiple voices depending on instrument zones and velocity/key of played note.

Examples

Here is an example of the fluidsynth opcodes. It uses the file *fluidAllOut.orc* [examples/fluidAllOut.orc].

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
odbfs = 32767

; LOAD SOUND FONTS
gienginenum1 fluidEngine
gienginenum2 fluidEngine
isfnum1 fluidLoad "Piano Steinway Grand Model C (21,738KB).sf2", gienginenum1, 1
; Bright Steinway, program 1, channel 1
fluidProgramSelect gienginenum1, 1, isfnum1, 0, 1
; Concert Steinway with reverb, program 2, channel 3
fluidProgramSelect gienginenum1, 3, isfnum1, 0, 2
isfnum2 fluidLoad "63.3mg The Sound Site Album Bank V1.0.SF2", gienginenum2, 1
; General MIDI, program 50, channel 2
fluidProgramSelect gienginenum2, 2, isfnum2, 0, 50

; SEND NOTES TO STEINWAY SOUND FONT

instr 1 ; FluidSynth Steinway Rev
; INITIALIZATION
mididefault 60, p3 ; Default duration of 60 -- overridden by score.
midinoteonkey p4, p5 ; Channels MIDI input to pfields.
; Use channel assigned in fluidload.
ichannel = 1
ikey = p4
ivelocity = p5
istatus = 144
fluidControl gienginenum1, istatus, ichannel, ikey, ivelocity
endin
```

```
instr 2 ; GM soundfont
; INITIALIZATION
    mididefault 60, p3 ; Default duration of 60 -- overridden by score.
    midinoteonkey p4, p5 ; Channels MIDI input to pfields.
; Use channel assigned in fluidload.
ichannel = 2
ikey = p4
ivelocity = p5
istatus = 144
    fluidNote gienginenum2, ichannel, ikey, ivelocity
endin

instr 3 ; FluidSynth Steinway Rev
; INITIALIZATION
    mididefault 60, p3 ; Default duration of 60 -- overridden by score.
    midinoteonkey p4, p5 ; Channels MIDI input to pfields.
; Use channel assigned in fluidload.
ichannel = 3
ikey = p4
ivelocity = p5
istatus = 144
    fluidNote gienginenum1, ichannel, ikey, ivelocity
endin

; COLLECT AUDIO FROM ALL SOUND FONTS

instr 100 ; Fluidsynth output
; INITIALIZATION
; Normalize so iamplitude for p5 of 80 == ampdb(80).
iamplitude = ampdb(p5) * (10000.0 / 0.1)
; AUDIO
aleft, aright fluidAllOut
    outs aleft * iamplitude, aright * iamplitude
endin
```

Here is another example of the fluidsynth opcodes using 2 engines. It uses the file *fluid-2.orc* [examples/fluid-2.orc].

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
odbfs = 32767

; LOAD SOUND FONTS
gienginenum1 fluidEngine
gienginenum2 fluidEngine
isfnum1 fluidLoad "Piano Steinway Grand Model C (21,738KB).sf2", gienginenum1, 1
; Bright Steinway, program 1, channel 1
    fluidProgramSelect gienginenum1, 1, isfnum1, 0, 1
; Concert Steinway with reverb, program 2, channel 3
    fluidProgramSelect gienginenum1, 3, isfnum1, 0, 2
isfnum2 fluidLoad "63.3mg The Sound Site Album Bank V1.0.SF2", gienginenum2, 1
; General MIDI, program 50, channel 2
    fluidProgramSelect gienginenum2, 2, isfnum2, 0, 50

; SEND NOTES TO STEINWAY SOUND FONT

instr 1 ; FluidSynth Steinway Rev
; INITIALIZATION
    mididefault 60, p3 ; Default duration of 60 -- overridden by score.
    midinoteonkey p4, p5 ; Channels MIDI input to pfields.
; Use channel assigned in fluidload.
ichannel = 1
ikey = p4
ivelocity = p5
    fluidNote gienginenum1, ichannel, ikey, ivelocity
endin

instr 2 ; GM soundfont
; INITIALIZATION
    mididefault 60, p3 ; Default duration of 60 -- overridden by score.
    midinoteonkey p4, p5 ; Channels MIDI input to pfields.
; Use channel assigned in fluidload.
ichannel = 2
ikey = p4
ivelocity = p5
    fluidNote gienginenum2, ichannel, ikey, ivelocity
endin

instr 3 ; FluidSynth Steinway Rev
; INITIALIZATION
    mididefault 60, p3 ; Default duration of 60 -- overridden by score.
    midinoteonkey p4, p5 ; Channels MIDI input to pfields.
```

```
; Use channel assigned in fluidload.
ichannel = 3
ikey = p4
ivelocity = p5
fluidNote gienginenum1, ichannel, ikey, ivelocity
endin

; COLLECT AUDIO FROM ALL SOUNDFONTS

instr 100 ; Fluidsynth output
; INITIALIZATION
; Normalize so iamplitude for p5 of 80 == ampdb(80).
iamplitude1 = ampdb(p5) * (10000.0 / 0.1)
iamplitude2 = ampdb(p6) * (10000.0 / 0.1)

; AUDIO
aleft1, aright1 fluidOut gienginenum1
aleft2, aright2 fluidOut gienginenum2
outs (aleft1 * iamplitude1) + (aleft2 * iamplitude2), \
(aright1 * iamplitude1) + (aright2 * iamplitude2)
endin
```

Here is another more complex example of the fluidsynth opcodes written by Istvan Varga. It uses the file *fluidcomplex.csd* [examples/fluidcomplex.csd].

```
<CsoundSynthesizer>
<CsOptions>
-d -m229 -o dac -T -F midifile.mid
</CsOptions>
<CsInstruments>
sr = 44100
ksmps = 128
nchnls = 2
0dbfs = 1

; Example by Istvan Varga

; disable triggering of instruments by MIDI events

ichn = 1
lp1:
    massign ichn, 0
    loop_le ichn, 1, 16, lp1
    pgmassign 0, 0

; initialize FluidSynth

gifld fluidEngine
gisf2 fluidLoad "07AcousticGuitar.sf2", gifld, 1

; k-rate version of fluidProgramSelect

opcode fluidProgramSelect_k, 0, kkkkk
keng, kchn, ksf2, kbnk, kpre xin
igoto skipInit
doInit:
    fluidProgramSelect i(keng), i(kchn), i(ksf2), i(kbnk), i(kpre)
    reinit doInit
    rireturn
skipInit:
endop

instr 1
; initialize channels
kchn init 1
if (kchn == 1) then
lp2:
    fluidControl gifld, 192, kchn - 1, 0, 0
    fluidControl gifld, 176, kchn - 1, 7, 100
    fluidControl gifld, 176, kchn - 1, 10, 64
    loop_le kchn, 1, 16, lp2
endif

; send any MIDI events received to FluidSynth
nxt:
kst, kch, kd1, kd2 midiin
if (kst != 0) then
    if (kst != 192) then
        fluidControl gifld, kst, kch - 1, kd1, kd2
    else
        fluidProgramSelect_k gifld, kch - 1, gisf2, 0, kd1
    endif
    kgoto nxt
endif
```

```
; get audio output from FluidSynth
aL, aR fluidOut gifld
      outs      aL, aR
endin

</CsInstruments>
<CsScore>

i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

fluidNote, fluidLoad

Credits

Michael Gogins (gogins at pipeline dot com), Steven Yi. Thanks to Peter Hanappe for Fluidsynth.

Optional *iNumChannels* and *iPolyphony* parameters added in 5.07

fluidLoad

fluidLoad — Loads a SoundFont into a fluidEngine, optionally listing SoundFont contents.

Syntax

```
isfnum fluidLoad soundfont, ienginenum[, ilistpresets]
```

Description

Loads a SoundFont into an instance of a fluidEngine, optionally listing banks and presets for SoundFont.

Initialization

isfnum -- number assigned to just-loaded soundfont.

soundfont -- string specifying a SoundFont filename. Note that any number of SoundFonts may be loaded (obviously, by different invocations of fluidLoad).

ienginenum -- engine number assigned from fluidEngine

ilistpresets -- optional, if specified, lists all Fluidsynth programs for the just-loaded SoundFont. A Fluidsynth program is a combination of SoundFont ID, bank number, and preset number that is assigned to a MIDI channel.

Performance

Invoke fluidLoad in the orchestra header, any number of times. The same SoundFont may be invoked to assign programs to MIDI channels any number of times; the SoundFont is only loaded the first time.

Examples

Here is an example of the fluidsynth opcodes. It uses the file *fluid.orc* [examples/fluid.orc].

```
sr = 44100
ksmps = 100
nchnls = 2

giengine fluidEngine
isfnum fluidLoad "07AcousticGuitar.sf2", giengine, 1
fluidProgramSelect giengine, 1, isfnum, 0, 0

instr 1
    mididefault 60, p3
    midinoteonkey p4, p5

    ikey init p4
    ivel init p5

    fluidNote giengine, 1, ikey, ivel
endin

instr 99
    imvol init 70000
    asigl, asigr fluidOut giengine
    outs asigl * imvol, asigr * imvol
endin
```

See *fluidEngine* for more examples.

See Also

fluidEngine, fluidNote

Credits

Michael Gogins (gogins at pipeline dot com), Steven Yi. Thanks to Peter Hanappe for Fluidsynth.

New in Csound5.00

fluidNote

fluidNote — Plays a note on a channel in a fluidSynth engine.

Syntax

```
fluidNote ienginenum, ichannelnum, imidikey, imidivel
```

Description

Plays a note at *imidikey* pitch and *imidivel* velocity on *ichannelnum* channel of number *ienginenum* fluidEngine.

Initialization

ienginenum -- engine number assigned from fluidEngine

ichannelnum -- which channel number to play a note on in the given fluidEngine

imidikey -- MIDI key for note (0-127)

imidivel -- MIDI velocity for note (0-127)

Examples

Here is an example of the fluidsynth opcodes. It uses the file *fluid.orc* [examples/fluid.orc].

```
sr = 44100
ksmps = 100
nchnls = 2

giengine fluidEngine
isfnum fluidLoad "07AcousticGuitar.sf2", giengine, 1
fluidProgramSelect giengine, 1, isfnum, 0, 0

instr 1
    mididefault 60, p3
    midinoteonkey p4, p5

    ikey init p4
    ivel init p5

    fluidNote giengine, 1, ikey, ivel
endin

instr 99
    imvol init 70000
    asigl, asigr fluidOut giengine
    outs asigl * imvol, asigr * imvol
endin
```

See *fluidEngine* for more examples.

See Also

fluidEngine, *fluidLoad*

Credits

Michael Gogins (gogins at pipeline dot com), Steven Yi. Thanks to Peter Hanappe for Fluidsynth.

fluidOut

fluidOut — Outputs sound from a given fluidEngine

Syntax

```
aleft, aright fluidOut ienginenum
```

Description

Outputs the sound from a fluidEngine.

Initialization

ienginenum -- engine number assigned from fluidEngine

Performance

aleft -- Left channel audio output.

aright -- Right channel audio output.

Invoke fluidOut in an instrument definition numbered higher than any fluidcontrol instrument definitions. All SoundFonts used in the fluidEngine numbered *ienginenum* send their audio output to this one opcode. Send a note with an indefinite duration to this instrument to turn the SoundFonts on for as long as required.

Examples

Here is an example of the fluidsynth opcodes. It uses the file *fluid.orc* [examples/fluid.orc].

```
sr = 44100
ksmps = 100
nchnls = 2

giengine fluidEngine
isfnum fluidLoad "07AcousticGuitar.sf2", giengine, 1
fluidProgramSelect giengine, 1, isfnum, 0, 0

instr 1
    mididefault 60, p3
    midinoteonkey p4, p5

    ikey init p4
    ivel init p5

    fluidNote giengine, 1, ikey, ivel
endin

instr 99
    imvol init 70000
    asigl, asigr fluidOut giengine
    outs asigl * imvol, asigr * imvol
endin
```

See *fluidEngine* for more examples.

See Also

fluidEngine, fluidNote, fluidLoad

Credits

Michael Gogins (gogins at pipeline dot com), Steven Yi. Thanks to Peter Hanappe for Fluidsynth.

New in Csound5.00

fluidProgramSelect

fluidProgramSelect — Assigns a preset from a SoundFont to a channel on a fluidEngine.

Syntax

```
fluidProgramSelect iengineum, ichannelnum, isfnum, ibanknum, ipresetnum
```

Description

Assigns a preset from a SoundFont to a channel on a fluidEngine.

Initialization

iengineum -- engine number assigned from fluidEngine

ichannelnum -- which channel number to use for the preset in the given fluidEngine

isfnum -- number of the SoundFont from which the preset is assigned

ibanknum -- number of the bank in the SoundFont from which the preset is assigned

ipresetnum -- number of the preset to assign

Examples

Here is an example of the fluidsynth opcodes. It uses the file *fluid.orc* [examples/fluid.orc].

```
sr = 44100
ksmps = 100
nchnls = 2

giengine fluidEngine
isfnum fluidLoad "07AcousticGuitar.sf2", giengine, 1
fluidProgramSelect giengine, 1, isfnum, 0, 0

instr 1
    mididefault 60, p3
    midinoteonkey p4, p5

    ikey init p4
    ivel init p5

    fluidNote giengine, 1, ikey, ivel
endin

instr 99
    imvol init 70000
    asigl, asigr fluidOut giengine
    outs asigl * imvol, asigr * imvol
endin
```

See *fluidEngine* for more examples.

See Also

fluidEngine, *fluidNote*, *fluidLoad*

Credits

Michael Gogins (gogins at pipeline dot com), Steven Yi. Thanks to Peter Hanappe for Fluidsynth.

fluidSetInterpMethod

fluidSetInterpMethod — Set interpolation method for channel in Fluid Engine

Syntax

```
fluidSetInterpMethod ienginenum, ichannelnum, iInterpMethod
```

Description

Set interpolation method for channel in Fluid Engine. Lower order interpolation methods will render faster at lower fidelity while higher order interpolation methods will render slower at higher fidelity. Default interpolation for a channel is 4th order interpolation.

Initialization

ienginenum -- engine number assigned from fluidEngine

ichannelnum -- which channel number to use for the preset in the given fluidEngine

iInterpMethod -- interpolation method, can be any of the following

- 0 -- No Interpolation
- 1 -- Linear Interpolation
- 4 -- 4th Order Interpolation (Default)
- 7 -- 7th Order Interpolation (Highest)

See Also

fluidEngine

Credits

Author: Steven Yi

New in version 5.07

FLvalue

FLvalue — Shows the current value of a FLTK valuator.

Description

FLvalue shows current the value of a valuator in a text field.

Syntax

```
ihandle FLvalue "label", iwidth, iheight, ix, iy
```

Initialization

ihandle -- handle value (an integer number) that unequivocally references the corresponding valuator. It can be used for the *idisp* argument of a valuator.

"label" -- a double-quoted string containing some user-provided text, placed near the corresponding widget.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the valuator, relative to the upper left corner of corresponding window (expressed in pixels).

Performance

FLvalue shows the current values of a valuator in a text field. It outputs *ihandle* that can then be used for the *idisp* argument of a valuator (see the *FLTK Valuators section*). In this way, the values of that valuator will be dynamically be shown in a text field.



Note

Note that *FLvalue* is not a valuator and its value cannot be modified. The value for an *FLvalue* widget should be set only by other widgets, and NOT from *FLsetVal* or *FLsetVal_i* since this can cause Csound to crash.

Examples

Here is an example of the FLvalue opcode. It uses the file *FLvalue.csd* [examples/FLvalue.csd].

Example 179. Example of the FLvalue opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
```

```
; Audio out   Audio in   No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o FLvalue.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Using the opcode flvalue to display the output of a slider
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

FLpanel "Value Display Box", 900, 200, 50, 50
; Width of the value display box in pixels
iwidth = 50
; Height of the value display box in pixels
iheight = 20
; Distance of the left edge of the value display
; box from the left edge of the panel
ix = 65
; Distance of the top edge of the value display
; box from the top edge of the panel
iy = 55

idisp FLvalue "Hertz", iwidth, iheight, ix, iy
gkfreq, ihandle FLslider "Frequency", 200, 5000, -1, 5, idisp, 750, 30, 125, 50
FLsetVal_i 500, ihandle
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
iamp = 15000
ifn = 1
asig oscili iamp, gkfreq, ifn
out asig
endin

</CsInstruments>
<CsScore>

; Function table that defines a single cycle
; of a sine wave.
f 1 0 1024 10 1

; Instrument 1 will play a note for 1 hour.
i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLbox, FLbutBank, FLbutton, FLprintk, FLprintk2

Credits

Author: Gabriel Maldonado

New in version 4.22

Example written by Iain McCurdy, edited by Kevin Conder.

FLvkeybd

FLvkeybd — An FLTK widget opcode that creates a virtual keyboard widget.

Description

An FLTK widget opcode that creates a virtual keyboard widget. This must be used in conjunction with the virtual midi keyboard driver for this to operate correctly. The purpose of this opcode is for making demo versions of MIDI orchestras with the virtual keyboard embedded within the main window.



Note

The widget version of the virtual keyboard does not include the MIDI sliders found in the full window version of the virtual keyboard.

Syntax

```
FLvkeybd "keyboard.map", iwidth, iheight, ix, iy
```

Initialization

"keyboard.map" -- a double-quoted string containing the keyboard map to use. An empty string ("") may be used to use the default bank/channel name values. See Virtual Midi Keyboard for more information on keyboard mappings.

iwidth -- width of widget.

iheight -- height of widget.

ix -- horizontal position of upper left corner of the keyboard, relative to the upper left corner of corresponding window (expressed in pixels).

iy -- vertical position of upper left corner of the keyboard, relative to the upper left corner of corresponding window (expressed in pixels).



Note

The standard width and height for the virtual keyboard is 624x120 for the dialog version that is shown when FLvkeybd is not used.

See Also

FLbutton, *FLbox*, *FLbutBank*, *FLprintk*, *FLprintk2*, *FLvalue*

Credits

Author: Steven Yi

New in version 5.05

FLvslidBnk

FLvslidBnk — A FLTK widget containing a bank of horizontal sliders.

Description

FLvslidBnk is a widget containing a bank of horizontal sliders.

Syntax

```
FLvslidBnk "names", inumsliders [, ioutable] [, iwidth] [, iheight] [, ix] \  
[, iy] [, itypetable] [, iexptable] [, istart_index] [, iminmaxtable]
```

Initialization

"names" -- a double-quoted string containing the names of each slider. Each slider can have a different name. Separate each name with "@" character, for example: "frequency@amplitude@cutoff". It is possible to not provide any name by giving a single space ". In this case, the opcode will automatically assign a progressive number as a label for each slider.

inumsliders -- the number of sliders.

ioutable (optional, default=0) -- number of a previously-allocated table in which to store output values of each slider. The user must be sure that table size is large enough to contain all output cells, otherwise a segfault will crash Csound. By assigning zero to this argument, the output will be directed to the zak space in the k-rate zone. In this case, the zak space must be previously allocated with the *zakin* opcode and the user must be sure that the allocation size is big enough to cover all sliders. The default value is zero (i.e. store output in zak space).

istart_index (optional, default=0) -- an integer number referring to a starting offset of output cell locations. It can be positive to allow multiple banks of sliders to output in the same table or in the zak space. The default value is zero (no offset).

iminmaxtable (optional, default=0) -- number of a previously-defined table containing a list of min-max pairs, referred to each slider. A zero value defaults to the 0 to 1 range for all sliders without necessity to provide a table. The default value is zero.

iexptable (optional, default=0) -- number of a previously-defined table containing a list of identifiers (i.e. integer numbers) provided to modify the behaviour of each slider independently. Identifiers can assume the following values:

- -1 -- exponential curve response
- 0 -- linear response
- number > than 0 -- follow the curve of a previously-defined table to shape the response of the corresponding slider. In this case, the number corresponds to table number.

You can assume that all sliders of the bank have the same response curve (exponential or linear). In this case, you can assign -1 or 0 to *iexptable* without worrying about previously defining any table. The default value is zero (all sliders have a linear response, without having to provide a table).

ityetable (optional, default=0) -- number of a previously-defined table containing a list of identifiers (i.e. integer numbers) provided to modify the aspect of each individual slider independently. Identifiers can assume the following values:

- 0 = Nice slider

- 1 = Fill slider
- 3 = Normal slider
- 5 = Nice slider
- 7 = Nice slider with down-box

You can assume that all sliders of the bank have the same aspect. In this case, you can assign a negative number to *ityetable* without worrying about previously defining any table. Negative numbers have the same meaning of the corresponding positive identifiers with the difference that the same aspect is assigned to all sliders. You can also assign a random aspect to each slider by setting *ityetable* to a negative number lower than -7. The default value is zero (all sliders have the aspect of nice sliders, without having to provide a table).

You can add 20 to a value inside the table to make the slider "plastic", or subtract 20 if you want to set the value for all widgets without defining a table (e.g. -21 to set all sliders types to Plastic Fill slider).

iwidth (optional) -- width of the rectangular area containing all sliders of the bank, excluding text labels, that are placed to the left of that area.

iheight (optional) -- height of the rectangular area containing all sliders of the bank, excluding text labels, that are placed to the left of that area.

ix (optional) -- horizontal position of the upper left corner of the rectangular area containing all sliders belonging to the bank. You have to leave enough space, at the left of that rectangle, in order to make sure labels of sliders to be visible. This is because the labels themselves are external to the rectangular area.

iy (optional) -- vertical position of the upper left corner of the rectangular area containing all sliders belonging to the bank. You have to leave enough space, at the left of that rectangle, in order to make sure labels of sliders to be visible. This is because the labels themselves are external to the rectangular area.

Performance

There are no k-rate arguments, even if cells of the output table (or the zak space) are updated at k-rate.

FLvslidBnk is a widget containing a bank of vertical sliders. Any number of sliders can be placed into the bank (*inumslders* argument). The output of all sliders is stored into a previously allocated table or into the zak space (*ioutable* argument). It is possible to determine the first location of the table (or of the zak space) in which to store the output of the first slider by means of *istart_index* argument.

Each slider can have an individual label that is placed below it. Labels are defined by the "names" argument. The output range of each slider can be individually set by means of an external table (*iminmaxtable* argument). The curve response of each slider can be set individually, by means of a list of identifiers placed in a table (*ixptable* argument). It is possible to define the aspect of each slider independently or to make all sliders have the same aspect (*ityetable* argument).

The *iwidth*, *iheight*, *ix*, and *iy* arguments determine width, height, horizontal and vertical position of the rectangular area containing sliders. Notice that the label of each slider is placed below them and is not included in the rectangular area containing sliders. So the user should leave enough space below the bank by assigning a proper *ix* value in order to leave labels visible.

FLvslidBnk is identical to *FLslidBnk* except it contains vertical sliders instead of horizontal. Since the width of each single slider is often small, it is recommended to leave only a single space in the names string (" "), in this case each slider will be automatically numbered.



IMPORTANT!

Notice that the tables used by *FLvslidBnk* must be created with the *ftgen* opcode and placed in the orchestra before the corresponding valuator. They can not be placed in the score. This is because tables placed in the score are created later than the initialization of the opcodes placed in the header section of the orchestra.

Examples

Here is an example of the *FLvslidBnk* opcode. It uses the file *FLvslidBnk.csd* [examples/*FLvslidBnk.csd*].

Example 180. Example of the *FLvslidBnk* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr = 44100
kr = 441
ksmps = 100
nchnls = 1

gitypetable ftgen 0, 0, 8, -2, 1, 1, 3, 3, 5, 5, 7, 7
giouttable ftgen 0, 0, 8, -2, 0, 0.2, 0.3, 0.4, 0.5, 0.6, 0.8, 1

FLpanel "Slider Bank", 400, 400, 50, 50
;Number of sliders
inum = 8
; Table to store output
iouttable = giouttable
; Width of the slider bank in pixels
iwidth = 350
; Height of the slider in pixels
iheight = 160
; Distance of the left edge of the slider
; from the left edge of the panel
ix = 30
; Distance of the top edge of the slider
; from the top edge of the panel
iy = 10
; Table containing fader types
itypetable = gitypetable
FLvslidBnk "1@2@3@4@5@6@7@8@9@10@11@12@13@14@15@16", 16 , iouttable , iwidth , iheight , ix \
, iy , itypetable
FLvslidBnk " ", inum , iouttable , iwidth , iheight , ix \
, iy + 200 , -23
; End of panel contents
FLpanelEnd
; Run the widget thread!
FLrun

instr 1
;Dummy instrument
endin

</CsInstruments>
<CsScore>

; Instrument 1 will play a note for 1 hour.
i 1 0 3600
e

</CsScore>
</CsoundSynthesizer>
```

See Also

FLslider, FLslidBnk

Credits

Author: Gabriel Maldonado

New in version 5.06

FLvslidBnk2

FLvslidBnk2 — A FLTK widget containing a bank of horizontal sliders.

Description

FLvslidBnk2 is a widget containing a bank of horizontal sliders.

Syntax

```
FLvslidBnk2 "names", inumsliders, ioutable, iconfigtable [,iwidth, iheight, ix, iy, istart_index]
```

Initialization

"names" -- a double-quoted string containing the names of each slider. Each slider can have a different name. Separate each name with "@" character, for example: "frequency@amplitude@cutoff". It is possible to not provide any name by giving a single space ". In this case, the opcode will automatically assign a progressive number as a label for each slider.

inumsliders -- the number of sliders.

ioutable (optional, default=0) -- number of a previously-allocated table in which to store output values of each slider. The user must be sure that table size is large enough to contain all output cells, otherwise a segfault will crash Csound. By assigning zero to this argument, the output will be directed to the zak space in the k-rate zone. In this case, the zak space must be previously allocated with the *zakinit* opcode and the user must be sure that the allocation size is big enough to cover all sliders. The default value is zero (i.e. store output in zak space).

iconfigtable -- in the *FLslidBnk2* and *FLvslidBnk2* opcodes, this table replaces *iminmaxtable*, *iexpstable* and *istyletable*, all these parameters being placed into a single table. This table has to be filled with a group of 5 parameters for each slider in this way:

min1, max1, exp1, style1, min2, max2, exp2, style2, min3, max3, exp3, style3 etc.

for example using GEN02 you can type:

```
inum ftgen 1,0,256, -2, 0,1,0,1, 100, 5000, -1, 3, 50, 200, -1, 5,..... [etcetera]
```

In this example the first slider will be affected by the [0,1,0,1] parameters (the range will be 0 to 1, it will have linear response, and its aspect will be a fill slider), the second slider will be affected by the [100,5000,-1,3] parameters (the range is 100 to 5000, the response is exponential and the aspect is a normal slider), the third slider will be affected by the [50,200,-1,5] parameters (the range is 50 to 200, the behavior exponential, and the aspect is a nice slider), and so on.

iwidth (optional) -- width of the rectangular area containing all sliders of the bank, excluding text labels, that are placed to the left of that area.

iheight (optional) -- height of the rectangular area containing all sliders of the bank, excluding text labels, that are placed to the left of that area.

ix (optional) -- horizontal position of the upper left corner of the rectangular area containing all sliders belonging to the bank. You have to leave enough space, at the left of that rectangle, in order to make sure labels of sliders to be visible. This is because the labels themselves are external to the rectangular area.

iy (optional) -- vertical position of the upper left corner of the rectangular area containing all sliders belonging to the bank. You have to leave enough space, at the left of that rectangle, in order to make

sure labels of sliders to be visible. This is because the labels themselves are external to the rectangular area.

istart_index (optional, default=0) -- an integer number referring to a starting offset of output cell locations. It can be positive to allow multiple banks of sliders to output in the same table or in the zak space. The default value is zero (no offset).

Performance

There are no k-rate arguments, even if cells of the output table (or the zak space) are updated at k-rate.

FLvslidBnk2 is a widget containing a bank of vertical sliders. Any number of sliders can be placed into the bank (*inumsliders* argument). The output of all sliders is stored into a previously allocated table or into the zak space (*ioutable* argument). It is possible to determine the first location of the table (or of the zak space) in which to store the output of the first slider by means of *istart_index* argument.

Each slider can have an individual label that is placed to the left of it. Labels are defined by the “*names*” argument. The output range of each slider can be individually set by means of the *min* and *max* values inside the *iconfigtable* table. The curve response of each slider can be set individually, by means of a list of identifiers placed in the *iconfigtable* table (*exp* argument). It is possible to define the aspect of each slider independently or to make all sliders have the same aspect (*style* argument in the *iconfigtable* table).

The *iwidth*, *iheight*, *ix*, and *iy* arguments determine width, height, horizontal and vertical position of the rectangular area containing sliders. Notice that the label of each slider is placed below them and is not included in the rectangular area containing sliders. So the user should leave enough space below the bank by assigning a proper *ix* value in order to leave labels visible.

FLvslidBnk2 is identical to *FLslidBnk2* except it contains vertical sliders instead of horizontal. Since the width of each single slider is often small, it is recommended to leave only a single space in the names string (“ ”), in this case each slider will be automatically numbered.



IMPORTANT!

Notice that the tables used by *FLvslidBnk2* must be created with the *ftgen* opcode and placed in the orchestra before the corresponding valuator. They can not be placed in the score. This is because tables placed in the score are created later than the initialization of the opcodes placed in the header section of the orchestra.

See Also

FLslider, *FLslidBnk*, *FLslidBnk2*, *FLvslidBnk2*

Credits

Author: Gabriel Maldonado

New in version 5.06

FLxyin

FLxyin — Senses the mouse cursor position in a user-defined area inside an FLpanel.

Description

Similar to *xyin*, sense the mouse cursor position in a user-defined area inside an FLpanel.

Syntax

```
koutx, kouty, kinside FLxyin ioutx_min, ioutx_max, iouty_min, iouty_max, \  
iwindx_min, iwindx_max, iwindy_min, iwindy_max [, iexpx, iexpy, ioutx, iouty]
```

Initialization

ioutx_min, ioutx_max - the minimum and maximum limits of the interval to be output (X or horizontal axis).

iouty_min, iouty_max - the minimum and maximum limits of the interval to be output (Y or vertical axis).

iwindx_min, iwindx_max - the X coordinate of the horizontal edges of the sensible area, relative to the FLpanel, in pixels.

iwindy_min, iwindy_max - the Y coordinates of the vertical edges of the sensible area, relative to the FLpanel, in pixels.

iexpx, iexpy - (optional) integer numbers denoting the behavior of the x or y output: 0 -> output is linear; -1 -> output is exponential; any other number indicates the number of an existing table that is used for indexing. With a positive value for table number, linear interpolation is provided in table indexing. A negative table number suppresses interpolation. Notice that in normal operations, the table should be normalized and unipolar (i.e. all table elements should be in the zero to one range). In this case all table elements will be rescaled according to imax and imin. Anyway, it is possible to use non-normalized tables (created with a negative table number, that can contain elements of any value), in order to access the actual values of table elements, without rescaling, by assigning 0 to iout_min and 1 to iout_max.

ioutx, iouty – (optional) initial output values.

Performance

koutx, kouty - output values, scaled according to user choices.

kinside - a flag that informs if the mouse cursor falls out of the rectangle of the user-defined area. If it is out of the area, kinside is set to zero.

FLxyin senses the mouse cursor position in a user-defined area inside an *FLpanel*. When *FLxyin* is called, the position of the mouse within the chosen area is returned at k-rate. It is possible to define the sensible area, as well the minimum and maximum values corresponding to the minimum and maximum mouse positions. Mouse buttons don't need to be pressed to make *FLxyin* to operate. It is able to function correctly even if other widgets (present in the *FLpanel*) overlap the sensible area.

FLxyin unlike most other FLTK opcodes can't be used inside the header, since it is not a widget. It is just a definition of an area for mouse sensing within an FLTK panel.

Examples

Here is an example of the FLxyin opcode. It uses the file *FLxyin.csd* [examples/FLxyin.csd].

Example 181. Example of the FLxyin opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr=48000
ksmps=128
nchnls=2

; Example by Andres Cabrera 2007

FLpanel "FLxyin", 200, 100, -1, -1, 3
FLpanelEnd
FLrun

instr 1
  koutx, kouty, kinside FLxyin 0, 10, 100, 1000, 10, 190, 10, 90
  aout buzz 10000, kouty, koutx, 1
  printk2 koutx
  outs aout, aout
endin

</CsInstruments>
<CsScore>
f 1 0 1024 10 1
i 1 0 3600

e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the FLxyin opcode. It uses the file *FLxyin-2.csd* [examples/FLxyin-2.csd].

Example 182. Example of the FLxyin opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr=44100
kr=441
ksmps=100
nchnls=2

; Example by Gabriel Maldonado

FLpanel "Move the mouse inside this panel to hear the effect",400,400
FLpanel_end
FLrun

instr 1

k1, k2, kinside FLxyin 50, 1000, 50, 1000, 100, 300, 50, 250, -2,-3
; if k1 <= 50 || k1 >=5000 || k2 <=100 || k2 >= 8000 kgoto end ; if cursor is outside bounds, then d
a1 oscili 3000, k1, 1
```

```
a2 oscili 3000, k2, 1

      outs a1,a2
printk2 k1
printk2 k2, 10
printk2 kinside, 20
end:      endin

</CsInstruments>
<CsScore>

f1 0 1024 10 1
f2 0 17 19 1 1 90 1
f3 0 17 19 2 1 90 1
i1 0 3600

</CsScore>
</CsoundSynthesizer>
```

See Also

FLpanel

Credits

Author: Gabriel Maldonado

New in version 5.06

fmb3

fmb3 — Uses FM synthesis to create a Hammond B3 organ sound.

Description

Uses FM synthesis to create a Hammond B3 organ sound. It comes from a family of FM sounds, all using 4 basic oscillators and various architectures, as used in the TX81Z synthesizer.

Syntax

```
ares fmb3 kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, \  
      ifn4, ivfn
```

Initialization

fmb3 takes 5 tables for initialization. The first 4 are the basic inputs and the last is the low frequency oscillator (LFO) used for vibrato. The last table should usually be a sine wave.

The initial waves should be:

- *ifn1* -- sine wave
- *ifn2* -- sine wave
- *ifn3* -- sine wave
- *ifn4* -- sine wave

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kc1, *kc2* -- Controls for the synthesizer:

- *kc1* -- Total mod index
- *kc2* -- Crossfade of two modulators
- *Algorithm* -- 4

kvdepth -- Vibrator depth

kvrate -- Vibrator rate

Examples

Here is an example of the fmb3 opcode. It uses the file *fmb3.csd* [examples/fmb3.csd].

Example 183. Example of the fmb3 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fmb3.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 15000
  kfreq = 440
  kc1 = 5
  kc2 = 5
  kvdepth = 0.005
  kvrate = 6
  ifn1 = 1
  ifn2 = 1
  ifn3 = 1
  ifn4 = 1
  ivfn = 1

  al fmb3 kamp, kfreq, kc1, kc2, kvdepth, kvrate, \
    ifn1, ifn2, ifn3, ifn4, ivfn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

fmbell, fmmetal, fmpercfl, fmrhode, fmwurlie

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.47

fmbell

fmbell — Uses FM synthesis to create a tublar bell sound.

Description

Uses FM synthesis to create a tublar bell sound. It comes from a family of FM sounds, all using 4 basic oscillators and various architectures, as used in the TX81Z synthesizer.

Syntax

```
ares fmbell kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, \  
      ifn4, ivfn
```

Initialization

All these opcodes take 5 tables for initialization. The first 4 are the basic inputs and the last is the low frequency oscillator (LFO) used for vibrato. The last table should usually be a sine wave.

The initial waves should be:

- *ifn1* -- sine wave
- *ifn2* -- sine wave
- *ifn3* -- sine wave
- *ifn4* -- sine wave

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kc1, *kc2* -- Controls for the synthesizer:

- *kc1* -- Mod index 1
- *kc2* -- Crossfade of two outputs
- *Algorithm* -- 5

kvdepth -- Vibrator depth

kvrate -- Vibrator rate

Examples

Here is an example of the fmbell opcode. It uses the file *fmbell.csd* [examples/fmbell.csd].

Example 184. Example of the fmbell opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fmbell.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 10000
  kfreq = 880
  kc1 = 5
  kc2 = 5
  kvdepth = 0.005
  kvrate = 6
  ifn1 = 1
  ifn2 = 1
  ifn3 = 1
  ifn4 = 1
  ivfn = 1

  al fmbell kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, ifn4, ivfn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

fmb3, fmmetal, fmpercfl, fmrhode, fmwurlie

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.47

fmmetal

fmmetal — Uses FM synthesis to create a “Heavy Metal” sound.

Description

Uses FM synthesis to create a “Heavy Metal” sound. It comes from a family of FM sounds, all using 4 basic oscillators and various architectures, as used in the TX81Z synthesizer.

Syntax

```
ares fmmetal kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, \  
      ifn4, ivfn
```

Initialization

All these opcodes take 5 tables for initialization. The first 4 are the basic inputs and the last is the low frequency oscillator (LFO) used for vibrato. The last table should usually be a sine wave.

The initial waves should be:

- *ifn1* -- sine wave
- *ifn2* -- *twopeaks.aiff* [examples/twopeaks.aiff]
- *ifn3* -- *twopeaks.aiff* [examples/twopeaks.aiff]
- *ifn4* -- sine wave



Note

The file “twopeaks.aiff” is also available at <ftp://ftp.cs.bath.ac.uk/pub/dream/documentation/sounds/modelling/>.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kc1, *kc2* -- Controls for the synthesizer:

- *kc1* -- Total mod index
- *kc2* -- Crossfade of two modulators
- *Algorithm* -- 3

kvdepth -- Vibrator depth

kvrate -- Vibrator rate

Examples

Here is an example of the `fmmetal` opcode. It uses the file `fmmetal.csd` [examples/fmmetal.csd], and `twopeaks.aiff` [examples/twopeaks.aiff].

Example 185. Example of the `fmmetal` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fmmetal.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 10000
  kfreq = 440
  kc1 = 6
  kc2 = 5
  kvdepth = 0
  kvrate = 0
  ifn1 = 1
  ifn2 = 2
  ifn3 = 2
  ifn4 = 1
  ivfn = 1

  al fmmetal kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, ifn4, ivfn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a normal sine wave.
f 1 0 32768 10 1
; Table #2, the "twopeaks.aiff" audio file.
f 2 0 256 1 "twopeaks.aiff" 0 0 0

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

fmb3, fmbell, fmpercfl, fmrhode, fmwurlie

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.47

fmpercfl

fmpercfl — Uses FM synthesis to create a percussive flute sound.

Description

Uses FM synthesis to create a percussive flute sound. It comes from a family of FM sounds, all using 4 basic oscillators and various architectures, as used in the TX81Z synthesizer.

Syntax

```
ares fmpercfl kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, \  
ifn3, ifn4, ivfn
```

Initialization

All these opcodes take 5 tables for initialization. The first 4 are the basic inputs and the last is the low frequency oscillator (LFO) used for vibrato. The last table should usually be a sine wave.

The initial waves should be:

- *ifn1* -- sine wave
- *ifn2* -- sine wave
- *ifn3* -- sine wave
- *ifn4* -- sine wave

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kc1, *kc2* -- Controls for the synthesizer:

- *kc1* -- Total mod index
- *kc2* -- Crossfade of two modulators
- *Algorithm* -- 4

kvdepth -- Vibrator depth

kvrate -- Vibrator rate

Examples

Here is an example of the fmpercfl opcode. It uses the file *fmpercfl.csd* [examples/fmpercfl.csd].

Example 186. Example of the fmpercfl opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fmpercfl.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kfreq = 220
  kc1 = 5
  kc2 = 5
  kvdepth = 0.005
  kvrate = 6
  ifn1 = 1
  ifn2 = 1
  ifn3 = 1
  ifn4 = 1
  ivfn = 1

  al fmpercfl kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, ifn4, ivfn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

fmb3, fmbell, fmmetal, fmrhode, fmwurlie

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.47

fmrhode

fmrhode — Uses FM synthesis to create a Fender Rhodes electric piano sound.

Description

Uses FM synthesis to create a Fender Rhodes electric piano sound. It comes from a family of FM sounds, all using 4 basic oscillators and various architectures, as used in the TX81Z synthesizer.

Syntax

```
ares fmrhode kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, \  
      ifn3, ifn4, ivfn
```

Initialization

All these opcodes take 5 tables for initialization. The first 4 are the basic inputs and the last is the low frequency oscillator (LFO) used for vibrato. The last table should usually be a sine wave.

The initial waves should be:

- *ifn1* -- sine wave
- *ifn2* -- sine wave
- *ifn3* -- sine wave
- *ifn4* -- *fwavblnk.aiff* [examples/fwavblnk.aiff]



Note

The file “fwavblnk.aiff” is also available at <ftp://ftp.cs.bath.ac.uk/pub/dream/documentation/sounds/modelling/>.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kc1, *kc2* -- Controls for the synthesizer:

- *kc1* -- Mod index 1
- *kc2* -- Crossfade of two outputs
- *Algorithm* -- 5

kvdepth -- Vibrator depth

kvrate -- Vibrator rate

Examples

Here is an example of the `fmrhode` opcode. It uses the file `fmrhode.csd` [examples/fmrhode.csd], and `fwavblnk.aiff` [examples/fwavblnk.aiff].

Example 187. Example of the `fmrhode` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fmrhode.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kfreq = 220
  kc1 = 6
  kc2 = 0
  kvdepth = 0.01
  kvrate = 3
  ifn1 = 1
  ifn2 = 1
  ifn3 = 1
  ifn4 = 2
  ivfn = 1

  al fmrhode kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, ifn4, ivfn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 32768 10 1
; Table #2, the "fwavblnk.aiff" audio file.
f 2 0 256 1 "fwavblnk.aiff" 0 0 0

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

fmb3, fmbell, fmmetal, fmpercfl, fmwurlie

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.47

fmvoice

fmvoice — FM Singing Voice Synthesis

Description

FM Singing Voice Synthesis

Syntax

```
ares fmvoice kamp, kfreq, kvowel, ktilt, kvibamt, kvibrate, ifn1, \  
      ifn2, ifn3, ifn4, ivibfn
```

Initialization

ifn1, ifn2, ifn3, ifn4 -- Tables, usually of sinewaves.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kvowel -- the vowel being sung, in the range 0-64

ktilt -- the spectral tilt of the sound in the range 0 to 99

kvibamt -- Depth of vibrato

kvibrate -- Rate of vibrato

Examples

Here is an example of the fmvoice opcode. It uses the file *fmvoice.csd* [examples/fmvoice.csd].

Example 188. Example of the fmvoice opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  No messages  
-odac      -iadc      -d      ;;RT audio I/O  
; For Non-realtime ouput leave only the line below:  
; -o fmvoice.wav -W ;; for file output any platform  
</CsOptions>  
<CsInstruments>  
  
; Initialize the global variables.  
sr = 44100  
kr = 4410  
ksmps = 10  
nchnls = 1  
  
; Instrument #1.  
instr 1  
  kamp = 30000  
  kfreq = 110  
  ; Use the fourth p-field for the vowel.  
;
```

```
kvowel = p4
ktilt = 0
kvibamt = 0.005
kvibrate = 6
ifn1 = 1
ifn2 = 1
ifn3 = 1
ifn4 = 1
ivibfn = 1

al fmvoice kamp, kfreq, kvowel, ktilt, kvibamt, kvibrate, ifn1, ifn2, ifn3, ifn4, ivibfn
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; p4 = vowel (a value from 0 to 64)
; Play Instrument #1 for one second, vowel=1.
i 1 0 1 1
; Play Instrument #1 for one second, vowel=2.
i 1 1 1 2
; Play Instrument #1 for one second, vowel=3.
i 1 2 1 3
; Play Instrument #1 for one second, vowel=4.
i 1 3 1 4
; Play Instrument #1 for one second, vowel=5.
i 1 4 1 5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitich (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

fmwurlie

fmwurlie — Uses FM synthesis to create a Wurlitzer electric piano sound.

Description

Uses FM synthesis to create a Wurlitzer electric piano sound. It comes from a family of FM sounds, all using 4 basic oscillators and various architectures, as used in the TX81Z synthesizer.

Syntax

```
ares fmwurlie kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, \  
      ifn4, ivfn
```

Initialization

All these opcodes take 5 tables for initialization. The first 4 are the basic inputs and the last is the low frequency oscillator (LFO) used for vibrato. The last table should usually be a sine wave.

The initial waves should be:

- *ifn1* -- sine wave
- *ifn2* -- sine wave
- *ifn3* -- sine wave
- *ifn4* -- *fwavblnk.aiff* [examples/fwavblnk.aiff]



Note

The file “fwavblnk.aiff” is also available at <ftp://ftp.cs.bath.ac.uk/pub/dream/documentation/sounds/modelling/>.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kc1, *kc2* -- Controls for the synthesizer:

- *kc1* -- Mod index 1
- *kc2* -- Crossfade of two outputs
- *Algorithm* -- 5

kvdepth -- Vibrator depth

kvrate -- Vibrator rate

Examples

Here is an example of the `fmwurlie` opcode. It uses the file `fmwurlie.csd` [examples/fmwurlie.csd], and `fwavblnk.aiff` [examples/fwavblnk.aiff].

Example 189. Example of the `fmwurlie` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadac     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fmwurlie.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kfreq = 440
  kc1 = 6
  kc2 = 1
  kvdepth = 0.005
  kvrate = 6
  ifn1 = 1
  ifn2 = 1
  ifn3 = 1
  ifn4 = 2
  ivfn = 1

  al fmwurlie kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, ifn4, ivfn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 32768 10 1
; Table #2, the "fwavblnk.aiff" audio file.
f 2 0 256 1 "fwavblnk.aiff" 0 0 0

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

fmb3, fmbell, fmmetal, fmpercfl, fmrhode

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.47

fof

fof — Produces sinusoid bursts useful for formant and granular synthesis.

Description

Audio output is a succession of sinusoid bursts initiated at frequency *xfund* with a spectral peak at *xform*. For *xfund* above 25 Hz these bursts produce a speech-like formant with spectral characteristics determined by the k-input parameters. For lower fundamentals this generator provides a special form of granular synthesis.

Syntax

```
ares fof xamp, xfund, xform, koct, kband, kris, kdur, kdec, iolaps, \  
      ifna, ifnb, itotdur [, iphs] [, ifmode] [, iskip]
```

Initialization

iolaps -- number of preallocated spaces needed to hold overlapping burst data. Overlaps are frequency dependent, and the space required depends on the maximum value of *xfund* * *kdur*. Can be over-estimated at no computation cost. Uses less than 50 bytes of memory per *iolap*.

ifna, *ifnb* -- table numbers of two stored functions. The first is a sine table for sineburst synthesis (size of at least 4096 recommended). The second is a rise shape, used forwards and backwards to shape the sineburst rise and decay; this may be linear (*GEN07*) or perhaps a sigmoid (*GEN19*).

itotdur -- total time during which this *fof* will be active. Normally set to p3. No new sineburst is created if it cannot complete its *kdur* within the remaining *itotdur*.

iphs (optional, default=0) -- initial phase of the fundamental, expressed as a fraction of a cycle (0 to 1). The default value is 0.

ifmode (optional, default=0) -- formant frequency mode. If zero, each sineburst keeps the *xform* frequency it was launched with. If non-zero, each is influenced by *xform* continuously. The default value is 0.

iskip (optional, default=0) -- If non-zero, skip initialisation (allows legato use).

Performance

xamp -- peak amplitude of each sineburst, observed at the true end of its rise pattern. The rise may exceed this value given a large bandwidth (say, $Q < 10$) and/or when the bursts are overlapping.

xfund -- the fundamental frequency (in Hertz) of the impulses that create new sinebursts.

xform -- the formant frequency, i.e. freq of the sinusoid burst induced by each *xfund* impulse. This frequency can be fixed for each burst or can vary continuously (see *ifmode*).

koct -- octavation index, normally zero. If greater than zero, lowers the effective *xfund* frequency by attenuating odd-numbered sinebursts. Whole numbers are full octaves, fractions transitional.

kband -- the formant bandwidth (at -6dB), expressed in Hz. The bandwidth determines the rate of exponential decay throughout the sineburst, before the enveloping described below is applied.

kris, *kdur*, *kdec* -- rise, overall duration, and decay times (in seconds) of the sinusoid burst. These values apply an enveloped duration to each burst, in similar fashion to a Csound *linen* generator but with rise and decay shapes derived from the *ifnb* input. *kris* inversely determines the skirtwidth (at -40 dB) of the induced formant region. *kdur* affects the density of sineburst overlaps, and thus the

speed of computation. Typical values for vocal imitation are .003,.02,.007.

Csound's *fof* generator is loosely based on Michael Clarke's C-coding of IRCAM's *CHANT* program (Xavier Rodet et al.). Each *fof* produces a single formant, and the output of four or more of these can be summed to produce a rich vocal imitation. *fof* synthesis is a special form of granular synthesis, and this implementation aids transformation between vocal imitation and granular textures. Computation speed depends on *kdur*, *xfund*, and the density of any overlaps.

Examples

Here is an example of the *fof* opcode. It uses the file *fof.csd* [examples/fof.csd].

Example 190. Example of the *fof* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fof.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Combine five formants together to create
; an alto-"a" sound.

; Values common to all of the formants.
kfund init 261.659
koct init 0
kris init 0.003
kdur init 0.02
kdec init 0.007
iolaps = 14850
ifna = 1
ifnb = 2
itotdur = p3

; First formant.
klamp = ampdb(0)
klform init 800
klband init 80

; Second formant.
k2amp = ampdb(-4)
k2form init 1150
k2band init 90

; Third formant.
k3amp = ampdb(-20)
k3form init 2800
k3band init 120

; Fourth formant.
k4amp = ampdb(-36)
k4form init 3500
k4band init 130

; Fifth formant.
k5amp = ampdb(-60)
k5form init 4950
k5band init 140

a1 fof klamp, kfund, klform, koct, klband, kris, \
    kdur, kdec, iolaps, ifna, ifnb, itotdur
```

```
a2 fof k2amp, kfund, k2form, koct, k2band, kris, \
      kdur, kdec, iolaps, ifna, ifnb, itotdur
a3 fof k3amp, kfund, k3form, koct, k3band, kris, \
      kdur, kdec, iolaps, ifna, ifnb, itotdur
a4 fof k4amp, kfund, k4form, koct, k4band, kris, \
      kdur, kdec, iolaps, ifna, ifnb, itotdur
a5 fof k5amp, kfund, k5form, koct, k5band, kris, \
      kdur, kdec, iolaps, ifna, ifnb, itotdur

; Combine all of the formants together.
out (a1+a2+a3+a4+a5) * 16384
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 4096 10 1
; Table #2.
f 2 0 1024 19 0.5 0.5 270 0.5

; Play Instrument #1 for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

The formant values for the alto-"a" sound were taken from the *Formant Values Appendix*.

See Also

fof2, *Formant Values Appendix*

Credits

Added in version 1 (1990)

fof2

fof2 — Produces sinusoid bursts including k-rate incremental indexing with each successive burst.

Description

Audio output is a succession of sinusoid bursts initiated at frequency *xfund* with a spectral peak at *xform*. For *xfund* above 25 Hz these bursts produce a speech-like formant with spectral characteristics determined by the k-input parameters. For lower fundamentals this generator provides a special form of granular synthesis.

fof2 implements k-rate incremental indexing into *ifna* function with each successive burst.

Syntax

```
ares fof2 xamp, xfund, xform, koct, kband, kris, kdur, kdec, iolaps, \  
      ifna, ifnb, itotdur, kphs, kgliss [, iskip]
```

Initialization

iolaps -- number of preallocated spaces needed to hold overlapping burst data. Overlaps are frequency dependent, and the space required depends on the maximum value of *xfund* * *kdur*. Can be over-estimated at no computation cost. Uses less than 50 bytes of memory per *iolap*.

ifna, *ifnb* -- table numbers of two stored functions. The first is a sine table for sineburst synthesis (size of at least 4096 recommended). The second is a rise shape, used forwards and backwards to shape the sineburst rise and decay; this may be linear (*GEN07*) or perhaps a sigmoid (*GEN19*).

itotdur -- total time during which this *fof* will be active. Normally set to p3. No new sineburst is created if it cannot complete its *kdur* within the remaining *itotdur*.

iskip (optional, default=0) -- If non-zero, skip initialization (allows legato use).

Performance

xamp -- peak amplitude of each sineburst, observed at the true end of its rise pattern. The rise may exceed this value given a large bandwidth (say, $Q < 10$) and/or when the bursts are overlapping.

xfund -- the fundamental frequency (in Hertz) of the impulses that create new sinebursts.

xform -- the formant frequency, i.e. freq of the sinusoid burst induced by each *xfund* impulse. This frequency can be fixed for each burst or can vary continuously (see *ifmode*).

koct -- octavation index, normally zero. If greater than zero, lowers the effective *xfund* frequency by attenuating odd-numbered sinebursts. Whole numbers are full octaves, fractions transitional.

kband -- the formant bandwidth (at -6dB), expressed in Hz. The bandwidth determines the rate of exponential decay throughout the sineburst, before the enveloping described below is applied.

kris, *kdur*, *kdec* -- rise, overall duration, and decay times (in seconds) of the sinusoid burst. These values apply an enveloped duration to each burst, in similar fashion to a Csound *linen* generator but with rise and decay shapes derived from the *ifnb* input. *kris* inversely determines the skirtwidth (at -40 dB) of the induced formant region. *kdur* affects the density of sineburst overlaps, and thus the speed of computation. Typical values for vocal imitation are .003,.02,.007.

kphs -- allows k-rate indexing of function table *ifna* with each successive burst, making it suitable for time-warping applications. Values of *kphs* are normalized from 0 to 1, 1 being the end of the function table *ifna*.

kgliss -- sets the end pitch of each grain relative to the initial pitch, in octaves. Thus *kgliss* = 2 means that the grain ends two octaves above its initial pitch, while *kgliss* = -3/4 has the grain ending a major sixth below. Each 1/12 added to *kgliss* raises the ending frequency one half-step. If you want no glissando, set *kgliss* to 0.

Csound's *fof* generator is loosely based on Michael Clarke's C-coding of IRCAM's *CHANT* program (Xavier Rodet et al.). Each *fof* produces a single formant, and the output of four or more of these can be summed to produce a rich vocal imitation. *fof* synthesis is a special form of granular synthesis, and this implementation aids transformation between vocal imitation and granular textures. Computation speed depends on *kdur*, *xfund*, and the density of any overlaps.



Note

The ending frequency of any grain is equal to $kform * (2 ^ kgliss)$, so setting *kgliss* too high may result in aliasing. For example, *kform* = 3000 and *kgliss* = 3 places the ending frequency over the Nyquist if *sr* = 44100. It is a good idea to scale *kgliss* accordingly.

Examples

Here is an example of the *fof2* opcode. It uses the file *fof2.csd* [examples/fof2.csd].

Example 191. Example of the *fof2* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o fof2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 128
nchnls = 2

;By Andres Cabrera 2007

instr 1          ;table-lookup vocal synthesis

kris init p12
kdur init p13
kdec init p14

iolaps init p15

ifna init 1      ; Sine wave
ifnb init 2      ; Straight line rise shape

itotdur init p3

kphs init 0      ; No phase modulation (constant kphs)

kfund line p4, p3, p5
kform line p6, p3, p7
kcoct line p8, p3, p9
kband line p10, p3, p11
kgliss line p16, p3, p17

kenv linen 5000, 0.03, p3, 0.03 ;to avoid clicking

aout      fof2      kenv, kfund, kform, kcoct, kband, kris, kdur, kdec, iolaps, \
          ifna, ifnb, itotdur, kphs, kgliss
```



```

    outs      aout, aout
    endin

</CsInstruments>
<CsScore>
f1 0 8192 10 1
f2 0 4096 7 0 4096 1

;
i1 0 4 kfund1 kfund2 kform1 kform2 koct1 koct2 kband1 kband2 kris kdur kdec iolaps
i1 + . 220 220 510 510 0 0 30 30 0.01 0.03 0.01 20
i1 + . 220 220 510 510 0 0 100 30 0.01 0.03 0.01 20
i1 + . 220 220 510 510 0 1 30 30 0.01 0.03 0.01 20
i1 + . 220 220 510 510 0 0 30 30 0.01 0.03 0.01 20
i1 + . 220 220 510 510 0 0 30 30 0.01 0.03 0.01 20
i1 + . 220 220 510 510 0 0 30 30 0.01 0.05 0.01 100
i1 + . 220 440 510 510 0 0 30 30 0.01 0.05 0.01 100

e

</CsScore>
</CsoundSynthesizer>

```

Here is another example of the fof2 opcode, which generates vowel tones using formants generated by fof2 coinciding with values from the *Formant Values* appendix. It uses the file *fof2-2.csd* [examples/fof2-2.csd].

Example 192. Example of the fof2 opcode to produce vowel sounds.

```

<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out Audio in
-odac -iadc ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fof2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 128
nchnls = 2

; Example by Chuckk Hubbard 2007

instr 1 ;table-lookup vocal synthesis

iolaps = 120
ifna = 1 ;f1 - sine wave
ifnb = 2 ;f2 - linear rise shape
itotdur = p3
iamp = p4 * 0dbfs
ifreq1 = p5 ;starting frequency
ifreq2 = p6 ;ending frequency

kamp linseg 0, .003, iamp, itotdur-.007, iamp, .003, 0, .001, 0
kfund expseg ifreq1, itotdur, ifreq2
koc1 init 0
kris init .003
kdur init .02
kdec init .007
kphs init 0
kgliss init 0

iforma = p7 ;starting spectrum
iformb = p8 ;ending spectrum

iform1a tab_i 0, iforma ;read values of 5 formants for 1st spectrum
iform2a tab_i 1, iforma
iform3a tab_i 2, iforma
iform4a tab_i 3, iforma
iform5a tab_i 4, iforma
idb1a tab_i 5, iforma ;read decibel levels for same 5 formants
idb2a tab_i 6, iforma
idb3a tab_i 7, iforma
idb4a tab_i 8, iforma
idb5a tab_i 9, iforma
iband1a tab_i 10, iforma ;read bandwidths for same 5 formants
iband2a tab_i 11, iforma

```

```

iband3a    tab_i    12, iforma
iband4a    tab_i    13, iforma
iband5a    tab_i    14, iforma
iamp1a     =    ampdb(idb1a)    ;convert db to linear multipliers
iamp2a     =    ampdb(idb2a)
iamp3a     =    ampdb(idb3a)
iamp4a     =    ampdb(idb4a)
iamp5a     =    ampdb(idb5a)

iform1b    tab_i    0, iformb    ;values of 5 formants for 2nd spectrum
iform2b    tab_i    1, iformb
iform3b    tab_i    2, iformb
iform4b    tab_i    3, iformb
iform5b    tab_i    4, iformb
idb1b      tab_i    5, iformb    ;decibel levels for 2nd set of formants
idb2b      tab_i    6, iformb
idb3b      tab_i    7, iformb
idb4b      tab_i    8, iformb
idb5b      tab_i    9, iformb
iband1b    tab_i    10, iformb    ;bandwidths for 2nd set of formants
iband2b    tab_i    11, iformb
iband3b    tab_i    12, iformb
iband4b    tab_i    13, iformb
iband5b    tab_i    14, iformb
iamp1b     =    ampdb(idb1b)    ;convert db to linear multipliers
iamp2b     =    ampdb(idb2b)
iamp3b     =    ampdb(idb3b)
iamp4b     =    ampdb(idb4b)
iamp5b     =    ampdb(idb5b)

kform1     line    iform1a, itotdur, iform1b    ;transition between formants
kform2     line    iform2a, itotdur, iform2b
kform3     line    iform3a, itotdur, iform3b
kform4     line    iform4a, itotdur, iform4b
kform5     line    iform5a, itotdur, iform5b
kband1     line    iband1a, itotdur, iband1b    ;transition of bandwidths
kband2     line    iband2a, itotdur, iband2b
kband3     line    iband3a, itotdur, iband3b
kband4     line    iband4a, itotdur, iband4b
kband5     line    iband5a, itotdur, iband5b
kamp1      line    iamp1a, itotdur, iamp1b    ;transition of amplitudes of formants
kamp2      line    iamp2a, itotdur, iamp2b
kamp3      line    iamp3a, itotdur, iamp3b
kamp4      line    iamp4a, itotdur, iamp4b
kamp5      line    iamp5a, itotdur, iamp5b

;5 formants for each spectrum
a1    fof2    kamp1, kfund, kform1, koct, kband1, kris, kdur, kdec, iolaps, ifna, ifnb, itotdur, kp
a2    fof2    kamp2, kfund, kform2, koct, kband2, kris, kdur, kdec, iolaps, ifna, ifnb, itotdur, kp
a3    fof2    kamp3, kfund, kform3, koct, kband3, kris, kdur, kdec, iolaps, ifna, ifnb, itotdur, kp
a4    fof2    kamp4, kfund, kform4, koct, kband4, kris, kdur, kdec, iolaps, ifna, ifnb, itotdur, kp
a5    fof2    kamp5, kfund, kform5, koct, kband5, kris, kdur, kdec, iolaps, ifna, ifnb, itotdur, kp

aout      =    (a1+a2+a3+a4+a5) * kamp/5    ;sum and scale

aenv linen 1, 0.05, p3, 0.05    ;to avoid clicking

outs      aout*aenv, aout*aenv
endin

</CsInstruments>
<CsScore>
f1 0 8192 10 1
f2 0 4096 7 0 4096 1

;*****
; tables of formant values adapted from MiscFormants.html
; 100's: soprano    200's: alto    300's: countertenor    400's: tenor    500's: bass
; -01: "a" sound    -02: "e" sound    -03: "i" sound    -04: "o" sound    -05: "u" sound
; p-5 through p-9: frequencies of 5 formants
; p-10 through p-14: decibel levels of 5 formants
; p-15 through p-19: bandwidths of 5 formants

;          formant frequencies          decibel levels          bandwidths
;soprano
f101 0 16 -2    800    1150    2900    3900    4950    0.001    -6    -32    -20    -5
f102 0 16 -2    350    2000    2800    3600    4950    0.001    -20    -15    -40    -56
f103 0 16 -2    270    2140    2950    3900    4950    0.001    -12    -26    -26    -44
f104 0 16 -2    450    800    2830    3800    4950    0.001    -11    -22    -22    -50
f105 0 16 -2    325    700    2700    3800    4950    0.001    -16    -35    -40    -60
;alto
f201 0 16 -2    800    1150    2800    3500    4950    0.001    -4    -20    -36    -60
f202 0 16 -2    400    1600    2700    3300    4950    0.001    -24    -30    -35    -60
f203 0 16 -2    350    1700    2700    3700    4950    0.001    -20    -30    -36    -
f204 0 16 -2    450    800    2830    3500    4950    0.001    -9    -16    -28    -55
f205 0 16 -2    325    700    2530    3500    4950    0.001    -12    -30    -40    -64
;countertenor
f301 0 16 -2    660    1120    2750    3000    3350    0.001    -6    -23    -24    -38

```

```

f302 0 16 -2 440 1800 2700 3000 3300 0.001 -14 -18 -20 -20
f303 0 16 -2 270 1850 2900 3350 3590 0.001 -24 -24 -36 -36
f304 0 16 -2 430 820 2700 3000 3300 0.001 -10 -26 -22 -34
f305 0 16 -2 370 630 2750 3000 3400 0.001 -20 -23 -30 -34
;tenor
f401 0 16 -2 650 1080 2650 2900 3250 0.001 -6 -7 -8 -22
f402 0 16 -2 400 1700 2600 3200 3580 0.001 -14 -12 -14 -20
f403 0 16 -2 290 1870 2800 3250 3540 0.001 -15 -18 -20 -30
f404 0 16 -2 400 800 2600 2800 3000 0.001 -10 -12 -12 -26
f405 0 16 -2 350 600 2700 2900 3300 0.001 -20 -17 -14 -26
;bass
f501 0 16 -2 600 1040 2250 2450 2750 0.001 -7 -9 -9 -20
f502 0 16 -2 400 1620 2400 2800 3100 0.001 -12 -9 -12 -2
f503 0 16 -2 250 1750 2600 3050 3340 0.001 -30 -16 -22 -2
f504 0 16 -2 400 750 2400 2600 2900 0.001 -11 -21 -20 -40
f505 0 16 -2 350 600 2400 2675 2950 0.001 -20 -32 -28 -36
;*****
; start dur amp start freq end freq start formant end formant
i1 0 1 .8 440 412.5 201 203
i1 + . .8 412.5 550 201 204
i1 + . .8 495 330 202 205

i1 + . .8 110 103.125 501 503
i1 + . .8 103.125 137.5 501 504
i1 + . .8 123.75 82.5 502 505

i1 7 . .4 440 412.5 201 203
i1 8 . .4 412.5 550 201 204
i1 9 . .4 495 330 202 205
i1 7 . .4 110 103.125 501 503
i1 8 . .4 103.125 137.5 501 504
i1 9 . .4 123.75 82.5 502 505
i1 + . .4 440 412.5 101 103
i1 + . .4 412.5 550 101 104
i1 + . .4 495 330 102 105
e

</CsScore>
</CsoundSynthesizer>

```

See Also

fof

Credits

Author: Rasmus Ekman

fof2 is a modification of *fof* by Rasmus Ekman

New in Csound 3.45

fofilter

fofilter — Formant filter.

Description

Fofilter generates a stream of overlapping sinewave grains, when fed with a pulse train. Each grain is the impulse response of a combination of two BP filters. The grains are defined by their attack time (determining the skirtwidth of the formant region at -60dB) and decay time (-6dB bandwidth). Overlapping will occur when $1/\text{freq} < \text{decay}$, but, unlike FOF, there is no upper limit on the number of overlaps. The original idea for this opcode came from J McCartney's formlet class in SuperCollider, but this is possibly implemented differently(?).

Syntax

```
asig fofilter ain, kcf, kris, kdec[, istor]
```

Initialization

istor --initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- input signal.

kcf -- filter centre frequency

kris -- impulse response attack time (secs).

kdec -- impulse response decay time (secs).

Examples

Example 193. Example

```
kfe      expseg 10, p3*0.9, 180, p3*0.1, 175
kenv     linen 1000, 0.05, p3, 0.05
asig     buzz kenv, kfe, sr/(2*kfe), 1
afil     fofilter asig, 900, 0.007, 0.04

out afil
```

Credits

Author: Victor Lazzarini;
January 2005

New plugin in version 5

January 2005.

fog

fog — Audio output is a succession of grains derived from data in a stored function table

Description

Audio output is a succession of grains derived from data in a stored function table *ifna*. The local envelope of these grains and their timing is based on the model of *fof* synthesis and permits detailed control of the granular synthesis.

Syntax

```
ares fog xamp, xdens, xtrans, aspd, koct, kband, kris, kdur, kdec, \  
      iolaps, ifna, ifnb, itotdur [, iphs] [, itmode] [, iskip]
```

Initialization

iolaps -- number of pre-located spaces needed to hold overlapping grain data. Overlaps are density dependent, and the space required depends on the maximum value of $xdens * kdur$. Can be overestimated at no computation cost. Uses less than 50 bytes of memory per *iolap*.

ifna, *ifnb* -- table numbers of two stored functions. The first is the data used for granulation, usually from a soundfile (*GEN01*). The second is a rise shape, used forwards and backwards to shape the grain rise and decay; this is normally a sigmoid (*GEN19*) but may be linear (*GEN05*).

itotdur -- total time during which this *fog* will be active. Normally set to p3. No new grain is created if it cannot complete its *kdur* within the remaining *itotdur*.

iphs (optional) -- initial phase of the fundamental, expressed as a fraction of a cycle (0 to 1). The default value is 0.

itmode (optional) -- transposition mode. If zero, each grain keeps the *xtrans* value it was launched with. If non-zero, each is influenced by *xtrans* continuously. The default value is 0.

iskip (optional, default=0) -- If non-zero, skip initialization (allows legato use).

Performance

xamp -- amplitude factor. Amplitude is also dependent on the number of overlapping grains, the interaction of the rise shape (*ifnb*) and the exponential decay (*kband*), and the scaling of the grain waveform (*ifna*). The actual amplitude may therefore exceed *xamp*.

xdens -- density. The frequency of grains per second.

xtrans -- transposition factor. The rate at which data from the stored function table *ifna* is read within each grain. This has the effect of transposing the original material. A value of 1 produces the original pitch. Higher values transpose upwards, lower values downwards. Negative values result in the function table being read backwards.

aspd -- Starting index pointer. *aspd* is the normalized index (0 to 1) to table *ifna* that determines the movement of a pointer used as the starting point for reading data within each grain. (*xtrans* determines the rate at which data is read starting from this pointer.)

koct -- octavation index. The operation of this parameter is identical to that in *fof*.

kband, *kris*, *kdur*, *kdec* -- grain envelope shape. These parameters determine the exponential decay (*kband*), and the rise (*kris*), overall duration (*kdur*), and decay (*kdec*) times of the grain envelope. Their operation is identical to that of the local envelope parameters in *fof*.

Examples

```
;p4 = transposition factor
;p5 = speed factor
;p6 = function table for grain data
i1 = sr/ftlen(p6) ;scaling to reflect sample rate and table length
a1 phasor i1*p5 ;index for speed
a2 fog 5000, 100, p4, a1, 0, 0, , .01, .02, .01, 2, p6, 1, p3, 0, 1
```

Credits

Author: Michael Clark
Huddersfield
May 1997

New in version 3.46

The Csound fog generator is by Michael Clarke, extending his earlier work based on IRCAM's FOF algorithm.

Added notes by Rasmus Ekman on September 2002.

fold

fold — Adds artificial foldover to an audio signal.

Description

Adds artificial foldover to an audio signal.

Syntax

```
ares fold asig, kincr
```

Performance

asig -- input signal

kincr -- amount of foldover expressed in multiple of sampling rate. Must be ≥ 1

fold is an opcode which creates artificial foldover. For example, when *kincr* is equal to 1 with *sr*=44100, no foldover is added. When *kincr* is set to 2, the foldover is equivalent to a downsampling to 22050, when it is set to 4, to 11025 etc. Fractional values of *kincr* are possible, allowing a continuous variation of foldover amount. This can be used for a wide range of special effects.

Examples

Here is an example of the fold opcode. It uses the file *fold.csd* [examples/fold.csd].

Example 194. Example of the fold opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o fold.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use an ordinary sine wave.
asig oscils 30000, 100, 1

; Vary the fold-over amount from 1 to 200.
kincr line 1, p3, 200
al fold asig, kincr

out al
endin

</CsInstruments>
<CsScore>
```



```
; Play Instrument #1 for four seconds.  
i 1 0 4  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

follow

follow — Envelope follower unit generator.

Description

Envelope follower unit generator.

Syntax

```
ares follow asig, idt
```

Initialization

idt -- This is the period, in seconds, that the average amplitude of *asig* is reported. If the frequency of *asig* is low then *idt* must be large (more than half the period of *asig*)

Performance

asig -- This is the signal from which to extract the envelope.

Examples

Here is an example of the follow opcode. It uses the file *follow.csd* [examples/follow.csd], and *beats.wav* [examples/beats.wav].

Example 195. Example of the follow opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o follow.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - play a WAV file.
instr 1
  al soundin "beats.wav"
  out al
endin

; Instrument #2 - have another waveform follow the WAV file.
instr 2
  ; Follow the WAV file.
  as soundin "beats.wav"
  af follow as, 0.01

  ; Use a sine waveform.
  as oscil 4000, 440, 1
  ; Have it use the amplitude of the followed WAV file.
  al balance as, af
```

```
    out a1
  endin

</CsInstruments>
<CsScore>

  ; Just generate a nice, ordinary sine wave.
  f 1 0 32768 10 1

  ; Play Instrument #1 for two seconds.
  i 1 0 2
  ; Play Instrument #2 for two seconds.
  i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

To avoid zipper noise, by discontinuities produced from complex envelope tracking, a lowpass filter could be used, to smooth the estimated envelope.

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

follow2

follow2 — Another controllable envelope extractor.

Description

A controllable envelope extractor using the algorithm attributed to Jean-Marc Jot.

Syntax

```
ares follow2 asig, katt, krel
```

Performance

asig -- the input signal whose envelope is followed

katt -- the attack rate (60dB attack time in seconds)

krel -- the decay rate (60dB decay time in seconds)

The output tracks the amplitude envelope of the input signal. The rate at which the output grows to follow the signal is controlled by the *katt*, and the rate at which it decreases in response to a lower amplitude, is controlled by the *krel*. This gives a smoother envelope than *follow*.

Examples

Here is an example of the follow2 opcode. It uses the file *follow2.csd* [examples/follow2.csd], and *beats.wav* [examples/beats.wav].

Example 196. Example of the follow2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o follow2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - play a WAV file.
instr 1
  al soundin "beats.wav"
  out al
endin

; Instrument #2 - have another waveform follow the WAV file.
instr 2
  ; Follow the WAV file.
  as soundin "beats.wav"
  af follow2 as, 0.01, 0.1

  ; Use a noise waveform.
```

```
ar rand 44100
; Have it use the amplitude of the followed WAV file.
al balance ar, af

out a1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch
The algorithm for the *follow2* is attributed to Jean-Marc Jot.
University of Bath, Codemist Ltd.
Bath, UK
February 2000

Example written by Kevin Conder.

New in Csound version 4.03

Added notes by Rasmus Ekman on September 2002.

foscil

foscil — A basic frequency modulated oscillator.

Description

A basic frequency modulated oscillator.

Syntax

```
ares foscil xamp, kcps, xcar, xmod, kndx, ifn [, iphs]
```

Initialization

ifn -- function table number. Requires a wrap-around guard point.

iphs (optional, default=0) -- initial phase of waveform in table *ifn*, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is 0.

Performance

xamp -- the amplitude of the output signal.

kcps -- a common denominator, in cycles per second, for the carrier and modulating frequencies.

xcar -- a factor that, when multiplied by the *kcps* parameter, gives the carrier frequency.

xmod -- a factor that, when multiplied by the *kcps* parameter, gives the modulating frequency.

kndx -- the modulation index.

foscil is a composite unit that effectively banks two *oscil* opcodes in the familiar Chowning FM setup, wherein the audio-rate output of one generator is used to modulate the frequency input of another (the “carrier”). Effective carrier frequency = $kcps * xcar$, and modulating frequency = $kcps * xmod$. For integral values of *xcar* and *xmod*, the perceived fundamental will be the minimum positive value of $kcps * (xcar - n * xmod)$, $n = 0, 1, 2, \dots$. The input *kndx* is the index of modulation (usually time-varying and ranging 0 to 4 or so) which determines the spread of acoustic energy over the partial positions given by $n = 0, 1, 2, \dots$, etc. *ifn* should point to a stored sine wave. Previous to version 3.50, *xcar* and *xmod* could be k-rate only.

The actual formula used for this implementation of FM synthesis is $xamp * \cos(2\pi * t * kcps * xcar + kndx * \sin(2\pi * t * kcps * xmod) - \#)$, assuming that the table is a sine wave.

Examples

Here is an example of the foscil opcode. It uses the file *foscil.csd* [examples/foscil.csd].

Example 197. Example of the foscil opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
```

```
-odac          -iadc      -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o foscil.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a basic FM waveform.
instr 1
  kamp = 10000
  kcps = 440
  kcar = 600
  kmod = 210
  kndx = 2
  ifn = 1

  al foscil kamp, kcps, kcar, kmod, kndx, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Example written by Kevin Conder.

foscili

foscili — Basic frequency modulated oscillator with linear interpolation.

Description

Basic frequency modulated oscillator with linear interpolation.

Syntax

```
ares foscili xamp, kcps, xcar, xmod, kndx, ifn [, iphs]
```

Initialization

ifn -- function table number. Requires a wrap-around guard point.

iphs (optional, default=0) -- initial phase of waveform in table *ifn*, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is 0.

Performance

xamp -- the amplitude of the output signal.

kcps -- a common denominator, in cycles per second, for the carrier and modulating frequencies.

xcar -- a factor that, when multiplied by the *kcps* parameter, gives the carrier frequency.

xmod -- a factor that, when multiplied by the *kcps* parameter, gives the modulating frequency.

kndx -- the modulation index.

foscili differs from *foscil* in that the standard procedure of using a truncated phase as a sampling index is here replaced by a process that interpolates between two successive lookups. Interpolating generators will produce a noticeably cleaner output signal, but they may take as much as twice as long to run. Adequate accuracy can also be gained without the time cost of interpolation by using large stored function tables of 2K, 4K or 8K points if the space is available.

Examples

Here is an example of the foscili opcode. It uses the file *foscili.csd* [examples/foscili.csd].

Example 198. Example of the foscili opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o foscili.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
```



```
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a basic FM waveform.
instr 1
  kamp = 10000
  kcps = 440
  kcar = 600
  kmod = 210
  kndx = 2
  ifn = 1

  al foscil kamp, kcps, kcar, kmod, kndx, ifn
  out al
endin

; Instrument #2 - the basic FM waveform with extra interpolation.
instr 2
  kamp = 10000
  kcps = 440
  kcar = 600
  kmod = 210
  kndx = 2
  ifn = 1

  al foscili kamp, kcps, kcar, kmod, kndx, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave table with a small amount of data.
f 1 0 4096 10 1

; Play Instrument #1, the basic FM instrument, for
; two seconds. This should sound relatively rough.
i 1 0 2

; Play Instrument #2, the interpolated FM instrument, for
; two seconds. This should sound relatively smooth.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Example written by Kevin Conder.

fout

fout — Outputs a-rate signals to an arbitrary number of channels.

Description

fout outputs *N* a-rate signals to a specified file of *N* channels.

Syntax

```
fout ifilename, iformat, aout1 [, aout2, aout3,...,aoutN]
```

Initialization

ifilename -- the output file's name (in double-quotes).

iformat -- a flag to choose output file format (note: Csound versions older than 5.0 may only support formats 0, 1, and 2):

- 0 - 32-bit floating point samples without header (binary PCM multichannel file)
- 1 - 16-bit integers without header (binary PCM multichannel file)
- 2 - 16-bit integers with a header. The header type depends on the render (-o) format. For example, if the user chooses the AIFF format (using the *-A flag*), the header format will be AIFF type.
- 3 - u-law samples with a header (see iformat=2).
- 4 - 16-bit integers with a header (see iformat=2).
- 5 - 32-bit integers with a header (see iformat=2).
- 6 - 32-bit floats with a header (see iformat=2).
- 7 - 8-bit unsigned integers with a header (see iformat=2).
- 8 - 24-bit integers with a header (see iformat=2).
- 9 - 64-bit floats with a header (see iformat=2).

In addition, Csound versions 5.0 and later allow for explicitly selecting a particular header type by specifying the format as 10 * fileType + sampleFormat, where fileType may be 1 for WAV, 2 for AIFF, 3 for raw (headerless) files, and 4 for IRCAM; sampleFormat is one of the above values in the range 0 to 9, except sample format 0 is taken from the command line (-o), format 1 is 8-bit signed integers, and format 2 is a-law. So, for example, iformat=25 means 32-bit integers with AIFF header.

Performance

aout1,... aoutN -- signals to be written to the file. In the case of raw files, the expected range of audio signals is determined by the selected sample format; for sound files with a header like WAV and AIFF, the audio signals should be in the range -0dbfs to 0dbfs.

fout (file output) writes samples of audio signals to a file with any number of channels. Channel number depends by the number of *aoutN* variables (i.e. a mono signal with only an a-rate argument, a stereo signal with two a-rate arguments etc.) Maximum number of channels is fixed to 64. Multiple *fout* opcodes can be present in the same instrument, referring to different files.

Notice that, unlike *out*, *outs* and *outq*, *fout* does not zero the audio variable so you must zero it after calling it. If polyphony is to be used, you can use *vincr* and *clear* opcodes for this task.

Notice that *fout* and *foutk* can use either a string containing a file pathname, or a handle-number generated by *fiopen*. Whereas, with *fouti* and *foutir*, the target file can be only specified by means of a handle-number.



Note

If you are using *fout* to generate an audio file for the global output of csound, you might want to use the *monitor* opcode, which can tap the output buffer, to avoid having to route

Examples

Here is a simple example of the *fout* opcode. It uses the file *fout.csd* [examples/fout.csd].

Example 199. Example of the *fout* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fout.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  iamp = 10000
  icps = 440
  iphs = 0

  ; Create an audio signal.
  asig oscils iamp, icps, iphs

  ; Write the audio signal to a headerless audio file
  ; called "fout.raw".
  fout "fout.raw", 1, asig
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the *fout* opcode with a polyphonic score. It uses the file *fout_poly.csd* [examples/fout_poly.csd] and *beats.wav* [examples/beats.wav].

Example 200. Example of the fout opcode with a polyphonic score.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in      No messages
-odac            -iadc         -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o fout_poly.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1 - Play an audio file.
instr 1
; Generate an audio signal using
; the audio file "beats.wav".
asig soundin "beats.wav"

out asig
endin

; Instrument #2 - Create a basic tone.
instr 2
iamp = 5000
icps = 440
iphs = 0

; Create an audio signal.
asig oscils iamp, icps, iphs

out asig
endin

; Instrument #99 - Save the global signal to a file.
instr 99
; Read the csound output buffer
aoutput monitor
; Write the output of csound to a headerless
; audio file called "fout_poly.raw".
fout "fout_poly.raw", 1, aoutput

endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2

; Play Instrument #2 every quarter-second.
i 2 0.00 0.1
i 2 0.25 0.1
i 2 0.50 0.1
i 2 0.75 0.1
i 2 1.00 0.1
i 2 1.25 0.1
i 2 1.50 0.1
i 2 1.75 0.1

; Make sure the global instrument, #99, is running
; during the entire performance (2 seconds).
i 99 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of *fout*, using it to save the contents of a table to an audio file. It uses the file *fout_ftable.csd* [examples/fout_ftable.csd] and *beats.wav* [examples/beats.wav].

Example 201. Example of the fout opcode to save the contents of an f-table.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o fout_ftable.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
; By: Jonathan Murphy 2007

gilen      =      131072
gicps      =      sr/gilen
gitab      ftgen      1, 0, gilen, 10, 1

instr 1

/***** write file to table *****/

ain      diskin      "beats.wav", 1, 0, 1
aphs     phasor      gicps
andx     =      apha * gilen
          tablew      ain, andx, gitab

/***** write table to file *****/

aosc     table      apha, gitab, 1
          out      aosc
          fout      "beats_copy.wav", 6, aosc

endin

</CsInstruments>
<CsScore>
i1 0 2
e
</CsScore>
</CsoundSynthesizer>
```

See Also

fiopen, fouti, foutir, foutk, monitor

Credits

Author: Gabriel Maldonado
Italy
1999

The simple example was written by Kevin Conder.

New in Csound version 3.56

October 2002. Added a note from Richard Dobson.

fouti

fouti — Outputs i-rate signals of an arbitrary number of channels to a specified file.

Description

fouti output *N* i-rate signals to a specified file of *N* channels.

Syntax

```
fouti ihandle, iformat, iflag, iout1 [, iout2, iout3, ..., ioutN]
```

Initialization

ihandle -- a number which specifies this file.

iformat -- a flag to choose output file format:

- 0 - floating point in text format
- 1 - 32-bit floating point in binary format

iflag -- choose the mode of writing to the ASCII file (valid only in ASCII mode; in binary mode *iflag* has no meaning, but it must be present anyway). *iflag* can be a value chosen among the following:

- 0 - line of text without instrument prefix
- 1 - line of text with instrument prefix (see below)
- 2 - reset the time of instrument prefixes to zero (to be used only in some particular cases. See below)

iout, ..., *ioutN* -- values to be written to the file

Performance

fouti and *foutir* write i-rate values to a file. The main use of these opcodes is to generate a score file during a realtime session. For this purpose, the user should set *iformat* to 0 (text file output) and *iflag* to 1, which enable the output of a prefix consisting of the strings *inum*, *actiontime*, and *duration*, before the values of *iout1*...*ioutN* arguments. The arguments in the prefix refer to instrument number, action time and duration of current note.

Notice that *fout* and *foutk* can use either a string containing a file pathname, or a handle-number generated by *fiopen*. Whereas, with *fouti* and *foutir*, the target file can be only specified by means of a handle-number.

See Also

fiopen, *fout*, *foutir*, *foutk*

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

foutir

foutir — Outputs i-rate signals from an arbitrary number of channels to a specified file.

Description

foutir output *N* i-rate signals to a specified file of *N* channels.

Syntax

```
foutir ihandle, iformat, iflag, iout1 [, iout2, iout3, ..., ioutN]
```

Initialization

ihandle -- a number which specifies this file.

iformat -- a flag to choose output file format:

- 0 - floating point in text format
- 1 - 32-bit floating point in binary format

iflag -- choose the mode of writing to the ASCII file (valid only in ASCII mode; in binary mode *iflag* has no meaning, but it must be present anyway). *iflag* can be a value chosen among the following:

- 0 - line of text without instrument prefix
- 1 - line of text with instrument prefix (see below)
- 2 - reset the time of instrument prefixes to zero (to be used only in some particular cases. See below)

iout, ..., *ioutN* -- values to be written to the file

Performance

fouti and *foutir* write i-rate values to a file. The main use of these opcodes is to generate a score file during a realtime session. For this purpose, the user should set *iformat* to 0 (text file output) and *iflag* to 1, which enable the output of a prefix consisting of the strings *inum*, *actiontime*, and *duration*, before the values of *iout1*...*ioutN* arguments. The arguments in the prefix refer to instrument number, action time and duration of current note.

The difference between *fouti* and *foutir* is that, in the case of *fouti*, when *iflag* is set to 1, the duration of the first opcode is undefined (so it is replaced by a dot). Whereas, *foutir* is defined at the end of note, so the corresponding text line is written only at the end of the current note (in order to recognize its duration). The corresponding file is linked by the *ihandle* value generated by the *fiopen* opcode. So *fouti* and *foutir* can be used to generate a Csound score while playing a realtime session.

Notice that *fout* and *foutk* can use either a string containing a file pathname, or a handle-number generated by *fiopen*. Whereas, with *fouti* and *foutir*, the target file can be only specified by means of a handle-number.

See Also

fiopen, fout, fouti, foutk

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

foutk

foutk — Outputs k-rate signals of an arbitrary number of channels to a specified file, in raw (headerless) format.

Description

foutk outputs *N* k-rate signals to a specified file of *N* channels.

Syntax

```
foutk ifilename, iformat, kout1 [, kout2, kout3,...,koutN]
```

Initialization

ifilename -- the output file's name (in double-quotes).

iformat -- a flag to choose output file format (note: Csound versions older than 5.0 may only support formats 0 and 1):

- 0 - 32-bit floating point samples without header (binary PCM multichannel file)
- 1 - 16-bit integers without header (binary PCM multichannel file)
- 2 - 16-bit integers without header (binary PCM multichannel file)
- 3 - u-law samples without header
- 4 - 16-bit integers without header
- 5 - 32-bit integers without header
- 6 - 32-bit floats without header
- 7 - 8-bit unsigned integers without header
- 8 - 24-bit integers without header
- 9 - 64-bit floats without header

Performance

kout1,...,koutN -- control-rate signals to be written to the file. The expected range of the signals is determined by the selected sample format.

foutk operates in the same way as *fout*, but with k-rate signals. *iformat* can be set only in the range 0 to 9, or 0 to 1 with an old version of Csound.

Notice that *fout* and *foutk* can use either a string containing a file pathname, or a handle-number generated by *fiopen*. Whereas, with *fouti* and *foutir*, the target file can be only specified by means of a handle-number.

See Also

fiopen, fout, fouti, foutir

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

fprintks

fprintks — Similar to printks but prints to a file.

Description

Similar to *printks* but prints to a file.

Syntax

```
fprintks "filename", "string", [, kval1] [, kval2] [...]
```

Initialization

"filename" -- name of the output file.

"string" -- the text string to be printed. Can be up to 8192 characters and must be in double quotes.

Performance

kval1, *kval2*, ... (optional) -- The k-rate values to be printed. These are specified in *"string"* with the standard C value specifier (%f, %d, etc.) in the order given.

fprintks is similar to the *printks* opcode except it outputs to a file and doesn't have a *itime* parameter. For more information about output formatting, please look at *printks's* [documentation](#).

Examples

Here is an example of the fprintks opcode. It uses the file *fprintks.csd* [examples/fprintks.csd].

Example 202. Example of the fprintks opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac        -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o fprintks.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Matt Ingalls, edited by Kevin Conder. */
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a score generator example.
instr 1
; K-rate stuff.
kstart init 0
kdur linrand 10
kpitch linrand 8

; Printing to to a file called "my.sco".
fprintks "my.sco", "i1\\t%2.2f\\t%2.2f\\t%2.2f\\n", kstart, kdur, 4*kpitch
```

```

    knext linrand 1
    kstart = kstart + knext
endin

</CsInstruments>
<CsScore>

/* Written by Matt Ingalls, edited by Kevin Conder. */
; Play Instrument #1.
i 1 0 0.001

</CsScore>
</CsoundSynthesizer>

```

This example will generate a file called “my.sco”. It should contain lines like this:

```

i1      0.00    3.94    10.26
i1      0.20    3.35    6.22
i1      0.67    3.65    11.33
i1      1.31    1.42    4.13

```

Here is an example of the `fprintks` opcode, which converts a standard MIDI file to a csound score. It uses the file `fprintks-2.csd` [examples/fprintks-2.csd].

Example 203. Example of the `fprintks` opcode to convert a MIDI file to a csound score.

```

<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
; -odac       -iadc       -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
-n -Fmidichn_advanced.mid
; Don't write audio ouput to disk and use the file midichn_advanced.mid as MIDI input
</CsOptions>
<CsInstruments>

sr      = 48000
ksmps   = 16
nchnls  = 2

; Example by Jonathan Murphy 2007

        ; assign all midi events to instr 1000
        massign 0, 1000
        pgmassign 0, 1000

instr 1000

    ktim timeinsts

    kst, kch, kd1, kd2 midiin
    if (kst != 0) then
    ; p4 = MIDI event type   p5 = channel   p6 = data1   p7 = data2
        fprintks "MIDI2cs.sco", "i1\\t%f\\t%f\\t%d\\t%d\\t%d\\t%d\\n", ktim, 1/kr, kst, kch, k
    endif

endin

</CsInstruments>
<CsScore>
i1000 0 10000
e
</CsScore>
</CsoundSynthesizer>

```

This example will generate a file called “MIDI2cs.sco” containing i-events according to the MIDI file

Here is an advanced example of the `fprintks` opcode, which generates scores for Csound. It uses the

file *scogen-2.csd* [examples/scogen.csd].

Example 204. Example of the `fprintks` opcode to create a Csound score file generator using Csound.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
; -odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
-n
;Don't write audio ouput to disk
</CsOptions>
<CsInstruments>
;=====
;          scogen.csd          by: Matt Ingalls
;
;  a "port" of sorts
;    of the old "mills" score generator (scogen)
;
;  this instrument creates a schottstaedt.sco file
;  to be used with the schottstaedt.orc file
;
;  as long as you dont save schottstaedt.orc as a .csd
;  file, you should be able to keep it open in MacCsound
;  and render each newly generated .sco file.
;
;=====

gScoName = "/Users/matt/Desktop/schottstaedt.sco"      ; the name of the file to be generated

sr      =      100      ; this defines our temporal resolution,
                      ; an sr of 100 means we will generate p2 and p3 values
                      ; to the nearest 1/100th of a second

ksmps   =      1      ; set kr=sr so we can do everything at k-rate

; some print opcodes
opcode PrintInteger, 0, k
  kval    xin
  fprintks gScoName, "%d", kval
endop

opcode PrintFloat, 0, k
  kval    xin
  fprintks gScoName, "%f", kval
endop

opcode PrintTab, 0, 0
  fprintks gScoName, "%n"
endop

opcode PrintReturn, 0, 0
  fprintks gScoName, "%r"
endop

; recursively calling opcode to handle all the optional parameters
opcode ProcessAdditionalPfields, 0, ikio
  iptable, kndx, iNumPfields, iPfield xin

  ; additional pfields start at 5, we use a default 0 to identify the first call
  iPfield = (iPfield == 0 ? 5 : iPfield)

  if (iPfield > iNumPfields) goto endloop
  ; find our tables
  iMinTable table 2*iPfield-1, iptable
  iMaxTable table 2*iPfield, iptable

  ; get values from our tables
  kMin tablei kndx, iMinTable
  kMax tablei kndx, iMaxTable

  ; find a random value in the range and write it to the score
  fprintks gScoName, "%t%f", kMin + rnd(kMax-kMin)

  ; recursively call for any additional pfields.
  ProcessAdditionalPfields iptable, kndx, iNumPfields, iPfield + 1
```

```
        endloop:

endop

/* =====
Generate a gesture of i-statements

p2 = start of the gesture
p3 = duration of the gesture
p4 = number of a function that contains a list of all
    function table numbers used to define the
    pfield random distribution
p5 = scale generated p4 values according to density (0=off, 1=on) [todo]
p6 = let durations overlap gesture duration (0=off, 1=on) [todo]
p7 = seed for random number generator seed [todo]
=====
*/
instr Gesture

; initialize
iResolution = 1/sr

kNextStart init p2
kCurrentTime init p2

iNumPfields table      0, p4
iInstrMinTable table   1, p4
iInstrMaxTable table   2, p4
iDensityMinTable table 3, p4
iDensityMaxTable table 4, p4
iDurMinTable table     5, p4
iDurMaxTable table     6, p4
iAmpMinTable table     7, p4
iAmpMaxTable table     8, p4

; check to make sure there is enough data
print iNumPfields
if iNumPfields < 4 then
    prints "dError: At least 4 p-fields (8 functions) need to be specified.%n", iNumPfields
    turnoff
endif

; initial comment
fprints gScoName, "%!Generated Gesture from %f to %f seconds%n %!%t%twith a p-max of %d%n%n"

; k-rate stuff
if (kCurrentTime >= kNextStart) then ; write a new note!

    kndx = (kCurrentTime-p2)/p3

    ; get the required pfield ranges
    kInstMin tablei kndx, iInstrMinTable
    kInstMax tablei kndx, iInstrMaxTable
    kDensMin tablei kndx, iDensityMinTable
    kDensMax tablei kndx, iDensityMaxTable
    kDurMin tablei kndx, iDurMinTable
    kDurMax tablei kndx, iDurMaxTable
    kAmpMin tablei kndx, iAmpMinTable
    kAmpMax tablei kndx, iAmpMaxTable

    ; find random values for all our required parametrts and print the i-statement
    fprintks gScoName, "i%d%f%f%f%f", kInstMin + rnd(kInstMax-kInstMin), kNextStart, kDurMi

    ; now any additional pfields
    ProcessAdditionalPfields p4, kndx, iNumPfields

    PrintReturn

    ; calculate next starttime
    kDensity = kDensMin + rnd(kDensMax-kDensMin)
    if (kDensity < iResolution) then
        kDensity = iResolution
    endif
    kNextStart = kNextStart + kDensity
endif

kCurrentTime = kCurrentTime + iResolution
endin

</CsInstruments>
<CsScore>
/*
=====
scogen.sco

this csound module generates a score file
```

```
you specify a gesture of notes by giving
the "gesture" instrument a number to a
(negative) gen2 table.

this table stores numbers to pairs of functions.
each function-pair represents a range (min-max)
of randomness for every pfield for the notes to
be generated.
=====
*/

; common tables for pfield ranges
f100 0 2 -7 0 2 0 ; static 0
f101 0 2 -7 1 2 1 ; static 1
f102 0 2 -7 0 2 1 ; ramp 0->1
f103 0 2 -7 1 2 0 ; ramp 1->0
f105 0 2 -7 10 2 10 ; static 10
f106 0 2 -7 .1 2 .1 ; static .1

; specific pfield ranges
f10 0 2 -7 .8 2 .01 ; density
f11 0 2 -7 8 2 4 ; pitchmin
f12 0 2 -7 8 2 12 ; pitchmax

;=== table containing the function numbers used for all the p-field distributions
;
; p1 - table number
; p2 - time table is instantiated
; p3 - size of table (must be >= p5!)
; p4 - gen# (should be = -2)
; p5 - number of pfields of each note to be generated
; p6 - table number of the function representing the minimum possible note number (p1) of a
; p7 - table number of the function representing the maximum possible note number (p1) of a
; p8 - table number of the function representing the minimum possible noteon-to-noteon time
; p9 - table number of the function representing the maximum possible noteon-to-noteon time
; p10 - table number of the function representing the minimum possible duration (p3) of a ge
; p11 - table number of the function representing the maximum possible duration (p3) of a ge
; p12 - table number of the function representing the maximum possible amplitude (p4) of a ge
; p13 - table number of the function representing the maximum possible amplitude (p5) of a ge
; p14,p16.. - table number of the function representing the minimum possible value for additi
; p15,p17.. - table number of the function representing the maximum possible value for additi

; siz 2 #pds p1min p1max p2min p2max p3min p3max p4min p4max p5min p5ma
f1 0 32 -2 6 101 101 10 10 101 105 100 106 11 12 100 1

;gesture definitions
; start dur pTble scale overlap seed
i"Gesture" 0 60 1 ;todo-->0 0 123
</CsScore>
</CsoundSynthesizer>
```

This example will generate a file called “schottstaedt.sco” which can be used as a score together with *schottstaedt.orc* [examples/schottstaedt.orc]

See Also

printks

Credits

Author: Matt Ingalls
January 2003

fprints

fprints — Similar to prints but prints to a file.

Description

Similar to *prints* but prints to a file.

Syntax

```
fprints "filename", "string" [, ival1] [, ival2] [...]
```

Initialization

"filename" -- name of the output file.

"string" -- the text string to be printed. Can be up to 8192 characters and must be in double quotes.

ival1, *ival2*, ... (optional) -- The i-rate values to be printed. These are specified in *"string"* with the standard C value specifier (%f, %d, etc.) in the order given.

Performance

fprints is similar to the *prints* opcode except it outputs to a file. For more information about output formatting, please look at *prints's documentation*.

Examples

Here is an example of the fprints opcode. It uses the file *fprints.csd* [examples/fprints.csd].

Example 205. Example of the fprints opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o fprints.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Matt Ingalls, edited by Kevin Conder. */
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a score generator example.
instr 1
; Print to the file "my.sco".
fprints "my.sco", "%!Generated score by ma++\n \n"
endin

</CsInstruments>
<CsScore>
```

```
/* Written by Matt Ingalls, edited by Kevin Conder. */  
; Play Instrument #1.  
i 1 0 0.001  
  
</CsScore>  
</CsoundSynthesizer>
```

This example will generate a file called “my.sco”. It should contain a line like this:

```
;Generated score by ma++
```

See Also

prints

Credits

Author: Matt Ingalls
January 2003

frac

frac — Returns the fractional part of a decimal number.

Description

Returns the fractional part of x .

Syntax

frac(x) (init-rate or control-rate args; also works at audio rate in Csound5)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

Examples

Here is an example of the frac opcode. It uses the file *frac.csd* [examples/frac.csd].

Example 206. Example of the frac opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o frac.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = 16 / 5
  i2 = frac(i1)

  print i2
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i2 = 0.200
```

See Also

abs, exp, int, log, log10, i, sqrt

Credits

Example written by Kevin Conder.

freeverb

freeverb — Opcode version of Jezar's Freeverb

Description

freeverb is a stereo reverb unit based on Jezar's public domain C++ sources, composed of eight parallel comb filters on both channels, followed by four allpass units in series. The filters on the right channel are slightly detuned compared to the left channel in order to create a stereo effect.

Syntax

```
aoutL, aoutR freeverb ainL, ainR, kRoomSize, kHFDamp[, iSRate[, iSkip]]
```

Initialization

iSRate (optional, defaults to 44100): adjusts the reverb parameters for use with the specified sample rate (this will affect the length of the delay lines in samples, and, as of the latest CVS version, the high frequency attenuation). Only integer multiples of 44100 will reproduce the original character of the reverb exactly, so it may be useful to set this to 44100 or 88200 for an orchestra sample rate of 48000 or 96000 Hz, respectively. While *iSRate* is normally expected to be close to the orchestra sample rate, different settings may be useful for special effects.

iSkip (optional, defaults to zero): if non-zero, initialization of the opcode will be skipped, whenever possible.

Performance

ainL, *ainR* -- input signals; usually both are the same, but different inputs can be used for special effect



Note

It is recommended to process the input signal(s) with the *denorm* opcode in order to avoid denormalized numbers which could significantly increase CPU usage in some cases

aoutL, *aoutR* -- output signals for left and right channel

kRoomSize (range: 0 to 1) -- controls the length of the reverb, a higher value means longer reverb. Settings above 1 may make the opcode unstable.

kHFDamp (range: 0 to 1): high frequency attenuation; a value of zero means all frequencies decay at the same rate, while higher settings will result in a faster decay of the high frequency range.

Examples

Here is an example of the *freeverb* opcode. It uses the file *freeverb.csd* [examples/freeverb.csd].

Example 207. An example of the freeverb opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o freeverb.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr          = 44100
ksmps      = 32
nchnls     = 2
0dbfs      = 1

          instr 1
a1         vco2 0.75, 440, 10
kfrq       port 100, 0.008, 20000
a1         butterlp a1, kfrq
a2         linseg 0, 0.003, 1, 0.01, 0.7, 0.005, 0, 1, 0
a1         = a1 * a2
          denorm a1
aL, aR     freeverb a1, a1, 0.9, 0.35, sr, 0
          outs a1 + aL, a1 + aR
          endin

</CsInstruments>
<CsScore>
i 1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Istvan Varga
2005

ftchnls

ftchnls — Returns the number of channels in a stored function table.

Description

Returns the number of channels in a stored function table.

Syntax

ftchnls(x) (init-rate args only)

Performance

Returns the number of channels of a *GEN01* table, determined from the header of the original file. If the original file has no header or the table was not created by these GEN01, *ftchnls* returns -1.

Examples

Here is an example of the *ftchnls* opcode. It uses the file *ftchnls.csd* [examples/ftchnls.csd], and *mary.wav* [examples/mary.wav].

Example 208. Example of the *ftchnls* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ftchnls.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the number of channels in Table #1.
ichnls = ftchnls(1)
print ichnls
endin

</CsInstruments>
<CsScore>

; Table #1: Use an audio file, Csound will determine its size.
f 1 0 0 1 "mary.wav" 0 0 0

; Play Instrument #1 for 1 second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Since the audio file “mary.wav” is monophonic (1 channel), its output should include a line like this:

```
instr 1:  ichnls = 1.000
```

See Also

flen, ftlptim, ftsr, nsamp

Credits

Author: Chris McCormick
Perth, Australia
December 2001

Example written by Kevin Conder.

ftconv

ftconv — Low latency multichannel convolution, using a function table as impulse response source.

Description

Low latency multichannel convolution, using a function table as impulse response source. The algorithm is to split the impulse response to partitions of length determined by the 'iplen' parameter, and delay and mix partitions so that the original, full length impulse response is reconstructed without gaps. The output delay (latency) is 'iplen' samples, and does not depend on the control rate, unlike in the case of other convolve opcodes.

Syntax

```
a1[, a2[, a3[, ... a8]]] ftconv ain, ift, iplen[, iskip samples \  
[, iirlen[, iskipinit]]]
```

Initialization

ift -- source ftable number. The table is expected to contain interleaved multichannel audio data, with the number of channels equal to the number of output variables (a1, a2, etc.). An interleaved table can be created from a set of mono tables with GEN52.

iplen -- length of impulse response partitions, in sample frames; must be an integer power of two. Lower settings allow for shorter output delay, but will increase CPU usage.

iskipsamples (optional, defaults to zero) -- number of sample frames to skip at the beginning of the table. Useful for reverb responses that have some amount of initial delay. If this delay is not less than 'iplen' samples, then setting iskip samples to the same value as iplen will eliminate any additional latency by ftconv.

iirlen (optional) -- total length of impulse response, in sample frames. The default is to use all table data (not including the guard point).

iskipinit (optional, defaults to zero) -- if set to any non-zero value, skip initialization whenever possible without causing an error.

Performance

ain -- input signal.

a1 ... a8 -- output signal(s).

Example

Here is an example of the ftconv opcode. It uses the file *ftconv.csd* [examples/ftconv.csd].

Example 209. Example of the ftconv opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in
```

```

-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o ftconv.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr             = 48000
ksmps         = 32
nchnls        = 2
0dbfs         = 1

garvb         init 0
gaW           init 0
gaX           init 0
gaY           init 0

itmp          ftgen 1, 0, 64, -2, 2, 40, -1, -1, -1, 123, \
              1, 13.000, 0.05, 0.85, 20000.0, 0.0, 0.50, 2, \
              1, 2.000, 0.05, 0.85, 20000.0, 0.0, 0.25, 2, \
              1, 16.000, 0.05, 0.85, 20000.0, 0.0, 0.35, 2, \
              1, 9.000, 0.05, 0.85, 20000.0, 0.0, 0.35, 2, \
              1, 12.000, 0.05, 0.85, 20000.0, 0.0, 0.35, 2, \
              1, 8.000, 0.05, 0.85, 20000.0, 0.0, 0.35, 2

itmp          ftgen 2, 0, 262144, -2, 0
spat3dt       2, -0.2, 1, 0, 1, 1, 2, 0.005

itmp          ftgen 3, 0, 262144, -52, 3, 2, 0, 4, 2, 1, 4, 2, 2, 4

instr 1
a1            vco2 1, 440, 10
kfrq          port 100, 0.008, 20000
a1            butterlp a1, kfrq
a2            linseg 0, 0.003, 1, 0.01, 0.7, 0.005, 0, 1, 0
a1            = a1 * a2 * 2
denorm a1
vincr garvb, a1
aw, ax, ay, az spat3di a1, p4, p5, p6, 1, 1, 2
vincr gaW, aw
vincr gaX, ax
vincr gaY, ay

endin

instr 2
denorm garvb
; skip as many samples as possible without truncating the IR
arW, arX, arY ftconv garvb, 3, 2048, 2048, (65536 - 2048)
aW            = gaW + arW
aX            = gaX + arX
aY            = gaY + arY
garvb         = 0
gaW           = 0
gaX           = 0
gaY           = 0

aWre, aWim     hilbert aW
aXre, aXim     hilbert aX
aYre, aYim     hilbert aY
aWXr          = 0.0928*aXre + 0.4699*aWre
aWXiYr        = 0.2550*aXim - 0.1710*aWim + 0.3277*aYre
aL            = aWXr + aWXiYr
aR            = aWXr - aWXiYr

outs aL, aR

endin

</CsInstruments>
<CsScore>

i 1 0 0.5 0.0 2.0 -0.8
i 1 1 0.5 1.4 1.4 -0.6
i 1 2 0.5 2.0 0.0 -0.4
i 1 3 0.5 1.4 -1.4 -0.2
i 1 4 0.5 0.0 -2.0 0.0
i 1 5 0.5 -1.4 -1.4 0.2
i 1 6 0.5 -2.0 0.0 0.4
i 1 7 0.5 -1.4 1.4 0.6
i 1 8 0.5 0.0 2.0 0.8
i 2 0 10
e

</CsScore>
</CsoundSynthesizer>

```

See also

pconvolve, *convolve*, *convolve*.

Credits

Author: Istvan Varga
2005

ftfree

ftfree — Deletes function table.

Description

Deletes function table.

Syntax

```
ftfree ifno, iwhen
```

Initialization

ifno -- the number of the table to be deleted

iwhen -- if zero the table is deleted at init time; otherwise the table number is registered for being deleted at note deactivation.

Credits

Authors: Steven Yi, Istvan Varga
2005

ftgen

ftgen — Generate a score function table from within the orchestra.

Description

Generate a score function table from within the orchestra.

Syntax

```
gir ftgen ifn, itime, isize, igen, iarga [, iargb ] [...]
```

Initialization

gir -- either a requested or automatically assigned table number above 100.

ifn -- requested table number If *ifn* is zero, the number is assigned automatically and the value placed in *gir*. Any other value is used as the table number

itime -- is ignored, but otherwise corresponds to p2 in the score *f statement*.

isize -- table size. Corresponds to p3 of the score *f statement*.

igen -- function table *GEN* routine. Corresponds to p4 of the score *f statement*.

iarga, *iargb*, ... -- function table arguments. Correspond to p5 through *pn* of the score *f statement*.

Performance

This is equivalent to table generation in the score with the *f statement*.



Note

Csound was originally designed to support tables with power of two sizes only. Though this has changed in recent versions (you can use any size by using a negative number), many opcodes will not accept them.



Warning

Although Csound will not protest if ftgen is used inside instr-endin statements, this is not the intended or supported use, and must be handled with care as it has global effects. (In particular, a different size usually leads to relocation of the table, which may cause a crash or otherwise erratic behaviour.

Examples

Here is an example of the ftgen opcode. It uses the file *ftgen.csd* [examples/ftgen.csd].

Example 210. Example of the ftgen opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ftgen.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Table #1, a sine wave using the GEN10 routine.
gitemp ftgen 1, 0, 16384, 10, 1

; Instrument #1 - a basic oscillator.
instr 1
  kamp = 10000
  kcps = 440
  ; Use Table #1.
  ifn = 1

  al oscil kamp, kcps, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the ftgen opcode. It uses the file *ftgen-2.csd* [examples/ftgen-2.csd].

Example 211. Example of the ftgen opcode.

This example queries a file for its length to create an f-table of the appropriate size.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ftgen-2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr      = 48000
ksmps   = 16
nchnls  = 2

;Example by Jonathan Murphy 2007

0dbfs   = 1

instr 1

Sfile    = "beats.wav"

ilen     filelen Sfile ; Find length
isr      filesr  Sfile  ; Find sample rate

isamps   = ilen * isr ; Total number of samples
isize    init      1

loop:
  isize   = isize * 2
```

```
; Loop until isize is greater than number of samples
if (isize < isamps) igoto loop

    itab      ftgen      0, 0, isize, 1, Sfile, 0, 0, 0
              print     isize
              print     isamps

turnoff
endin

</CsInstruments>
<CsScore>
i1 0 10
e
</CsScore>
</CsoundSynthesizer>
```

See also

GEN routine overview

Credits

Author: Barry L. Vercoe
M.I.T., Cambridge, Mass
1997

Example written by Kevin Conder.

Added warning April 2002 by Rasmus Ekman

ftgentmp

ftgentmp — Generate a score function table from within the orchestra, which is deleted at the end of the note.

Description

Generate a score function table from within the orchestra, which is optionally deleted at the end of the note.

Syntax

```
ifno ftgentmp ip1, ip2dummy, isize, igen, iarga, iargb, ...
```

Initialization

ifno -- either a requested or automatically assigned table number above 100.

ip1 -- the number of the table to be generated or 0 if the number is to be assigned, in which case the table is deleted at the end of the note activation.

ip2dummy -- ignored.

isize -- table size. Corresponds to p3 of the score *f statement*.

igen -- function table *GEN* routine. Corresponds to p4 of the score *f statement*.

iarga, iargb, ... -- function table arguments. Correspond to p5 through *pn* of the score *f statement*.



Note

Csound was originally designed to support tables with power of two sizes only. Though this has changed in recent versions (you can use any size by using a negative number), many opcodes will not accept them.

Credits

Authors: Istvan Varga
2005

ftlen

ftlen — Returns the size of a stored function table.

Description

Returns the size of a stored function table.

Syntax

```
ftlen(x) (init-rate args only)
```

Performance

Returns the size (number of points, excluding guard point) of stored function table, number *x*. While most units referencing a stored table will automatically take its size into account (so tables can be of arbitrary length), this function reports the actual size if that is needed. Note that *ftlen* will always return a power-of-2 value, i.e. the function table guard point (see *f Statement*) is not included. As of Csound version 3.53, *ftlen* works with deferred function tables (see *GEN01*).

ftlen differs from *nsamp* in that *nsamp* gives the number of sample frames loaded, while *ftlen* gives the total number of samples without the guard point. For example, with a stereo sound file of 10000 samples, *ftlen()* would return 19999 (i.e. a total of 20000 mono samples, not including a guard point), but *nsamp()* returns 10000.

Examples

Here is an example of the *ftlen* opcode. It uses the file *ftlen.csd* [examples/ftlen.csd], and *mary.wav* [examples/mary.wav].

Example 212. Example of the *ftlen* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o ftlen.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the size of Table #1.
; The size will be the number of points excluding the guard point.
ilen = ftlen(1)
print ilen
endin

</CsInstruments>
<CsScore>
```

```
; Table #1: Use an audio file, Csound will determine its size.  
f 1 0 0 1 "mary.wav" 0 0 0  
  
; Play Instrument #1 for 1 second.  
i 1 0 1  
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

The audio file “mary.wav” is 154390 samples long. The `flen` opcode reports it as 154389 samples long because it reserves 1 point for the guard point. Its output should include a line like this:

```
instr 1:   ilen = 154389.000
```

See Also

ftchnls, ftlptim, ftsr, nsamp

Credits

Author: Barry L. Vercoe
MIT
Cambridge, Massachusetts
1997

Example written by Kevin Conder.

ftload

ftload — Load a set of previously-allocated tables from a file.

Description

Load a set of previously-allocated tables from a file.

Syntax

```
ftload "filename", iflag, ifn1 [, ifn2] [...]
```

Initialization

"filename" -- A quoted string containing the name of the file to load.

iflag -- Type of the file to load/save. (0 = binary file, Non-zero = text file)

ifn1, *ifn2*, ... -- Numbers of tables to load.

Performance

ftload loads a list of tables from a file. (The tables have to be already allocated though.) The file's format can be binary or text.



Warning

The file's format is not compatible with a WAV-file and is not endian-safe.

Examples

See the example for *ftsave*.

See Also

ftloadk, *ftsavek*, *ftsave*

Credits

Author: Gabriel Maldonado

New in version 4.21

ftloadk

ftloadk — Load a set of previously-allocated tables from a file.

Description

Load a set of previously-allocated tables from a file.

Syntax

```
ftloadk "filename", ktrig, iflag, ifn1 [, ifn2] [...]
```

Initialization

"filename" -- A quoted string containing the name of the file to load.

iflag -- Type of the file to load/save. (0 = binary file, Non-zero = text file)

ifn1, *ifn2*, ... -- Numbers of tables to load.

Performance

ktrig -- The trigger signal. Load the file each time it is non-zero.

ftloadk loads a list of tables from a file. (The tables have to be already allocated though.) The file's format can be binary or text. Unlike *ftload*, the loading operation can be repeated numerous times within the same note by using a trigger signal.



Warning

The file's format is not compatible with a WAV-file and is not endian-safe.

See Also

ftload, *ftsavek*, *ftsave*

Credits

Author: Gabriel Maldonado

New in version 4.21

ftlptim

ftlptim — Returns the loop segment start-time of a stored function table number.

Description

Returns the loop segment start-time of a stored function table number.

Syntax

`ftlptim(x)` (init-rate args only)

Performance

Returns the loop segment start-time (in seconds) of stored function table number *x*. This reports the duration of the direct recorded attack and decay parts of a sound sample, prior to its looped segment. Returns zero (and a warning message) if the sample does not contain loop points.

Examples

Here is an example of the ftlptim opcode. It uses the file *ftlptim.csd* [examples/ftlptim.csd], and *mary.wav* [examples/mary.wav].

Example 213. Example of the ftlptim opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ftlptim.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the loop-segment start time in Table #1.
itim = ftlptim(1)
print itim
endin

</CsInstruments>
<CsScore>

; Table #1: Use an audio file, Csound will determine its size.
f 1 0 0 1 "mary.wav" 0 0 0

; Play Instrument #1 for 1 second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Since the audio file “mary.wav” is non-looping, its output should include lines like this:

```
WARNING: non-looping sample
instr 1:  itim = 0.000
```

See Also

ftchnls, filen, ftsr, nsamp

Credits

Author: Barry L. Vercoe
MIT
Cambridge, Massachusetts
1997

Example written by Kevin Conder.

ftmorf

ftmorf — Morphs between multiple ftables as specified in a list.

Description

Uses an index into a table of ftable numbers to morph between adjacent tables in the list. This morphed function is written into the table referenced by *iresfn* on every k-cycle.

Syntax

```
ftmorf kftndx, iftfn, iresfn
```

Initialization

iftfn -- The ftable function. The list of values are expected to be pre-existing ftable numbers.

iresfn -- Table number of the morphed function

The length of all the tables in *iftfn* must equal the length of *iresfn*.

Performance

kftndx -- the index into the *iftfn* table.

If *iftfn* contains (6, 4, 6, 8, 7, 4):

- *kftndx*=4 will write the contents of f7 into *iresfn*.
- *kftndx*=4.5 will write the average of the contents of f7 and f4 into *iresfn*.

Examples

Here is an example of the ftmorf opcode. It uses the file *ftmorf.csd* [examples/ftmorf.csd].

Example 214. Example of the ftmorf opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o ftmorf.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 1
kndx line 0, p3, 7
      ftmorf kndx, 1, 2
```

```
asig      oscili 30000, 440, 2
          out   asig
endin

</CsInstruments>
<CsScore>

f1 0 8 -2 3 4 5 6 7 8 9 10
f2 0 1024 10 1 /*contents of f2 dont matter */
f3 0 1024 10 1
f4 0 1024 10 0 1
f5 0 1024 10 0 0 1
f6 0 1024 10 0 0 0 1
f7 0 1024 10 0 0 0 0 1
f8 0 1024 10 0 0 0 0 0 1
f9 0 1024 10 0 0 0 0 0 0 1
f10 0 1024 10 1 1 1 1 1 1 1 1

i1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: William “Pete” Moss
University of Texas at Austin
Austin, Texas USA
Jan. 2002

New in version 4.18

ftsave

ftsave — Save a set of previously-allocated tables to a file.

Description

Save a set of previously-allocated tables to a file.

Syntax

```
ftsave "filename", iflag, ifn1 [, ifn2] [...]
```

Initialization

"filename" -- A quoted string containing the name of the file to save.

iflag -- Type of the file to save. (0 = binary file, Non-zero = text file)

ifn1, *ifn2*, ... -- Numbers of tables to save.

Performance

ftsave saves a list of tables to a file. The file's format can be binary or text.



Warning

The file's format is not compatible with a WAV-file and is not endian-safe.

Examples

Here is an example of the ftsave opcode. It uses the file *ftsave.csd* [examples/ftsave.csd].

Example 215. Example of the ftsave opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o ftsave.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Table #1, make a sine wave using the GEN10 routine.
gitmp1 ftgen 1, 0, 32768, 10, 1
; Table #2, create an empty table.
gitmp2 ftgen 2, 0, 32768, 7, 0, 32768, 0

; Instrument #1 - a basic oscillator.
instr 1
```

```
kamp = 20000
kcps = 440
; Use Table #1.
ifn = 1

al oscil kamp, kcps, ifn
out al
endin

; Instrument #2 - Load Table #1 into Table #2.
instr 2
; Save Table #1 to a file called "table1.ftsav".
ftsav "table1.ftsav", 0, 1

; Load the "table1.ftsav" file into Table #2.
ftload "table1.ftsav", 0, 2

kamp = 20000
kcps = 440
; Use Table #2, it should contain Table #1's sine wave now.
ifn = 2

al oscil kamp, kcps, ifn
out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 1 second.
i 1 0 1
; Play Instrument #2 for 1 second.
i 2 2 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

ftloadk, ftload, ftsavek

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

New in version 4.21

ftsavek

ftsavek — Save a set of previously-allocated tables to a file.

Description

Save a set of previously-allocated tables to a file.

Syntax

```
ftsavek "filename", ktrig, iflag, ifn1 [, ifn2] [...]
```

Initialization

"filename" -- A quoted string containing the name of the file to save.

iflag -- Type of the file to save. (0 = binary file, Non-zero = text file)

ifn1, *ifn2*, ... -- Numbers of tables to save.

Performance

ktrig -- The trigger signal. Save the file each time it is non-zero.

ftsavek saves a list of tables to a file. The file's format can be binary or text. Unlike *ftsave*, the saving operation can be repeated numerous times within the same note by using a trigger signal.



Warning

The file's format is not compatible with a WAV-file and is not endian-safe.

See Also

ftloadk, *ftload*, *ftsave*

Credits

Author: Gabriel Maldonado

New in version 4.21

ftsr

ftsr — Returns the sampling-rate of a stored function table.

Description

Returns the sampling-rate of a stored function table.

Syntax

```
ftsr(x) (init-rate args only)
```

Performance

Returns the sampling-rate of a *GEN01* generated table. The sampling-rate is determined from the header of the original file. If the original file has no header or the table was not created by these *GEN01*, *ftsr* returns 0. New in Csound version 3.49.

Examples

Here is an example of the *ftsr* opcode. It uses the file *ftsr.csd* [examples/ftsr.csd], and *mary.wav* [examples/mary.wav].

Example 216. Example of the *ftsr* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ftsr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the sampling rate of Table #1.
isr = ftsr(1)
print isr
endin

</CsInstruments>
<CsScore>

; Table #1: Use an audio file.
f 1 0 262144 1 "mary.wav" 0 0 0

; Play Instrument #1 for 1 second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Since the audio file “mary.wav” uses a 44.1 Khz sampling rate, its output should a line like this:

```
instr 1:  isr = 44100.000
```

See Also

fchnls, flen, flptim, nsamp

Credits

Author: Gabriel Maldonado
Italy
October 1998

Example written by Kevin Conder.

gain

gain — Adjusts the amplitude audio signal according to a root-mean-square value.

Description

Adjusts the amplitude audio signal according to a root-mean-square value.

Syntax

```
ares gain asig, krms [, ihp] [, iskip]
```

Initialization

ihp (optional, default=10) -- half-power point (in Hz) of a special internal low-pass filter. The default value is 10.

iskip (optional, default=0) -- initial disposition of internal data space (see *reson*). The default value is 0.

Performance

asig -- input audio signal

gain provides an amplitude modification of *asig* so that the output *ares* has rms power equal to *krms*. *rms* and *gain* used together (and given matching *ihp* values) will provide the same effect as *balance*.

Examples

```
asrc buzz      10000,440, sr/440, 1 ; band-limited pulse train
a1  reson      asrc, 1000,100      ; sent through
a2  reson      a1,3000,500         ; 2 filters
afin balance a2, asrc              ; then balanced with source
```

See Also

balance, *rms*

gainslider

gainslider — An implementation of a logarithmic gain curve which is similar to the `gainslider~` object from Cycling 74 Max / MSP.

Description

This opcode is intended for use to multiply by an audio signal to give a console mixer like feel. There is no bounds in the source code so you can for example give higher than 127 values for extra amplitude but possibly clipped audio.

Syntax

`kout gainslider kindex`

Performance

kindex -- Index value. Nominal range from 0-127. For example a range of 0-152 will give you a range from -# to +18.0 dB.

kout -- Scaled output.

Examples

Here is an example of the `gainslider` opcode. It uses the file `gainslider.csd` [examples/gainslider.csd].

Example 217. Example of the gainslider opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  Silent
-odac       -idac      -d      ;;realtime output
</CsOptions>
<CsInstruments>

sr = 48000
ksmps = 100
nchnls = 2

/*--- */

instr 1 ; gainslider test

; uncomment for realtime midi
;kmod ctrl7 1, 1, 0, 127

; uncomment for non realtime
km0d phasor 1/10
kmod scale km0d, 127, 0

kout gainslider kmod

printk2 kmod
printk2 kout

aout diskio "soundfile.aiff", 1, 0, 1

aout = aout*kout
```

```
outs aout, aout

    endin

/*--- */
</CsInstruments>
<CsScore>

i1 0 8888

e
</CsScore>
</CsoundSynthesizer>
```

See Also

scale, logcurve, expcurve

Credits

Author: David Akbari
October
2006

gauss

gauss — Gaussian distribution random number generator.

Description

Gaussian distribution random number generator. This is an x-class noise generator.

Syntax

```
ares gauss krange
```

```
ires gauss krange
```

```
kres gauss krange
```

Performance

krange -- the range of the random numbers (*-krange* to *+krange*). Outputs both positive and negative numbers.

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the gauss opcode. It uses the file *gauss.csd* [examples/gauss.csd].

Example 218. Example of the gauss opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gauss.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number between -1 and 1.
; krange = 1

il gauss 1
```

```
    print i1
  endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 0.252
```

See Also

seed, betarand, bexprnd, cauchy, exprand, linrand, pcauchy, poisson, trirand, unirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder.

Existed in 3.30

gbuzz

gbuzz — Output is a set of harmonically related cosine partials.

Description

Output is a set of harmonically related cosine partials.

Syntax

```
ares gbuzz xamp, xcps, knh, klh, kmul, ifn [, iphs]
```

Initialization

ifn -- table number of a stored function containing a cosine wave. A large table of at least 8192 points is recommended.

iphs (optional, default=0) -- initial phase of the fundamental frequency, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is zero

Performance

The buzz units generate an additive set of harmonically related cosine partials of fundamental frequency *xcps*, and whose amplitudes are scaled so their summation peak equals *xamp*. The selection and strength of partials is determined by the following control parameters:

knh -- total number of harmonics requested. If *knh* is negative, the absolute value is used. If *knh* is zero, a value of 1 is used.

klh -- lowest harmonic present. Can be positive, zero or negative. In *gbuzz* the set of partials can begin at any partial number and proceeds upwards; if *klh* is negative, all partials below zero will reflect as positive partials without phase change (since cosine is an even function), and will add constructively to any positive partials in the set.

kmul -- specifies the multiplier in the series of amplitude coefficients. This is a power series: if the *klh*th partial has a strength coefficient of A, the (*klh* + n)th partial will have a coefficient of A * (*kmul* ** n), i.e. strength values trace an exponential curve. *kmul* may be positive, zero or negative, and is not restricted to integers.

buzz and *gbuzz* are useful as complex sound sources in subtractive synthesis. *buzz* is a special case of the more general *gbuzz* in which *klh* = *kmul* = 1; it thus produces a set of *knh* equal-strength harmonic partials, beginning with the fundamental. (This is a band-limited pulse train; if the partials extend to the Nyquist, i.e. *knh* = int (sr / 2 / fundamental freq.), the result is a real pulse train of amplitude *xamp*.)

Although both *knh* and *klh* may be varied during performance, their internal values are necessarily integer and may cause “pops” due to discontinuities in the output. *kmul*, however, can be varied during performance to good effect. *gbuzz* can be amplitude- and/or frequency-modulated by either control or audio signals.

N.B. This unit has its analog in *GENII*, in which the same set of cosines can be stored in a function table for sampling by an oscillator. Although computationally more efficient, the stored pulse train has a fixed spectral content, not a time-varying one as above.

Examples

Here is an example of the `gbuzz` opcode. It uses the file `gbuzz.csd` [examples/gbuzz.csd].

Example 219. Example of the `gbuzz` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o gbuzz.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 20000
  kcps = 440
  knh = 3
  klh = 2
  kmul = 0.7
  ifn = 1

  al gbuzz kamp, kcps, knh, klh, kmul, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a simple cosine waveform.
f 1 0 16384 11 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

buzz

Credits

Example written by Kevin Conder.

September 2003. Thanks to Kanata Motohashi for correcting the mentions of the *kmul* parameter.

getcfg

getcfg — Return Csound settings.

Description

Return various configuration settings in Svalue as a string at init time.

Syntax

Svalue **getcfg** iopt

Initialization

iopt -- The parameter to be returned, can be one of:

- 1: the maximum length of string variables in characters; this is at least the value of the -+max_str_len command line option - 1
- 2: the input sound file name (-i), or empty if there is no input file
- 3: the output sound file name (-o), or empty if there is no output file
- 4: return "1" if real time audio input or output is being used, and "0" otherwise
- 5: return "1" if running in beat mode (-t command line option), and "0" otherwise
- 6: the host operating system name
- 7: return "1" if a callback function for the chnrecv and chnsend opcodes has been set, and "0" otherwise (which means these opcodes do nothing)

Credits

Author: Istvan Varga
2006

New in version 5.02

gogobel

gogobel — Audio output is a tone related to the striking of a cow bell or similar.

Description

Audio output is a tone related to the striking of a cow bell or similar. The method is a physical model developed from Perry Cook, but re-coded for Csound.

Syntax

```
ares gogobel kamp, kfreq, ihrd, ipos, imp, kvibf, kvamp, ivfn
```

Initialization

ihrd -- the hardness of the stick used in the strike. A range of 0 to 1 is used. 0.5 is a suitable value.

ipos -- where the block is hit, in the range 0 to 1.

imp -- a table of the strike impulses. The file *marmstk1.wav* [examples/marmstk1.wav] is a suitable function from measurements and can be loaded with a *GEN01* table. It is also available at <ftp://ftp.cs.bath.ac.uk/pub/dream/documentation/sounds/modelling/>.

ivfn -- shape of vibrato, usually a sine table, created by a function.

Performance

A note is played on a cowbell-like instrument, with the arguments as below.

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kvibf -- frequency of vibrato in Hertz. Suggested range is 0 to 12

kvamp -- amplitude of the vibrato

Examples

Here is an example of the gogobel opcode. It uses the file *gogobel.csd* [examples/gogobel.csd], and *marmstk1.wav* [examples/marmstk1.wav],

Example 220. Example of the gogobel opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o gogobel.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
```

```
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; kamp = 31129.60
; kfreq = 440
; ihrd = 0.5
; ipos = 0.561
; imp = 1
; kvibf = 6.0
; kvamp = 0.3
; ivfn = 2

a1 gogobel 31129.60, 440, 0.5, 0.561, 1, 6.0, 0.3, 2
out a1
endin

</CsInstruments>
<CsScore>

; Table #1, the "marmstk1.wav" audio file.
f 1 0 256 1 "marmstk1.wav" 0 0 0
; Table #2, a sine wave for the vibrato.
f 2 0 128 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

goto

goto — Transfer control on every pass.

Description

Transfer control to *label* on every pass. (Combination of *igoto* and *kgoto*)

Syntax

```
goto label
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

Examples

Here is an example of the goto opcode. It uses the file *goto.csd* [examples/goto.csd].

Example 221. Example of the goto opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o goto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  al oscil 10000, 440, 1
  goto playit

  ; The goto will go to the playit label.
  ; It will skip any code in between like this comment.

playit:
  out al
endin

</CsInstruments>
<CsScore>

; Table #1: a simple sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```


See Also

cggoto, cigoto, ckgoto, if, igoto, kgoto, tigoto, timeout

Credits

Example written by Kevin Conder.

Added a note by Jim Aikin.

grain

grain — Generates granular synthesis textures.

Description

Generates granular synthesis textures.

Syntax

```
ares grain xamp, xpitch, xdens, kampoff, kpitchoff, kgdur, igfn, \  
      iwfn, imgdur [, igrnd]
```

Initialization

igfn -- The ftable number of the grain waveform. This can be just a sine wave or a sampled sound.

iwfn -- Ftable number of the amplitude envelope used for the grains (see also *GEN20*).

imgdur -- Maximum grain duration in seconds. This is the biggest value to be assigned to *kgdur*.

igrnd (optional) -- if non-zero, turns off grain offset randomness. This means that all grains will begin reading from the beginning of the *igfn* table. If zero (the default), grains will start reading from random *igfn* table positions.

Performance

xamp -- Amplitude of each grain.

xpitch -- Grain pitch. To use the original frequency of the input sound, use the formula:

$$\text{sndsr} / \text{ftlen}(\text{igfn})$$

where *sndsr* is the original sample rate of the *igfn* sound.

xdens -- Density of grains measured in grains per second. If this is constant then the output is synchronous granular synthesis, very similar to *fof*. If *xdens* has a random element (like added noise), then the result is more like asynchronous granular synthesis.

kampoff -- Maximum amplitude deviation from *xamp*. This means that the maximum amplitude a grain can have is *xamp* + *kampoff* and the minimum is *xamp*. If *kampoff* is set to zero then there is no random amplitude for each grain.

kpitchoff -- Maximum pitch deviation from *xpitch* in Hz. Similar to *kampoff*.

kgdur -- Grain duration in seconds. The maximum value for this should be declared in *imgdur*. If *kgdur* at any point becomes greater than *imgdur*, it will be truncated to *imgdur*.

The grain generator is based primarily on work and writings of Barry Truax and Curtis Roads.

Examples

This example generates a texture with gradually shorter grains and wider amp and pitch spread. It uses the file *grain.csd* [examples/grain.csd], and *mary.wav* [examples/mary.wav].

Example 222. Example of the grain opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o grain.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

instr 1
  insnd = 10
  ibasfrq = 44100 / ftlen(insnd)  ; Use original sample rate of insnd file

  kamp   expseg 220, p3/2, 600, p3/2, 220
  kpitch line ibasfrq, p3, ibasfrq * .8
  kdens  line 600, p3, 200
  kaoff  line 0, p3, 5000
  kpoff  line 0, p3, ibasfrq * .5
  kgdur  line .4, p3, .1
  imaxgdur = .5

  ar grain kamp, kpitch, kdens, kaoff, kpoff, kgdur, insnd, 5, imaxgdur, 0.0
  out ar
endin

</CsInstruments>
<CsScore>

f5 0 512 20 2          ; Hanning window
f10 0 262144 1  "mary.wav" 0 0 0
i1 0 6
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Paris Smaragdis
MIT
May 1997

grain2

grain2 — Easy-to-use granular synthesis texture generator.

Description

Generate granular synthesis textures. *grain2* is simpler to use, but *grain3* offers more control.

Syntax

```
ares grain2 kcps, kfmd, kgdur, iovrlp, kfn, iwfn [, irpow] \  
      [, iseed] [, imode]
```

Initialization

iovrlp -- (fixed) number of overlapping grains.

iwfn -- function table containing window waveform (Use GEN20 to calculate *iwfn*).

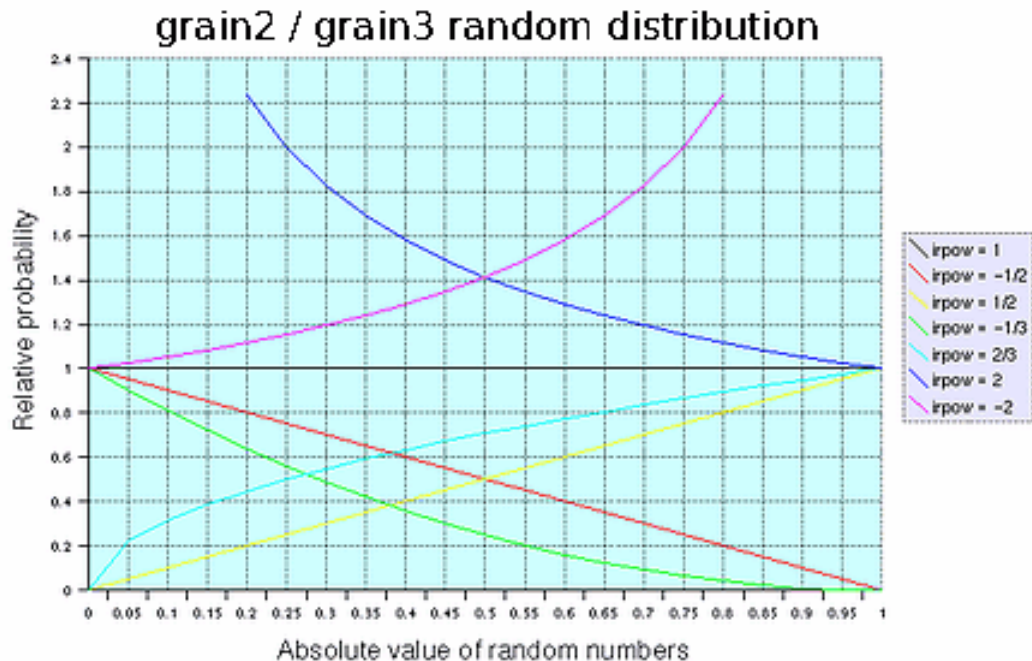
irpow (optional, default=0) -- this value controls the distribution of grain frequency variation. If *irpow* is positive, the random distribution (x is in the range -1 to 1) is

$$\text{abs}(x) ^ ((1 / \text{irpow}) - 1);$$

for negative *irpow* values, it is

$$(1 - \text{abs}(x)) ^ ((-1 / \text{irpow}) - 1)$$

Setting *irpow* to -1, 0, or 1 will result in uniform distribution (this is also faster to calculate). The image below shows some examples for *irpow*. The default value of *irpow* is 0.



A graph of distributions for different values of *irpow*.

iseed (optional, default=0) -- seed value for random number generator (positive integer in the range

1 to 2147483646 ($2^{31} - 2$)). Zero or negative value seeds from current time (this is also the default).

imode (optional default=0) -- sum of the following values:

- 8: interpolate window waveform (slower).
- 4: do not interpolate grain waveform (fast, but lower quality).
- 2: grain frequency is continuously modified by *kcps* and *kfmd* (by default, each grain keeps the frequency it was launched with). This may be slower at high control rates.
- 1: skip initialization.

Performance

ares -- output signal.

kcps -- grain frequency in Hz.

kfmd -- random variation (bipolar) in grain frequency in Hz.

kgdur -- grain duration in seconds. *kgdur* also controls the duration of already active grains (actually the speed at which the window function is read). This behavior does not depend on the *imode* flags.

kfn -- function table containing grain waveform. Table number can be changed at k-rate (this is useful to select from a set of band-limited tables generated by GEN30, to avoid aliasing).



Note

grain2 internally uses the same random number generator as *rnd31*. So reading *its documentation* is also recommended.

Examples

Here is an example of the *grain2* opcode. It uses the file *grain2.csd* [examples/grain2.csd].

Example 223. Example of the *grain2* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o grain2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 48000
kr = 750
ksmps = 64
nchnls = 2

/* square wave */
i_ ftgen 1, 0, 4096, 7, 1, 2048, 1, 0, -1, 2048, -1
/* window */
i_ ftgen 2, 0, 16384, 7, 0, 4096, 1, 4096, 0.3333, 8192, 0
/* sine wave */
i_ ftgen 3, 0, 1024, 10, 1
/* room parameters */
i_ ftgen 7, 0, 64, -2, 4, 50, -1, -1, -1, 11, \
```

```
1, 26.833, 0.05, 0.85, 10000, 0.8, 0.5, 2, \
1, 1.753, 0.05, 0.85, 5000, 0.8, 0.5, 2, \
1, 39.451, 0.05, 0.85, 7000, 0.8, 0.5, 2, \
1, 33.503, 0.05, 0.85, 7000, 0.8, 0.5, 2, \
1, 36.151, 0.05, 0.85, 7000, 0.8, 0.5, 2, \
1, 29.633, 0.05, 0.85, 7000, 0.8, 0.5, 2

ga01 init 0

/* generate bandlimited square waves */

i0 = 0
loop1:
imaxh = sr / (2 * 440.0 * exp (log(2.0) * (i0 - 69) / 12))
i_ ftgen i0 + 256, 0, 4096, -30, 1, 1, imaxh
i0 = i0 + 1
    if (i0 < 127.5) igoto loop1

    instr 1

p3 = p3 + 0.2

/* note velocity */
iamp = 0.0039 + p5 * p5 / 16192
/* vibrato */
kcps oscili 1, 8, 3
kenv linseg 0, 0.05, 0, 0.1, 1, 1, 1
/* frequency */
kcps = (kcps * kenv * 0.01 + 1) * 440 * exp(log(2) * (p4 - 69) / 12)
/* grain ftable */
kfn = int(256 + 69 + 0.5 + 12 * log(kcps / 440) / log(2))
/* grain duration */
kgdur port 100, 0.1, 20
kgdur = kgdur / kcps

a1 grain2 kcps, kcps * 0.02, kgdur, 50, kfn, 2, -0.5, 22, 2
a1 butterlp a1, 3000
a2 grain2 kcps, kcps * 0.02, 4 / kcps, 50, kfn, 2, -0.5, 23, 2
a2 butterbp a2, 12000, 8000
a2 butterbp a2, 12000, 8000
aenv1 linseg 0, 0.01, 1, 1, 1
aenv2 linseg 3, 0.05, 1, 1, 1
aenv3 linseg 1, p3 - 0.2, 1, 0.07, 0, 1, 0

a1 = aenv1 * aenv3 * (a1 + a2 * 0.7 * aenv2)

ga01 = ga01 + a1 * 10000 * iamp

    endin

/* output instr */

    instr 81

i1 = 0.000001
aLl, aLh, aRl, aRh spat3di ga01 + i1*i1*i1*i1, 3.0, 4.0, 0.0, 0.5, 7, 4
ga01 = 0
aLl butterlp aLl, 800.0
aRl butterlp aRl, 800.0

    outs aLl + aLh, aRl + aRh

    endin

</CsInstruments>
<CsScore>

t 0 60

i 1 0.0 1.3 60 127
i 1 2.0 1.3 67 127
i 1 4.0 1.3 64 112
i 1 4.0 1.3 72 112

i 81 0 6.4

e

</CsScore>
</CsoundSynthesizer>
```

See Also

grain3

Credits

Author: Istvan Varga

New in version 4.15

Updated April 2002 by Istvan Varga

grain3

grain3 — Generate granular synthesis textures with more user control.

Description

Generate granular synthesis textures. *grain2* is simpler to use but *grain3* offers more control.

Syntax

```
ares grain3 kcps, kphs, kfmd, kpm, kgdur, kdens, imaxovr, kfn, iwfn, \  
      kfpow, kprpow [, iseed] [, imode]
```

Initialization

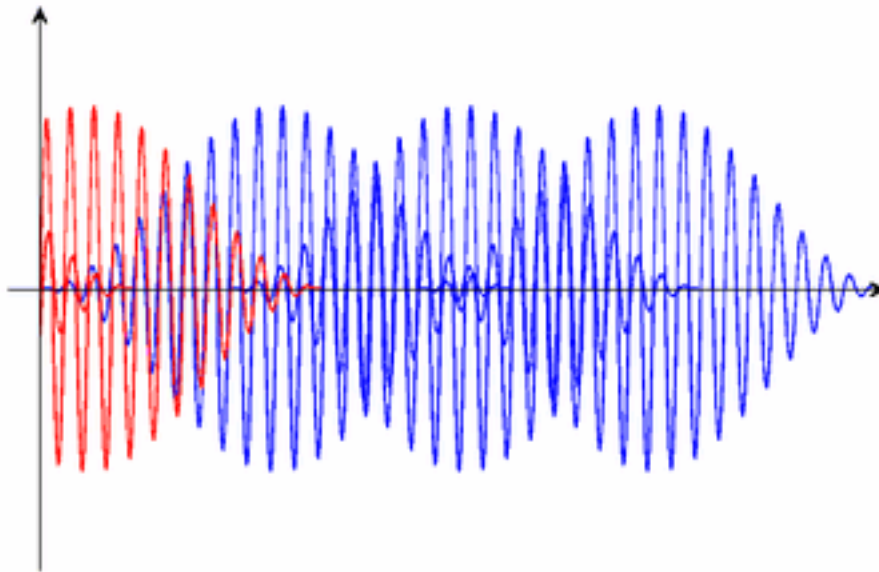
imaxovr -- maximum number of overlapping grains. The number of overlaps can be calculated by (*kdens* * *kgdur*); however, it can be overestimated at no cost in rendering time, and a single overlap uses (depending on system) 16 to 32 bytes of memory.

iwfn -- function table containing window waveform (Use GEN20 to calculate *iwfn*).

iseed (optional, default=0) -- seed value for random number generator (positive integer in the range 1 to 2147483646 ($2^{31} - 2$)). Zero or negative value seeds from current time (this is also the default).

imode (optional, default=0) -- sum of the following values:

- 64: synchronize start phase of grains to *kcps*.
- 32: start all grains at integer sample location. This may be faster in some cases, however it also makes the timing of grain envelopes less accurate.
- 16: do not render grains with start time less than zero. (see the image below; this option turns off grains marked with red on the image).
- 8: interpolate window waveform (slower).
- 4: do not interpolate grain waveform (fast, but lower quality).
- 2: grain frequency is continuously modified by *kcps* and *kfmd* (by default, each grain keeps the frequency it was launched with). This may be slower at high control rates. It also controls phase modulation (*kphs*).
- 1: skip initialization.



A diagram showing grains with a start time less than zero in red.

Performance

ares -- output signal.

kcps -- grain frequency in Hz.

kphs -- grain phase. This is the location in the grain waveform table, expressed as a fraction (between 0 to 1) of the table length.

kfmd -- random variation (bipolar) in grain frequency in Hz.

kpmf -- random variation (bipolar) in start phase.

kgdur -- grain duration in seconds. *kgdur* also controls the duration of already active grains (actually the speed at which the window function is read). This behavior does not depend on the *imode* flags.

kdens -- number of grains per second.

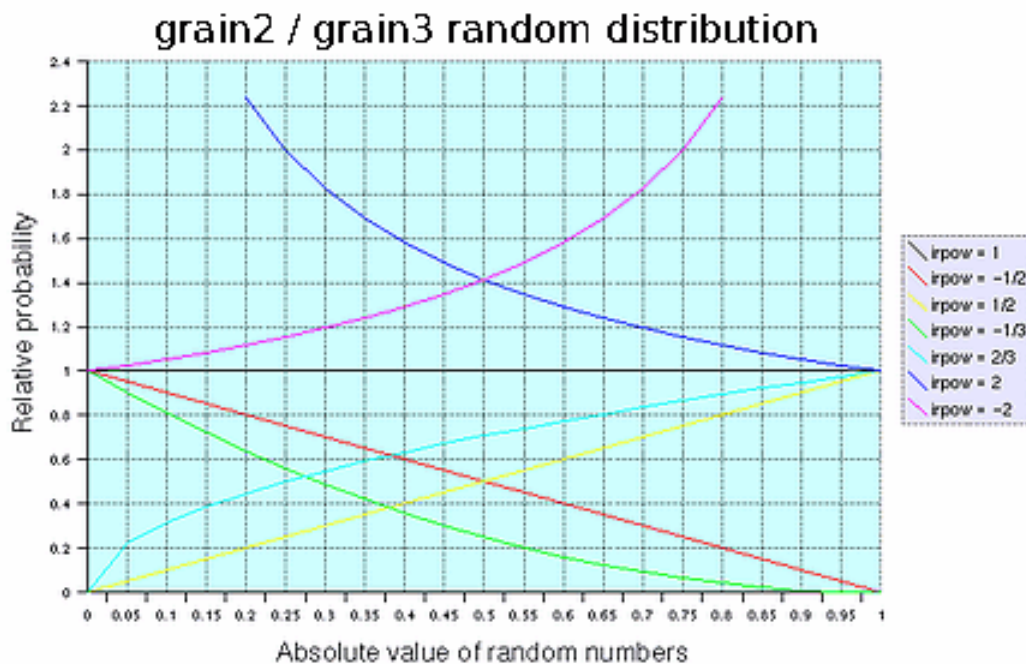
kfrpow -- this value controls the distribution of grain frequency variation. If *kfrpow* is positive, the random distribution (*x* is in the range -1 to 1) is

$\text{abs}(x)^{((1 / \text{kfrpow}) - 1)}$

; for negative *kfrpow* values, it is

$(1 - \text{abs}(x))^{((-1 / \text{kfrpow}) - 1)}$

Setting *kfrpow* to -1, 0, or 1 will result in uniform distribution (this is also faster to calculate). The image below shows some examples for *kfrpow*. The default value of *kfrpow* is 0.



A graph of distributions for different values of *kfrpow*.

kprpow -- distribution of random phase variation (see *kfrpow*). Setting *kphs* and *kpmid* to 0.5, and *kprpow* to 0 will emulate *grain2*.

kfn -- function table containing grain waveform. Table number can be changed at k-rate (this is useful to select from a set of band-limited tables generated by GEN30, to avoid aliasing).



Note

grain3 internally uses the same random number generator as *rnd31*. So reading its *documentation* is also recommended.

Examples

Here is an example of the *grain3* opcode. It uses the file *grain3.csd* [examples/grain3.csd].

Example 224. Example of the *grain3* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o grain3.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 48000
kr = 1000
ksmps = 48
nchnls = 1

/* Bartlett window */
itmp fngen 1, 0, 16384, 20, 3, 1
```

```
/* sawtooth wave */
itmp ftgen 2, 0, 16384, 7, 1, 16384, -1
/* sine */
itmp ftgen 4, 0, 1024, 10, 1
/* window for "soft sync" with 1/32 overlap */
itmp ftgen 5, 0, 16384, 7, 0, 256, 1, 7936, 1, 256, 0, 7936, 0
/* generate bandlimited sawtooth waves */
itmp ftgen 3, 0, 4096, -30, 2, 1, 2048
icnt = 0
loop01:
; 100 tables for 8 octaves from 30 Hz
ifrq = 30 * exp(log(2) * 8 * icnt / 100)
itmp ftgen icnt + 100, 0, 4096, -30, 3, 1, sr / (2 * ifrq)
icnt = icnt + 1
if (icnt < 99.5) igoto loop01
/* convert frequency to table number */
#define FRQ2FNUM(xout'xcps'xbsfn) #
$cout = int(($xbsfn) + 0.5 + (100 / 8) * log(($xcps) / 30) / log(2))
$cout limit $cout, $xbsfn, $xbsfn + 99

#

/* instr 1: pulse width modulated grains */

instr 1

kfrq = 523.25 ; frequency
$FRQ2FNUM(kfnum'kfrq'100) ; table number
kfmd = kfrq * 0.02 ; random variation in frequency
kgdur = 0.2 ; grain duration
kdens = 200 ; density
iseed = 1 ; random seed

kphs oscili 0.45, 1, 4 ; phase

a1 grain3 kfrq, 0, kfmd, 0.5, kgdur, kdens, 100, \
kfnum, 1, -0.5, 0, iseed, 2
a2 grain3 kfrq, 0.5 + kphs, kfmd, 0.5, kgdur, kdens, 100, \
kfnum, 1, -0.5, 0, iseed, 2

; de-click
aenv linseg 0, 0.01, 1, p3 - 0.05, 1, 0.04, 0, 1, 0

out aenv * 2250 * (a1 - a2)

endin

/* instr 2: phase variation */

instr 2

kfrq = 220 ; frequency
$FRQ2FNUM(kfnum'kfrq'100) ; table number
kgdur = 0.2 ; grain duration
kdens = 200 ; density
iseed = 2 ; random seed

kprdst expon 0.5, p3, 0.02 ; distribution

a1 grain3 kfrq, 0.5, 0, 0.5, kgdur, kdens, 100, \
kfnum, 1, 0, -kprdst, iseed, 64

; de-click
aenv linseg 0, 0.01, 1, p3 - 0.05, 1, 0.04, 0, 1, 0

out aenv * 1500 * a1

endin

/* instr 3: "soft sync" */

instr 3

kdens = 130.8 ; base frequency
kgdur = 2 / kdens ; grain duration

kfrq expon 880, p3, 220 ; oscillator frequency
$FRQ2FNUM(kfnum'kfrq'100) ; table number

a1 grain3 kfrq, 0, 0, 0, kgdur, kdens, 3, kfnum, 5, 0, 0, 0, 2
a2 grain3 kfrq, 0.667, 0, 0, kgdur, kdens, 3, kfnum, 5, 0, 0, 0, 2

; de-click
aenv linseg 0, 0.01, 1, p3 - 0.05, 1, 0.04, 0, 1, 0

out aenv * 10000 * (a1 - a2)
```

```
        endin

</CsInstruments>
<CsScore>

t 0 60
i 1 0 3
i 2 4 3
i 3 8 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

grain2

Credits

Author: Istvan Varga

New in version 4.15

Updated April 2002 by Istvan Varga

granule

granule — A more complex granular synthesis texture generator.

Description

The *granule* unit generator is more complex than *grain*, but does add new possibilities.

granule is a Csound unit generator which employs a wavetable as input to produce granularly synthesized audio output. Wavetable data may be generated by any of the GEN subroutines such as *GEN01* which reads an audio data file into a wavetable. This enable a sampled sound to be used as the source for the grains. Up to 128 voices are implemented internally. The maximum number of voices can be increased by redefining the variable MAXVOICE in the grain4.h file. *granule* has a build-in random number generator to handle all the random offset parameters. Thresholding is also implemented to scan the source function table at initialization stage. This facilitates features such as skipping silence passage between sentences.

The characteristics of the synthesis are controlled by 22 parameters. *xamp* is the amplitude of the output and it can be either audio rate or control rate variable.

Syntax

```
ares granule xamp, ivoice, iratio, imode, ithd, ifn, ipshift, igskip, \  
    igskip_os, ilength, kgap, igap_os, kgsz, igsz_os, iatt, idec \  
    [, iseed] [, ipitch1] [, ipitch2] [, ipitch3] [, ipitch4] [, ifnenv]
```

Initialization

ivoice -- number of voices.

iratio -- ratio of the speed of the gskip pointer relative to output audio sample rate. eg. 0.5 will be half speed.

imode -- +1 grain pointer move forward (same direction of the gskip pointer), -1 backward (oppose direction to the gskip pointer) or 0 for random.

ithd -- threshold, if the sampled signal in the wavetable is smaller then *ithd*, it will be skipped.

ifn -- function table number of sound source.

ipshift -- pitch shift control. If *ipshift* is 0, pitch will be set randomly up and down an octave. If *ipshift* is 1, 2, 3 or 4, up to four different pitches can be set amount the number of voices defined in *ivoice*. The optional parameters *ipitch1*, *ipitch2*, *ipitch3* and *ipitch4* are used to quantify the pitch shifts.

igskip -- initial skip from the beginning of the function table in sec.

igskip_os -- gskip pointer random offset in sec, 0 will be no offset.

ilength -- length of the table to be used starting from *igskip* in sec.

igap_os -- gap random offset in % of the gap size, 0 gives no offset.

igsz_os -- grain size random offset in % of grain size, 0 gives no offset.

iatt -- attack of the grain envelope in % of grain size.

idec -- decade of the grain envelope in % of grain size.

iseed (optional, default=0.5) -- seed for the random number generator.

ipitch1, *ipitch2*, *ipitch3*, *ipitch4* (optional, default=1) -- pitch shift parameter, used when *ipshift* is set to 1, 2, 3 or 4. Time scaling technique is used in pitch shift with linear interpolation between data points. Default value is 1, the original pitch.

ifnenv (optional, default=0) -- function table number to be used to generate the shape of the envelope.

Performance

xamp -- amplitude.

kgap -- gap between grains in sec.

kgsize -- grain size in sec.

Examples

Here is an example of the granule opcode. It uses the file *granule.csd* [examples/granule.csd], and *mary.wav* [examples/mary.wav].

Example 225. Example of the granule opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o granule.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
instr 1
;
k1      linseg 0,0.5,1,(p3-p2-1),1,0.5,0
a1      granule p4*k1,p5,p6,p7,p8,p9,p10,p11,p12,p13,p14,p15,\
        p16,p17,p18,p19,p20,p21,p22,p23,p24
a2      granule p4*k1,p5,p6,p7,p8,p9,p10,p11,p12,p13,p14,p15,\
        p16,p17,p18,p19, p20+0.17,p21,p22,p23,p24
outs a1,a2
endin

</CsInstruments>
<CsScore>

; f statement read sound file mary.wav in the SFDIR
; directory into f-table 1
f1      0 262144 1 "mary.wav" 0 0 0
i1      0 10 2000 64 0.5 0 0 1 4 0 0.005 5 0.01 50 0.02 50 30 30 0.39 \
        1 1.42 0.29 2
e

</CsScore>
</CsoundSynthesizer>
```

The above example reads a sound file called *mary.wav* into wavetable number 1 with 262,144 samples. It generates 10 seconds of stereo audio output using the wavetable. In the orchestra file, all

parameters required to control the synthesis are passed from the score file. A *linseg* function generator is used to generate an envelope with 0.5 second of linear attack and decay. Stereo effect is generated by using different seeds for the two *granule* function calls. In the example, 0.17 is added to p20 before passing into the second *granule* call to ensure that all of the random offset events are different from the first one.

In the score file, the parameters are interpreted as:

Parameter	Interpreted As
p5 (<i>ivoice</i>)	the number of voices is set to 64
p6 (<i>iratio</i>)	set to 0.5, it scans the wavetable at half of the speed of the audio output rate
p7 (<i>imode</i>)	set to 0, the grain pointer only move forward
p8 (<i>ithd</i>)	set to 0, skipping the thresholding process
p9 (<i>ifn</i>)	set to 1, function table number 1 is used
p10 (<i>ipshift</i>)	set to 4, four different pitches are going to be generated
p11 (<i>igskip</i>)	set to 0 and p12 (<i>igskip_os</i>) is set to 0.005, no skipping into the wavetable and a 5 mSec random offset is used
p13 (<i>ilength</i>)	set to 5, 5 seconds of the wavetable is to be used
p14 (<i>kgap</i>)	set to 0.01 and p15 (<i>igap_os</i>) is set to 50, 10 mSec gap with 50% random offset is to be used
p16 (<i>kgsz</i>)	set to 0.02 and p17 (<i>igsize_os</i>) is set to 50, 20 mSec grain with 50% random offset is used
p18 (<i>iatt</i>) and p19 (<i>idec</i>)	set to 30, 30% of linear attack and decade is applied to the grain
p20 (<i>iseed</i>)	seed for the random number generator is set to 0.39
p21 - p24	pitches set to 1 which is the original pitch, 1.42 which is a 5th up, 0.29 which is a 7th down and finally 2 which is an octave up.

Credits

Author: Allan Lee

Belfast

1996

New in version 3.35

guiro

guiro — Semi-physical model of a guiro sound.

Description

guiro is a semi-physical model of a guiro sound. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares guiro kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] [, ifreq1]
```

Initialization

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 128.

idamp (optional) -- the damping factor of the instrument. *Not used*.

imaxshake (optional, default=0) -- amount of energy to add back into the system. The value should be in range 0 to 1.

ifreq (optional) -- the main resonant frequency. The default value is 2500.

ifreq1 (optional) -- the first resonant frequency.

Performance

kamp -- Amplitude of output. Note: As these instruments are stochastic, this is only an approximation.

Examples

Here is an example of the guiro opcode. It uses the file *guiro.csd* [examples/guiro.csd].

Example 226. Example of the guiro opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o guiro.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 01 ;example of a guiro
```



```
a1 guiro p4, 0.01
   out a1
   endin

</CsInstruments>
<CsScore>

i1 0 1 20000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

bamboo, dripwater, sleighbells, tambourine

Credits

Author: Perry Cook, part of the PhISEM (Physically Informed Stochastic Event Modeling)
Adapted by John ffitch
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

harmon

harmon — Analyze an audio input and generate harmonizing voices in synchrony.

Description

Analyze an audio input and generate harmonizing voices in synchrony.

Syntax

```
ares harmon asig, kestfrq, kmaxvar, kgenfreq1, kgenfreq2, imode, \  
    iminfrq, iprd
```

Initialization

imode -- interpreting mode for the generating frequency inputs *kgenfreq1*, *kgenfreq2*. 0: input values are ratios with respect to the audio signal analyzed frequency. 1: input values are the actual requested frequencies in Hz.

iminfrq -- the lowest expected frequency (in Hz) of the audio input. This parameter determines how much of the input is saved for the running analysis, and sets a lower bound on the internal pitch tracker.

iprd -- period of analysis (in seconds). Since the internal pitch analysis can be time-consuming, the input is typically analyzed only each 20 to 50 milliseconds.

Performance

kestfrq -- estimated frequency of the input.

kmaxvar -- the maximum variance (expects a value between 0 and 1).

kgenfreq1 -- the first generated frequency.

kgenfreq2 -- the second generated frequency.

This unit is a harmonizer, able to provide up to two additional voices with the same amplitude and spectrum as the input. The input analysis is assisted by two things: an input estimated frequency *kestfrq* (in Hz), and a fractional maximum variance *kmaxvar* about that estimate which serves to limit the size of the search. Once the real input frequency is determined, the most recent pulse shape is used to generate the other voices at their requested frequencies.

The three frequency inputs can be derived in various ways from a score file or MIDI source. The first is the expected pitch, with a variance parameter allowing for inflections or inaccuracies; if the expected pitch is zero the harmonizer will be silent. The second and third pitches control the output frequencies; if either is zero the harmonizer will output only the non-zero request; if both are zero the harmonizer will be silent. When the requested frequency is higher than the input, the process requires additional computation due to overlapped output pulses. This is currently limited for efficiency reasons, with the result that only one voice can be higher than the input at any one time.

This unit is useful for supplying a background chorus effect on demand, or for correcting the pitch of a faulty input vocal. There is essentially no delay between input and output. Output includes only the generated parts, and does not include the input.

Examples

Here is an example of the harmon opcode. It uses the file *harmon.csd* [examples/harmon.csd].

Example 227. Example of the harmon opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o harmon.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; The frequency of the base note.
inote = 440

; Generate the base note.
avco vco 20000, inote, 1

kestfrq = inote
kmaxvar = 0.4

; Calculate frequencies 3 semitones above and
; below the base note.
kgenfreq1 = inote * semitone(3)
kgenfreq2 = inote * semitone(-3)

imode = 1
iminfrq = inote - 200
iprd = 0.1

; Generate the harmony notes.
al harmon avco, kestfrq, kmaxvar, kgenfreq1, kgenfreq2, \
    imode, iminfrq, iprd

out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Barry L. Vercoe
M.I.T., Cambridge, Mass
1997

New in version 3.47

Example written by Kevin Conder.

harmon2

harmon2 — Analyze an audio input and generate harmonizing voices in synchrony with formants preserved.

Description

Generate harmonizing voices with formants preserved.

Syntax

```
ares harmon2 asig, koct, kfrq1, kfrq2, icpsmode, ilowest[, ipolarity]
```

```
ares harmon3 asig, koct, kfrq1, \  
    kfrq2, kfrq3, icpsmode, ilowest[, ipolarity]
```

```
ares harmon4 asig, koct, kfrq1, \  
    kfrq2, kfrq3, kfrq4, icpsmode, ilowest[, ipolarity]
```

Initialization

icpsmode -- interpreting mode for the generating frequency inputs *kfrq1*, *kfrq2*, *kfrq3* and *kfrq4*: 0: input values are ratios w.r.t. the cps equivalent of *koct*. 1: input values are the actual requested frequencies in cps.

ilowest -- owest value of the koct input for which harmonizing voices will be generated.

ipolarity -- polarity of asig input, 1 = positive glottal pulses, 0 = negative. Default is 1.

Performance

Harmon2, **harmon3** and **harmon4** are high-performance harmonizers, able to provide up to four pitch-shifted copies of the input *asig* with spectral formants preserved. The pitch-shifting algorithm requires an accurate running estimate (*koct*, in decimal oct units) of the pitched content of *asig*, normally gained from an independent pitch tracker such as *specptrk*. The algorithm then isolates the most recent full pulse within *asig*, and uses this to generate the other voices at their required pulse rates.

If the frequency (or ratio) presented to *kfrq1*, *kfrq2*, *kfrq3* or *kfrq4* is zero, then no signal is generated for that voice. If any of them is non-zero, but the *koct* input is below the value *ilowest*, then that voice will output a direct copy of the input *asig*. As a consequence, the data arriving at the k-rate inputs can variously cause the generated voices to be turned on or off, to pass a direct copy of a non-voiced fricative source, or to harmonize the source according to some constructed algorithm. The transition from one mode to another is cross-faded, giving seamless alternating between voiced (harmonized) and non-voiced fricatives during spoken or sung input.

harmon2, *harmon3*, *harmon4* are especially matched to the output of *specptrk*. The latter generates pitch data in decimal octave format; it also emits its base value if no pitch is identified (as in fricative noise) and emits zero if the energy falls below a threshold, so that *harmon2*, *harmon3*, *harmon4* can be set to pass the direct signal in both cases. Of course, any other form of pitch estimation could also be used. Since pitch trackers usually incur a slight delay for accurate estimation (for *specptrk* the delay is printed by the spectrum unit), it is normal to delay the audio signal by the same amount so that *harmon2*, *harmon3*, *harmon4* can work from a fully concurrent estimate.

Examples

Here is an example of the harmon opcode. It uses the file *harmon.csd* [examples/harmon.csd].

Example 228. Example of the `harmon2` opcode.

```
a1,a2 ins                                ; get mic input
w1 spectrum      a1, .02, 7, 24, 12, 1, 3 ; and examine it
kocf,kamp specptrk      w1, 1, 6.5, 9.5, 7.5, 10, 7, .7, 0, 3, 1
a3 delay      a1, .065 ; allow for ptrk delay
a4 harmon2      a3, kocf, 1.25, 0.75, 0, 6.9 ; output a fixed 6-4 harmony
      outs      a3, a4 ; as well as the original
```

Credits

Author: Barry L. Vercoe
M.I.T., Cambridge, Mass
2006

New in version 5.04

hilbert

hilbert — A Hilbert transformer.

Description

An IIR implementation of a Hilbert transformer.

Syntax

```
ar1, ar2 hilbert asig
```

Performance

asig -- input signal

ar1 -- sine output of *asig*

ar2 -- cosine output of *asig*

hilbert is an IIR filter based implementation of a broad-band 90 degree phase difference network. The input to *hilbert* is an audio signal, with a frequency range from 15 Hz to 15 kHz. The outputs of *hilbert* have an identical frequency response to the input (i.e. they sound the same), but the two outputs have a constant phase difference of 90 degrees, plus or minus some small amount of error, throughout the entire frequency range. The outputs are in quadrature.

hilbert is useful in the implementation of many digital signal processing techniques that require a signal in phase quadrature. *ar1* corresponds to the cosine output of *hilbert*, while *ar2* corresponds to the sine output. The two outputs have a constant phase difference throughout the audio range that corresponds to the phase relationship between cosine and sine waves.

Internally, *hilbert* is based on two parallel 6th-order allpass filters. Each allpass filter implements a phase lag that increases with frequency; the difference between the phase lags of the parallel allpass filters at any given point is approximately 90 degrees.

Unlike an FIR-based Hilbert transformer, the output of *hilbert* does not have a linear phase response. However, the IIR structure used in *hilbert* is far more efficient to compute, and the nonlinear phase response can be used in the creation of interesting audio effects, as in the second example below.

Examples

The first example implements frequency shifting, or single sideband amplitude modulation. Frequency shifting is similar to ring modulation, except the upper and lower sidebands are separated into individual outputs. By using only one of the outputs, the input signal can be "detuned," where the harmonic components of the signal are shifted out of harmonic alignment with each other, e.g. a signal with harmonics at 100, 200, 300, 400 and 500 Hz, shifted up by 50 Hz, will have harmonics at 150, 250, 350, 450, and 550 Hz.

Here is the first example of the hilbert opcode. It uses the file *hilbert.csd* [examples/hilbert.csd], and *mary.wav* [examples/mary.wav].

Example 229. Example of the hilbert opcode implementing frequency shifting.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o hilbert.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 1
  idur = p3
  ; Initial amount of frequency shift.
  ; It can be positive or negative.
  ibegshift = p4
  ; Final amount of frequency shift.
  ; It can be positive or negative.
  iendshift = p5

  ; A simple envelope for determining the
  ; amount of frequency shift.
  kfreq linseg ibegshift, idur, iendshift

  ; Use the sound of your choice.
  ain soundin "mary.wav"

  ; Phase quadrature output derived from input signal.
  areal, aimag hilbert ain

  ; Quadrature oscillator.
  asin oscili 1, kfreq, 1
  acos oscili 1, kfreq, 1, .25

  ; Use a trigonometric identity.
  ; See the references for further details.
  amod1 = areal * acos
  amod2 = aimag * asin

  ; Both sum and difference frequencies can be
  ; output at once (sum is made up by the positive freqs, diff by the negative)
  ; aupshift corresponds to the positive freq side of ain, shifted up by kfreq.
  aupshift = (amod1 - amod2) * 0.7
  ; adownshift corresponds to the negative freq side of ain, also shifted up by kfreq.
  adownshift = (amod1 + amod2) * 0.7

  ; Notice that the adding of the two together is
  ; identical to the output of ring modulation.

  out aupshift
endin

</CsInstruments>
<CsScore>

; Sine table for quadrature oscillator.
f 1 0 16384 10 1

; Starting with no shift, ending with all
; frequencies shifted up by 200 Hz.
i 1 0 2 0 200

; Starting with no shift, ending with all
; frequencies shifted down by 200 Hz.
i 1 2 2 0 -200
e

</CsScore>
</CsoundSynthesizer>
```

The second example is a variation of the first, but with the output being fed back into the input. With very small shift amounts (i.e. between 0 and +6 Hz), the result is a sound that has been described as a “barberpole phaser” or “Shepard tone phase shifter.” Several notches appear in the spectrum, and are constantly swept in the direction opposite that of the shift, producing a filtering effect that is reminiscent of Risset’s “endless glissando”.

Here is the second example of the hilbert opcode. It uses the file *hilbert_barberpole.csd* [examples/hilbert_barberpole.csd], and *mary.wav* [examples/mary.wav].

Example 230. Example of the hilbert opcode sounding like a “barberpole phaser”.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadac     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o hilbert_barberpole.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
; kr must equal sr for the barberpole effect to work.
kr = 44100
ksmps = 1
nchnls = 2

; Instrument #1
instr 1
  idur = p3
  ibegshift = p4
  iendshift = p5

  ; sawtooth wave, not bandlimited
  asaw phasor 100
  ; add offset to center phasor amplitude between -.5 and .5
  asaw = asaw - .5
  ; sawtooth wave, with amplitude of 10000
  ain = asaw * 20000

  ; The envelope of the frequency shift.
  kfreq linseg ibegshift, idur, iendshift

  ; Phase quadrature output derived from input signal.
  areal, aimag hilbert ain

  ; The quadrature oscillator.
  asin oscili 1, kfreq, 1
  acos oscili 1, kfreq, 1, .25

  ; Based on trigonometric identities.
  amod1 = areal * acos
  amod2 = aimag * asin

  ; Calculate the up-shift and down-shift.
  aupshift = (amod1 + amod2) * 0.7
  adownshift = (amod1 - amod2) * 0.7

  ; Mix in the original signal to achieve the barberpole effect.
  amix1 = aupshift + ain
  amix2 = adownshift + ain

  ; Make sure the output doesn't get louder than the original signal.
  aout1 balance amix1, ain
  aout2 balance amix2, ain

  outs aout1, aout2
endin

</CsInstruments>
<CsScore>

; Table 1: A sine wave for the quadrature oscillator.
f 1 0 16384 10 1

; The score.
; p4 = frequency shifter, starting frequency.
; p5 = frequency shifter, ending frequency.
i 1 0 6 -10 10
e
```



```
</CsScore>  
</CsoundSynthesizer>
```

Technical History

The use of phase-difference networks in frequency shifters was pioneered by Harald Bode.¹ Bode and Bob Moog provide an excellent description of the implementation and use of a frequency shifter in the analog realm in;² this would be an excellent first source for those that wish to explore the possibilities of single sideband modulation. Bernie Hutchins provides more applications of the frequency shifter, as well as a detailed technical analysis.³ A recent paper by Scott Wardle⁴ describes a digital implementation of a frequency shifter, as well as some unique applications.

References

1. H. Bode, "Solid State Audio Frequency Spectrum Shifter." AES Preprint No. 395 (1965).
2. H. Bode and R.A. Moog, "A High-Accuracy Frequency Shifter for Professional Audio Applications." *Journal of the Audio Engineering Society*, July/August 1972, vol. 20, no. 6, p. 453.
3. B. Hutchins. *Musical Engineer's Handbook* (Ithaca, NY: Electronotes, 1975), ch. 6a.
4. S. Wardle, "A Hilbert-Transformer Frequency Shifter for Audio." Available online at <http://www.iaa.upf.es/dafx98/papers/>.

Credits

Author: Sean Costello
Seattle, Washington
1999

New in Csound version 3.55

The examples were updated April 2002. Thanks go to Sean Costello for fixing the barberpole example.

hrtfer

hrtfer — Creates 3D audio for two speakers.

Description

Output is binaural (headphone) 3D audio.

Syntax

```
aleft, aright hrtfer asig, kaz, kelev, "HRTFcompact"
```

Initialization

kAz -- azimuth value in degrees. Positive values represent position on the right, negative values are positions on the left.

kElev -- elevation value in degrees. Positive values represent position above horizontal, negative values are positions below horizontal.

At present, the only file which can be used with *hrtfer* is *HRTFcompact* [examples/HRTFcompact]. It must be passed to the opcode as the last argument within quotes as shown above.

HRTFcompact may also be obtained via anonymous ftp from:
<ftp://ftp.cs.bath.ac.uk/pub/dream/utilities/Analysis/HRTFcompact>

Performance

These unit generators place a mono input signal in a virtual 3D space around the listener by convolving the input with the appropriate HRTF data specified by the opcode's azimuth and elevation values. *hrtfer* allows these values to be k-values, allowing for dynamic spatialization. *hrtfer* can only place the input at the requested position because the HRTF is loaded in at i-time (remember that currently, CSound has a limit of 20 files it can hold in memory, otherwise it causes a segmentation fault). The output will need to be scaled either by using balance or by multiplying the output by some scaling constant.



Note

The sampling rate of the orchestra must be 44.1kHz. This is because 44.1kHz is the sampling rate at which the HRTFs were measured. In order to be used at a different rate, the HRTFs would need to be re-sampled at the desired rate.

Examples

Here is an example of the *hrtfer* opcode. It uses the file *hrtfer.csd* [examples/hrtfer.csd], *HRTFcompact* [examples/HRTFcompact], and *beats.wav* [examples/beats.wav].

Example 231. Example of the *hrtfer* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>
```

```
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-o dac        -i adc     -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o hrtfer.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

instr 1
  kaz          linseg 0, p3, -360 ; move the sound in circle
  kel          linseg -40, p3, 45 ; around the listener, changing
                                ; elevation as its turning

  asrc         soundin "beats.wav"
  aleft,aright hrtfer asrc, kaz, kel, "HRTFcompact"
  aleftscale   = aleft * 200
  arightscale  = aright * 200

  outs         aleftscale, arightscale
endin

</CsInstruments>
<CsScore>

i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

hrtfmove, hrtfmove2, hrtfstat.

Credits

Authors: Eli Breder and David MacIntyre
Montreal
1996

Fixed the example thanks to a message from Istvan Varga.

hrtfmove

hrtfmove — Generates dynamic 3d binaural audio for headphones using magnitude interpolation and phase truncation.

Description

This opcode takes a source signal and spatialises it in the 3 dimensional space around a listener by convolving the source with stored head related transfer function (HRTF) based filters.

Syntax

```
aleft, aright hrtfmove asrc, kAz, kElev, ifilel, ifiler [, imode, ifade, isr]
```

Initialization

Initialization

ifilel -- left HRTF spectral data file

ifiler -- right HRTF spectral data file



Note

Spectral datafiles (based on the MIT HRTF database) are available in 3 different sampling rates: 44.1, 48 and 96 khz and are labelled accordingly. Input and processing *sr* should match datafile *sr*. Files should be in the current directory or the SADIR (see *Environment Variables*).

imode -- optional, default 0 for phase truncation, 1 for minimum phase

ifade -- optional, number of processing buffers for phase change crossfade (default 8). Legal range is 1-24. A low value is recommended for complex sources (4 or less: a higher value may make the crossfade audible), a higher value (8 or more: a lower value may make the inconsistency when the filter changes phase values audible) for narrowband sources. Does not effect minimum phase processing.



Note

Ocasionally fades may overlap (when unnaturally fast/complex trajectories are requested). In this case, a warning will be printed. Use a smaller crossfade or slightly change trajectory to avoid any possible inconsistencies that may arise.

isr - optional, default 44.1kHz, legal values: 44100, 48000 and 96000.

kAz -- azimuth value in degrees. Positive values represent position on the right, negative values are positions on the left.

kElev -- elevation value in degrees. Positive values represent position above horizontal, negative values are positions below horizontal (min -40).

Artifact-free user-defined trajectories are made possible using an interpolation algorithm based on spectral magnitude interpolation and phase truncation. Crossfades are implemented to minimise/eliminate any inconsistencies caused by updating phase values. These crossfades are performed over a user definable number of convolution processing buffers. Complex sources may only need to

crossfade over 1 buffer; narrow band sources may need several. The opcode also offers minimum phase based processing, a more traditional and complex method. In this mode, the hrtf filters used are reduced to minimum phase representations and the interpolation process then uses the relationship between minimum phase magnitude and phase spectra. Interaural time difference, which is inherent to the phase truncation process, is reintroduced in the minimum phase process using variable delay lines.

Examples

Here is an example of the `hrtfmove` opcode. It uses the file `hrtfmove.csd` [examples/hrtfmove.csd].

Example 232. Example of the `hrtfmove` opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select flags here
; realtime audio out
; -o dac
; For Non-realtime ouput leave only the line below:
-o hrtf.wav
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

gasrc init 0

instr 1          ;a plucked string

    kamp = p4
    kcps = cspch(p5)
    icps = cspch(p5)

    al pluck kamp, kcps, icps, 0, 1

    gasrc = al

endin

instr 10 ;uses output from instr1 as source

    kaz linseg 0, p3, 720          ;2 full rotations

    aleft,aright hrtfmove gasrc, kaz,0, "hrtf-44100-left.dat","hrtf-44100-right.dat"

    outs aleft, aright

endin

</CsInstruments>
<CsScore>

; Play Instrument 1: a simple arpeggio
i1 0 .2 15000 8.00
i1 + .2 15000 8.04
i1 + .2 15000 8.07
i1 + .2 15000 8.11
i1 + .2 15000 9.02
i1 + 1.5 15000 8.11
i1 + 1.5 15000 8.07
i1 + 1.5 15000 8.04
i1 + 1.5 15000 8.00
i1 + 1.5 15000 7.09
i1 + 1.5 15000 8.00

; Play Instrument 10 for 10 seconds.
i10 0 10

</CsScore>
</CsoundSynthesizer>
```

See Also

hrtfmove2, hrtfstat, hrtfer.

Credits

Author: Brian Carty
Maynooth
2008

hrtfmove2

hrtfmove2 — Generates dynamic 3d binaural audio for headphones using a Woodworth based spherical head model with improved low frequency phase accuracy.

Description

This opcode takes a source signal and spatialises it in the 3 dimensional space around a listener using head related transfer function (HRTF) based filters.

Syntax

```
aleft, aright hrtfmove2 asrc, kAz, kElev, ifilel, ifiler [,ioverlap, iradius, isr]
```

Initialization

ifilel -- left HRTF spectral data file

ifiler -- right HRTF spectral data file



Note

Spectral datafiles (based on the MIT HRTF database) are available in 3 different sampling rates: 44.1, 48 and 96 khz and are labelled accordingly. Input and processing *sr* should match datafile *sr*. Files should be in the current directory or the SADIR (see *Environment Variables*).

ioverlap -- optional, number of overlaps for STFT processing (default 4). See STFT section of manual.

iradius -- optional, head radius used for phase spectra calculation in centimeters (default 9.0)

isr - optional, default 44.1kHz, legal values: 44100, 48000 and 96000.

Performance

asrc -- Input/source signal.

kAz -- azimuth value in degrees. Positive values represent position on the right, negative values are positions on the left.

kElev -- elevation value in degrees. Positive values represent position above horizontal, negative values are positions below horizontal (min -40).

Artifact-free user-defined trajectories are made possible using an interpolation algorithm based on spectral magnitude interpolation and a derived phase spectrum based on the Woodworth spherical head model. Accuracy is increased for the data set provided by extracting and applying a frequency dependent scaling factor to the phase spectra, leading to a more precise low frequency interaural time difference. Users can control head radius for the phase derivation, allowing a crude level of individualisation. The dynamic source version of the opcode uses a Short Time Fourier Transform algorithm to avoid artefacts caused by derived phase spectra changes. STFT processing means this opcode is more computationally intensive than *hrtfmove* using phase truncation, but phase is constantly updated by *hrtfmove2*.

Examples

Here is an example of the `hrtfmove2` opcode. It uses the file `hrtfmove2.csd` [examples/hrtfmove2.csd].

Example 233. Example of the `hrtfmove2` opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select flags here
; realtime audio out
; -o dac
; For Non-realtime ouput leave only the line below:
-o hrtf.wav
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

gasrc init 0

instr 1          ;a plucked string

    kamp = p4
    kcps = cpspch(p5)
    icps = cpspch(p5)

    al pluck kamp, kcps, icps, 0, 1

    gasrc = al

endin

instr 10 ;uses output from instr1 as source

    kaz linseg 0, p3, 720          ;2 full rotations

    aleft,aright hrtfmove2 gasrc, kaz,0, "hrtf-44100-left.dat","hrtf-44100-right.dat"

    outs aleft, aright

endin

</CsInstruments>
<CsScore>

; Play Instrument 1: a simple arpeggio
i1 0 .2 15000 8.00
i1 + .2 15000 8.04
i1 + .2 15000 8.07
i1 + .2 15000 8.11
i1 + .2 15000 9.02
i1 + 1.5 15000 8.11
i1 + 1.5 15000 8.07
i1 + 1.5 15000 8.04
i1 + 1.5 15000 8.00
i1 + 1.5 15000 7.09
i1 + 1.5 15000 8.00

; Play Instrument 10 for 10 seconds.
i10 0 10

</CsScore>
</CsoundSynthesizer>
```

See Also

hrtfmove, hrtfstat, hrtfer.

Credits

Author: Brian Carty

Maynooth
2008

hrtfstat

hrtfstat — Generates static 3d binaural audio for headphones using a Woodworth based spherical head model with improved low frequency phase accuracy.

Description

This opcode takes a source signal and spatialises it in the 3 dimensional space around a listener using head related transfer function (HRTF) based filters. It produces a static output (azimuth and elevation parameters are i-rate), because a static source allows much more efficient processing than *hrtfmove* and *hrtfmove2*..

Syntax

```
aleft, aright hrtfstat asrc, iAz, iElev, ifilel, ifiler [,iradius, isr]
```

Initialization

iAz -- azimuth value in degrees. Positive values represent position on the right, negative values are positions on the left.

iElev -- elevation value in degrees. Positive values represent position above horizontal, negative values are positions below horizontal (min -40).

ifilel -- left HRTF spectral data file

ifiler -- right HRTF spectral data file



Note

Spectral datafiles (based on the MIT HRTF database) are available in 3 different sampling rates: 44.1, 48 and 96 khz and are labelled accordingly. Input and processing sr should match datafile sr. Files should be in the current directory or the SADIR (see *Environment Variables*).

iradius -- optional, head radius used for phase spectra calculation in centimeters (default 9.0)

isr - optional (default 44.1kHz). Legal values are 44100, 48000 and 96000.

Performance

Artifact-free user-defined static spatialisation is made possible using an interpolation algorithm based on spectral magnitude interpolation and a derived phase based on the Woodworth spherical head model. Accuracy is increased for the data set provided by extracting and applying a frequency dependent scaling factor to the phase spectra, leading to a more precise low frequency interaural time difference. Users can control head radius for the phase derivation, allowing a crude level of individualisation. The static source version of the opcode uses overlap add convolution (it does not need STFT processing, see *hrtfmove2*), and is thus considerably more efficient than *hrtfmove2* or *hrtfmove*, but cannot generate moving sources.

Examples

It can be found in the file *htrfstat.csd* [examples/htrfstat.csd].

Example 234. Example of the hrtfstat opcode.

```
<CsoundSynthesizer>
<CsOptions>
  ; Select flags here
  ; realtime audio out
  ; -o dac
  ; For Non-realtime ouput leave only the line below:
  -o hrtf.wav

</CsOptions>
<CsInstruments>

  sr = 44100
  kr = 4410
  ksmps = 10
  nchnls = 2

  gasrc init 0

  instr 1          ;a plucked string

  kamp = p4
  kcps = cpspch(p5)
  icps = cpspch(p5)

  a1 pluck kamp, kcps, icps, 0, 1

  gasrc = a1

  endin

  instr 10 ;uses output from instr1 as source

  aleft,aright hrtfstat gasrc, 90,0, "hrtf-44100-left.dat","hrtf-44100-right.dat"

  outs aleft, aright

  endin

</CsInstruments>
<CsScore>

  ; Play Instrument 1: a plucked string
  i1 0 2 20000 8.00

  ; Play Instrument 10 for 2 seconds.
  i10 0 2

</CsScore>
</CsoundSynthesizer>
```

See Also

hrtfmove, hrtfmove2, hrtfer.

Credits

Author: Brian Carty
Maynooth
2008

hsboscil

hsboscil — An oscillator which takes tonality and brightness as arguments.

Description

An oscillator which takes tonality and brightness as arguments, relative to a base frequency.

Syntax

```
ares hsboscil kamp, ktone, kbrite, ibasfreq, iwfn, ioctfn \  
      [, ioctcnt] [, iphs]
```

Initialization

ibasfreq -- base frequency to which tonality and brightness are relative

iwfn -- function table of the waveform, usually a sine

ioctfn -- function table used for weighting the octaves, usually something like:

```
f1 0 1024 -19 1 0.5 270 0.5
```

ioctcnt (optional) -- number of octaves used for brightness blending. Must be in the range 2 to 10. Default is 3.

iphs (optional, default=0) -- initial phase of the oscillator. If *iphs* = -1, initialization is skipped.

Performance

kamp -- amplitude of note

ktone -- cyclic tonality parameter relative to *ibasfreq* in logarithmic octave, range 0 to 1, values > 1 can be used, and are internally reduced to *frac(ktone)*.

kbrite -- brightness parameter relative to *ibasfreq*, achieved by weighting *ioctcnt* octaves. It is scaled in such a way, that a value of 0 corresponds to the original value of *ibasfreq*, 1 corresponds to one octave above *ibasfreq*, -2 corresponds to two octaves below *ibasfreq*, etc. *kbrite* may be fractional.

hsboscil takes tonality and brightness as arguments, relative to a base frequency (*ibasfreq*). Tonality is a cyclic parameter in the logarithmic octave, brightness is realized by mixing multiple weighted octaves. It is useful when tone space is understood in a concept of polar coordinates.

Making *ktone* a line, and *kbrite* a constant, produces Risset's glissando.

Oscillator table *iwfn* is always read interpolated. Performance time requires about *ioctcnt* * *oscili*.

Examples

Here is an example of the hsboscil opcode. It uses the file *hsboscil.csd* [examples/hsboscil.csd].

Example 235. Example of the hsboscil opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o hsboscil.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; synth waveform
giwave ftgen 1, 0, 1024, 10, 1, 1, 1, 1
; blending window
giblend ftgen 2, 0, 1024, -19, 1, 0.5, 270, 0.5

; Instrument #1 - produces Risset's glissando.
instr 1
  kamp = 10000
  kbrite = 0.5
  ibasfreq = 200
  ioctcnt = 5

  ; Change ktone linearly from 0 to 1,
  ; over the period defined by p3.
  ktone line 0, p3, 1

  al hsboscil kamp, ktone, kbrite, ibasfreq, giwave, giblend, ioctcnt
  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the `hsboscil` opcode in a MIDI instrument. It uses the file `hsboscil_midi.csd` [examples/hsboscil_midi.csd].

Example 236. Example of the `hsboscil` opcode in a MIDI instrument.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages   MIDI in
-odac         -iadc      -d            -M0      ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o hsboscil_midi.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; synth waveform
giwave ftgen 1, 0, 1024, 10, 1, 1, 1, 1
; blending window
giblend ftgen 2, 0, 1024, -19, 1, 0.5, 270, 0.5
```

```
; Instrument #1 - use hsboscil in a MIDI instrument.
instr 1
  ibase = cpsoct(6)
  ioctcnt = 5

  ; all octaves sound alike.
  itona octmidi
  ; velocity is mapped to brightness
  ibrite ampmidi 3

  ; Map an exponential envelope for the amplitude.
  kenv expon 20000, 1, 100

  asig hsboscil kenv, itona, ibrite, ibase, giwave, giblend, ioctcnt
  out asig
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten minutes
i 1 0 600
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Peter Neubäcker
Munich, Germany
August, 1999

New in Csound version 3.58

hvs1

hvs1 — Allows one-dimensional Hyper Vectorial Synthesis (HVS) controlled by externally-updated k-variables.

Description

hvs1 allows one-dimensional Hyper Vectorial Synthesis (HVS) controlled by externally-updated k-variables.

Syntax

```
hvs1 kx, inumParms, inumPointsX, iOutTab, iPositionsTab, iSnapTab [, iConfigTab]
```

Initialization

inumParms - number of parameters controlled by the HVS. Each HVS snapshot is made up of *inumParms* elements.

inumPointsX - number of points that each dimension of the HVS cube (or square in case of two-dimensional HVS; or line in case of one-dimensional HVS) is made up.

iOutTab - number of the table receiving the set of output-parameter instant values of the HVS. The total amount of parameters is defined by the *inumParms* argument.

iPositionsTab – a table filled with the individual positions of snapshots in the HVS matrix (see below for more information).

iSnapTab – a table filled with all the snapshots. Each snapshot is made up of a set of parameter values. The amount of elements contained in each snapshots is specified by the *inumParms* argument. The set of elements of each snapshot follows (and is adjacent) to the previous one in this table. So the total size of this table should be \geq to *inumParms* multiplied the number of snapshots you intend to store for the HVS.

iConfigTab – (optional) a table containing the behavior of the HVS for each parameter. If the value of *iConfigTab* is zero (default), this argument is ignored, meaning that each parameter is treated with linear interpolation by the HVS. If *iConfigTab* is different than zero, then it must refer to an existing table whose contents are in its turn referring to a particular kind of interpolation. In this table, a value of -1 indicates that corresponding parameter is leaved unchanged (ignored) by the HVS; a value of zero indicates that corresponding parameter is treated with linear-interpolation; each other values must be integer numbers indicating an existing table filled with a shape which will determine the kind of special interpolation to be used (table-based interpolation).

Performance

kx - these are externally-modified variables which controls the motion of the pointer in the HVS matrix cube (or square or line in case of HVS matrices made up of less than 3 dimensions). The range of these input arguments must be 0 to 1.

Hyper Vectorial Synthesis is a technique that allows control of a huge set of parameters by using a simple and global approach. The key concepts of the HVS are:

The set of HVS parameters, whose amount is fixed and defined by the *inumParms* argument. During the HVS performance, all these parameters are variant and can be applied to any sound synthesis technique, as well as to any global control for algorithmic composition and any other kind of level. The user must previously define several sets of fixed values for each HVS parameter, each set corresponding to a determinate synthesis configuration. Each set of values is called snapshot, and can

be considered as the coordinates of a bound of a multi-dimensional space. The HVS consists on moving a point in this multi-dimensional space (by using a special motion pointer, see below), according and inside the bounds defined by the snapshots. You can fix any amount of HVS parameters (each parameter being a dimension of the multi-dimensional space), even a huge number, the limit only depends on the processing power (and the memory) of your computer and on the complexity of the sound-synthesis you will use.

The HVS cube (or square or line). This is the matrix (of 3, 2 or 1 dimensions, according to the hvs opcode you intend to use) of “mainstays” (or pivot) points of HVS. The total amount of pivot-points depends on the value of the *inumPointsX*, *inumPointsY* and *inumPointsZ* arguments. In the case of a 3-dimensional HVS matrix you can define, for instance, 3 points for the X dimension, 5 for the Y dimension and 2 for the Z dimension. In this case, the total number of pivot-points is $3 * 5 * 2 = 30$. With this set of pivot points, the cube is divided into smaller cubed zones each one bounded by eight nearby points. Each point is numbered. The numeral order of these points is established in the following way: number zero is the first point, number 1 the second and so on. Assuming you are using a 3-dimensional HVS cube having the number of points above mentioned (i.e. 3, 5 and 2 respectively for the X, Y and Z axis), the first point (point zero) is the upper-left-front vertex of the cube, by facing the XY plane of the cube. The second point is the middle point of the upper front edge of the cube and so on. You can refer to the figure below in order to understand how the numeral order of the pivot-points proceeds:

For the 2-dimensional HVS, it is the same, by only omitting the rear cube face, so each zone is bounded by 4 pivot-points instead of 8. For the 1-dimensional HVS, the whole thing is even simpler because it is a line with the pivot-points proceeding from left to right. Each point is coupled with a snapshot.

Snapshot order, as stored into the *iSnapTab*, can or cannot follow the order of the pivot-points numbers. In fact it is possible to alter this order by means the *iPositionsTab*, a table that remaps the position of each snapshot in relation to the pivot points. The *iPositionsTab* is made up of the positions of the snapshots (contained in the *iSnapTab*) in the two-dimensional grid. Each subsequent element is actually a pointer representing the position in the *iSnapTab*. For example, in a 2-dimensional HVS matrix such as the following (in this case having *inumPointsX* = 3 and *inumPointsY* = 5:

Table 10.

5	7	1
3	4	9
6	2	0
4	1	3
8	2	7

These numbers (to be stored in the *iSnapTab* table by using, for instance, the GEN02 function generator) represents the snapshot position within the grid (in this case a 3x5 matrix). So, the first element 5, has index zero and represents the 6th (element zero is the first) snapshot contained in the *iSnapTab*, the second element 7 represents the 8th element of *iSnapTab* and so on. Summing up, the vertices of each zone (a cubed zone is delimited by 8 vertices; a squared zone by 4 vertices and a linear zone by 2 points) are coupled with a determinate snapshot, whose number is remapped by the *iSnapTab*.

Output values of the HVS are influenced by the motion pointer, a point whose position, in the HVS cube (or square or segment) is determined by the *kx*, *ky* and *kz* arguments. The values of these arguments, which must be in the 0 to 1 range, are externally set by the user. The output values, whose amount is equal to the *inumParms* argument, are stored in the *iOutTab*, a table that must be already allocated by the user, and must be at least *inumParms* size. In what way the motion pointer influences the output? Well, when the motion pointer falls in a determinate cubed zone, delimited, for instance, by 8 vertices (or pivot points), we assume that each vertex has associated a different snapshot (i.e. a set of *inumParms* values), well, the output will be the weighted average value of the 8 vertices, calculated according on the distance of the motion pointer from each of the 8 vertices. In the case of a default behavior, when the *iConfigTab* argument is not set, the exact output is calculated by using linear interpolation which is applied to each different parameter of the HVS. Anyway,

it is possible to influence this behavior by setting the *iConfigTab* argument to a number of a table whose contents can affect one or more HVS parameters. The *iConfigTab* table elements are associated to each HVS parameter and their values affect the HVS output in the following way:

- If *iConfigTab* is equal to -1, corresponding output is skipped, i.e. the element is not calculated, leaving corresponding element value in the *iOutTab* unchanged;
- If *iConfigTab* is equal to zero, then the normal HVS output is calculated (by using weighted average of the nearest vertex of current zone where it falls the motion pointer);
- If *iConfigTab* element is equal to an integer number > zero, then the contents of a table having that number is used as a shape of a table-based interpolation.

Examples

Here is an example of the hvs1 opcode. It uses the file *hvs1.csd* [examples/hvs1.csd].

Example 237. Example of the hvs1 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr=44100
ksmps=100
nchnls=2

; Example by Gabriel Maldonado and Andres Cabrera

0dbfs = 1

ginumLinesX init 16
ginumParms  init 3

giOutTab ftgen 5,0,8, -2,      0
giPosTab ftgen 6,0,32, -2,      3,2,1,0,4,5,6,7,8,9,10, 11, 15, 14, 13, 12
giSnapTab ftgen 8,0,64, -2,      1,1,1,  2,0,0,  3,2,0,  2,2,2,  5,2,1,  2,3,4,  6,1,7,  0,0,0, \
                                1,3,5,  3,4,4,  1,5,8,  1,1,5,  4,3,2,  3,4,5,  7,6,5,  7,8,9
tb0_init giOutTab

gk1, ih1 FLpanel "hsv1", 500, 100, 10, 10, 0
FLslider "X", 0, 1, 0, 5, -1, 400, 30, 50, 20
FLpanel_end
FLrun

instr 1
;      kx,      inumParms, inumPointsX, iOutTab, iPosTab, iSnapTab [, iConfigTab]
hvs1    gk1,    ginumParms, ginumLinesX, giOutTab, giPosTab, giSnapTab ;, iConfigTab

k0 init 0
k1 init 1
k2 init 2

printk2 tb0(k0)
printk2 tb0(k1), 10
printk2 tb0(k2), 20

aosc1 oscil tb0(k0)/20, tb0(k1)*100 + 200, 1
aosc2 oscil tb0(k1)/20, tb0(k2)*100 + 200, 1
aosc3 oscil tb0(k2)/20, tb0(k0)*100 + 200, 1
aosc4 oscil tb0(k1)/20, tb0(k0)*100 + 200, 1
aosc5 oscil tb0(k2)/20, tb0(k1)*100 + 200, 1
aosc6 oscil tb0(k0)/20, tb0(k2)*100 + 200, 1

outs aosc1 + aosc2 + aosc3, aosc4 + aosc5 + aosc6
```

```
        endin

</CsInstruments>
<CsScore>

f1 0 1024 10 1
f0 3600
i1 0 3600

</CsScore>
</CsoundSynthesizer>
```

See Also

hvs2, hvs3, vphaseseg

Credits

Author: Gabriel Maldonado

New in version 5.06

hvs2

hvs2 — Allows two-dimensional Hyper Vectorial Synthesis (HVS) controlled by externally-updated k-variables.

Description

hvs2 allows two-dimensional Hyper Vectorial Synthesis (HVS) controlled by externally-updated k-variables.

Syntax

```
hvs2 kx, ky, inumParms, inumPointsX, iOutTab, iPositionsTab, iSnapTab [, iConfigTab]
```

Initialization

inumParms - number of parameters controlled by the HVS. Each HVS snapshot is made up of *inumParms* elements.

inumPointsX - number of points that each dimension of the HVS cube (or square in case of two-dimensional HVS; or line in case of one-dimensional HVS) is made up.

iOutTab - number of the table receiving the set of output-parameter instant values of the HVS. The total amount of parameters is defined by the *inumParms* argument.

iPositionsTab – a table filled with the individual positions of snapshots in the HVS matrix (see below for more information).

iSnapTab – a table filled with all the snapshots. Each snapshot is made up of a set of parameter values. The amount of elements contained in each snapshots is specified by the *inumParms* argument. The set of elements of each snapshot follows (and is adjacent) to the previous one in this table. So the total size of this table should be \geq to *inumParms* multiplied the number of snapshots you intend to store for the HVS.

iConfigTab – (optional) a table containing the behavior of the HVS for each parameter. If the value of *iConfigTab* is zero (default), this argument is ignored, meaning that each parameter is treated with linear interpolation by the HVS. If *iConfigTab* is different than zero, then it must refer to an existing table whose contents are in its turn referring to a particular kind of interpolation. In this table, a value of -1 indicates that corresponding parameter is leaved unchanged (ignored) by the HVS; a value of zero indicates that corresponding parameter is treated with linear-interpolation; each other values must be integer numbers indicating an existing table filled with a shape which will determine the kind of special interpolation to be used (table-based interpolation).

Performance

kx, *ky* - these are externally-modified variables which controls the motion of the pointer in the HVS matrix cube (or square or line in case of HVS matrices made up of less than 3 dimensions). The range of these input arguments must be 0 to 1.

Hyper Vectorial Synthesis is a technique that allows control of a huge set of parameters by using a simple and global approach. The key concepts of the HVS are:

The set of HVS parameters, whose amount is fixed and defined by the *inumParms* argument. During the HVS performance, all these parameters are variant and can be applied to any sound synthesis technique, as well as to any global control for algorithmic composition and any other kind of level. The user must previously define several sets of fixed values for each HVS parameter, each set corresponding to a determinate synthesis configuration. Each set of values is called snapshot, and can

be considered as the coordinates of a bound of a multi-dimensional space. The HVS consists on moving a point in this multi-dimensional space (by using a special motion pointer, see below), according and inside the bounds defined by the snapshots. You can fix any amount of HVS parameters (each parameter being a dimension of the multi-dimensional space), even a huge number, the limit only depends on the processing power (and the memory) of your computer and on the complexity of the sound-synthesis you will use.

The HVS cube (or square or line). This is the matrix (of 3, 2 or 1 dimensions, according to the hvs opcode you intend to use) of “mainstays” (or pivot) points of HVS. The total amount of pivot-points depends on the value of the *inumPointsX*, *inumPointsY* and *inumPointsZ* arguments. In the case of a 3-dimensional HVS matrix you can define, for instance, 3 points for the X dimension, 5 for the Y dimension and 2 for the Z dimension. In this case, the total number of pivot-points is $3 * 5 * 2 = 30$. With this set of pivot points, the cube is divided into smaller cubed zones each one bounded by eight nearby points. Each point is numbered. The numeral order of these points is established in the following way: number zero is the first point, number 1 the second and so on. Assuming you are using a 3-dimensional HVS cube having the number of points above mentioned (i.e. 3, 5 and 2 respectively for the X, Y and Z axis), the first point (point zero) is the upper-left-front vertex of the cube, by facing the XY plane of the cube. The second point is the middle point of the upper front edge of the cube and so on. You can refer to the figure below in order to understand how the numeral order of the pivot-points proceeds:

For the 2-dimensional HVS, it is the same, by only omitting the rear cube face, so each zone is bounded by 4 pivot-points instead of 8. For the 1-dimensional HVS, the whole thing is even simpler because it is a line with the pivot-points proceeding from left to right. Each point is coupled with a snapshot.

Snapshot order, as stored into the *iSnapTab*, can or cannot follow the order of the pivot-points numbers. In fact it is possible to alter this order by means the *iPositionsTab*, a table that remaps the position of each snapshot in relation to the pivot points. The *iPositionsTab* is made up of the positions of the snapshots (contained in the *iSnapTab*) in the two-dimensional grid. Each subsequent element is actually a pointer representing the position in the *iSnapTab*. For example, in a 2-dimensional HVS matrix such as the following (in this case having *inumPointsX* = 3 and *inumPointsY* = 5:

Table 11.

5	7	1
3	4	9
6	2	0
4	1	3
8	2	7

These numbers (to be stored in the *iSnapTab* table by using, for instance, the GEN02 function generator) represents the snapshot position within the grid (in this case a 3x5 matrix). So, the first element 5, has index zero and represents the 6th (element zero is the first) snapshot contained in the *iSnapTab*, the second element 7 represents the 8th element of *iSnapTab* and so on. Summing up, the vertices of each zone (a cubed zone is delimited by 8 vertices; a squared zone by 4 vertices and a linear zone by 2 points) are coupled with a determinate snapshot, whose number is remapped by the *iSnapTab*.

Output values of the HVS are influenced by the motion pointer, a point whose position, in the HVS cube (or square or segment) is determined by the *kx*, *ky* and *kz* arguments. The values of these arguments, which must be in the 0 to 1 range, are externally set by the user. The output values, whose amount is equal to the *inumParms* argument, are stored in the *iOutTab*, a table that must be already allocated by the user, and must be at least *inumParms* size. In what way the motion pointer influences the output? Well, when the motion pointer falls in a determinate cubed zone, delimited, for instance, by 8 vertices (or pivot points), we assume that each vertex has associated a different snapshot (i.e. a set of *inumParms* values), well, the output will be the weighted average value of the 8 vertices, calculated according on the distance of the motion pointer from each of the 8 vertices. In the case of a default behavior, when the *iConfigTab* argument is not set, the exact output is calculated by using linear interpolation which is applied to each different parameter of the HVS. Anyway,

it is possible to influence this behavior by setting the *iConfigTab* argument to a number of a table whose contents can affect one or more HVS parameters. The *iConfigTab* table elements are associated to each HVS parameter and their values affect the HVS output in the following way:

- If *iConfigTab* is equal to -1, corresponding output is skipped, i.e. the element is not calculated, leaving corresponding element value in the *iOutTab* unchanged;
- If *iConfigTab* is equal to zero, then the normal HVS output is calculated (by using weighted average of the nearest vertex of current zone where it falls the motion pointer);
- If *iConfigTab* element is equal to an integer number > zero, then the contents of a table having that number is used as a shape of a table-based interpolation.

Examples

Here is an example of the hvs2 opcode. It uses the file *hvs2.csd* [examples/hvs2.csd].

Example 238. Example of the hvs2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr=44100
ksmps=100
nchnls=2

0dbfs = 1

ginumLinesX init 4
ginumLinesY init 4
ginumParms init 3

giOutTab ftgen 5,0,8, -2,      0
giPosTab ftgen 6,0,32, -2,    3,2,1,0,4,5,6,7,8,9,10, 11, 15, 14, 13, 12
giSnapTab ftgen 8,0,64, -2,   1,1,1,  2,0,0,  3,2,0,  2,2,2,  5,2,1,  2,3,4,  6,1,7,  0,0,0, \
                             1,3,5,   3,4,4,   1,5,8,   1,1,5,   4,3,2,   3,4,5,   7,6,5,   7,8,9
tb0_init giOutTab

    FLpanel "Prova HVS2",600,400,10,100,0

gk1,    gk2,    ih1, ih2  FLjoy    "HVS controller XY", 0,    1,    1,    0,    0,    0,    -1,
; *ihandle,    *numlinesX,    *numlinesY, *iwidth, *iheight, *ix, *iy, *image;
gihandle FLhvsBox ginumLinesX,    ginumLinesY,    300,    300,    300,    50,    1
    FLpanel_end
    FLrun

instr 1

; Smooth control signals to avoid clicks
kx portk gk1, 0.02
ky portk gk2, 0.02

;          kx, ky, inumParms, inumlinesX, inumlinesY, iOutTab, iPosTab, iSnapTab [, iCon
hvs2 kx, ky, ginumParms, ginumLinesX, ginumLinesY, giOutTab, giPosTab, giSnapTab ;, iCon
;          *kx, *ky, *ihandle;
    FLhvsBoxSetValue gk1, gk2, gihandle

k0 init 0
k1 init 1
k2 init 2

printk2 tb0(k0)
printk2 tb0(k1), 10
printk2 tb0(k2), 20
```

```
kris init 0.003
kdur init 0.02
kdec init 0.007

; Make parameters of synthesis depend on the table values produced by hvs
ares1 fof 0.2, tb0(k0)*100 + 50, tb0(k1)*100 + 200, 0, tb0(k2) * 10 + 50, 0.003, 0.02, 0.007, 20, \
1, 2, p3
ares2 fof 0.2, tb0(k1)*100 + 50, tb0(k2)*100 + 200, 0, tb0(k0) * 10 + 50, 0.003, 0.02, 0.007, 20, \
1, 2, p3

outs ares1, ares2
endin

</CsInstruments>
<CsScore>

f 1 0 1024 10 1 ;Sine wave
f 2 0 1024 19 0.5 0.5 270 0.5 ;Grain envelope table

f0 3600
i1 0 3600

</CsScore>
</CsoundSynthesizer>
```

Here is second example of the hvs2 opcode. It uses the file *hvs2.csd* [examples/hvs2-2.csd].

Example 239. Example of the hvs2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out Audio in
-odac -iadc ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o hvs2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=48000
ksmps=100
nchnls=2

; Example by James Hearon 2008
; Edited by Andres Cabrera

ginumPointsX init 16
ginumPointsY init 16
ginumParms init 3

;Generate 9 tables with arbitrary points
gitmp ftgen 100, 0, 16, -2, 70, 260, 390, 180, 200, 300, 980, 126, \
330, 860, 580, 467, 220, 399, 1026, 1500
gitmp ftgen 200, 0, 16, -2, 100, 200, 300, 140, 600, 700, 880, 126, \
330, 560, 780, 167, 220, 999, 1026, 1500
gitmp ftgen 300, 0, 16, -2, 400, 200, 300, 540, 600, 700, 880, 126, \
330, 160, 780, 167, 820, 999, 1026, 1500
gitmp ftgen 400, 0, 16, -2, 100, 200, 800, 640, 600, 300, 880, 126, \
330, 660, 780, 167, 220, 999, 1026, 1500
gitmp ftgen 500, 0, 16, -2, 200, 200, 360, 440, 600, 700, 880, 126, \
330, 560, 380, 167, 220, 499, 1026, 1500
gitmp ftgen 600, 0, 16, -2, 100, 600, 300, 840, 600, 700, 880, 126, \
330, 260, 980, 367, 120, 399, 1026, 1500
gitmp ftgen 700, 0, 16, -2, 100, 200, 300, 340, 200, 500, 380, 126, \
330, 860, 780, 867, 120, 999, 1026, 1500
gitmp ftgen 800, 0, 16, -2, 100, 600, 300, 240, 200, 700, 880, 126, \
130, 560, 980, 167, 220, 499, 1026, 1500
gitmp ftgen 900, 0, 16, -2, 100, 800, 200, 140, 600, 700, 680, 126, \
330, 560, 780, 167, 120, 299, 1026, 1500

giOutTab ftgen 5,0,8, -2, 0
giPosTab ftgen 6,0,32, -2, 0,1,2,3,4,5,6,7,8,9,10, 11, 15, 14, 13, 12
giSnapTab ftgen 8,0,64, -2, 1,1,1, 2,0,0, 3,2,0, 2,2,2, \
5,2,1, 2,3,4, 6,1,7, 0,0,0, 1,3,5, 3,4,4, 1,5,8, 1,1,5, \
```

```
4,3,2, 3,4,5, 7,6,5, 7,8,9

tb0_init          giOutTab

                FLpanel "hsv2",440,100,10,10,0
gk1,ih1 FLslider "X", 0,1, 0, 5, -1, 400,20, 20,10
gk2, ih2 FLslider "Y", 0, 1, 0, 5, -1, 400, 20, 20, 50
                FLpanel_end

                FLpanel "hvsBox",280,280,500,1000,0
;ihandle FLhvsBox inumlinesX, inumlinesY, iwidth, iheight, ix, iy [, image]
gih1 FLhvsBox 16, 16, 250, 250, 10, 1
                FLpanel_end
                FLrun

                instr 1
FLhvsBoxSetValue gk1, gk2, gih1

hvs2      gk1,gk2,  ginumParms, ginumPointsX, ginumPointsY, giOutTab, giPosTab, giSnapTab  ;, iConfig

k0        init      0
k1        init      1
k2        init      2
kspeed    init      0

kspeed = int((tb0(k2)) + 1)*.10

kenv      oscil      25000, kspeed*16, 10

k1        phasor      kspeed ;slow phasor: 200 sec.
kpch      tableikt k1 * 16, int((tb0(k1)) +1)*100 ;scale phasor * length
a1        oscilikt kenv, kpch, int(tb0(k0)) +1000;scale pitch slightly
ahp      butterlp a1, 2500
outs      ahp, ahp

                endin

</CsInstruments>
<CsScore>

f 10 0 1024 20 5 ;use of windowing function
f1000 0 1024 10 .33 .25 .5
f1001 0 1024 10 1
f1002 0 1024 10 .5 .25 .05
f1003 0 1024 10 .05 .10 .3 .5 1
f1004 0 1024 10 1 .5 .25 .125 .625
f1005 0 1024 10 .33 .44 .55 .66
f1006 0 1024 10 1 1 1 1 1
f1007 0 1024 10 .05 .25 .05 .25 .05 1

f0 3600
i1 0 3600

</CsScore>
</CsoundSynthesizer>
```

See Also

hvs1, hvs3, vphaseseg

Credits

Author: Gabriel Maldonado

New in version 5.06

hvs3

hvs3 — Allows three-dimensional Hyper Vectorial Synthesis (HVS) controlled by externally-updated k-variables.

Description

hvs3 allows three-dimensional Hyper Vectorial Synthesis (HVS) controlled by externally-updated k-variables.

Syntax

```
hvs3 kx, ky, kz, inumParms, inumPointsX, iOutTab, iPositionsTab, iSnapTab [, iConfigTab]
```

Initialization

inumParms - number of parameters controlled by the HVS. Each HVS snapshot is made up of *inumParms* elements.

inumPointsX - number of points that each dimension of the HVS cube (or square in case of two-dimensional HVS; or line in case of one-dimensional HVS) is made up.

iOutTab - number of the table receiving the set of output-parameter instant values of the HVS. The total amount of parameters is defined by the *inumParms* argument.

iPositionsTab – a table filled with the individual positions of snapshots in the HVS matrix (see below for more information).

iSnapTab – a table filled with all the snapshots. Each snapshot is made up of a set of parameter values. The amount of elements contained in each snapshots is specified by the *inumParms* argument. The set of elements of each snapshot follows (and is adjacent) to the previous one in this table. So the total size of this table should be \geq to *inumParms* multiplied the number of snapshots you intend to store for the HVS.

iConfigTab – (optional) a table containing the behavior of the HVS for each parameter. If the value of *iConfigTab* is zero (default), this argument is ignored, meaning that each parameter is treated with linear interpolation by the HVS. If *iConfigTab* is different than zero, then it must refer to an existing table whose contents are in its turn referring to a particular kind of interpolation. In this table, a value of -1 indicates that corresponding parameter is leaved unchanged (ignored) by the HVS; a value of zero indicates that corresponding parameter is treated with linear-interpolation; each other values must be integer numbers indicating an existing table filled with a shape which will determine the kind of special interpolation to be used (table-based interpolation).

Performance

kx, *ky*, *kz* - these are externally-modified variables which controls the motion of the pointer in the HVS matrix cube (or square or line in case of HVS matrices made up of less than 3 dimensions). The range of these input arguments must be 0 to 1.

Hyper Vectorial Synthesis is a technique that allows control of a huge set of parameters by using a simple and global approach. The key concepts of the HVS are:

The set of HVS parameters, whose amount is fixed and defined by the *inumParms* argument. During the HVS performance, all these parameters are variant and can be applied to any sound synthesis technique, as well as to any global control for algorithmic composition and any other kind of level. The user must previously define several sets of fixed values for each HVS parameter, each set corresponding to a determinate synthesis configuration. Each set of values is called snapshot, and can

be considered as the coordinates of a bound of a multi-dimensional space. The HVS consists on moving a point in this multi-dimensional space (by using a special motion pointer, see below), according and inside the bounds defined by the snapshots. You can fix any amount of HVS parameters (each parameter being a dimension of the multi-dimensional space), even a huge number, the limit only depends on the processing power (and the memory) of your computer and on the complexity of the sound-synthesis you will use.

The HVS cube (or square or line). This is the matrix (of 3, 2 or 1 dimensions, according to the hvs opcode you intend to use) of “mainstays” (or pivot) points of HVS. The total amount of pivot-points depends on the value of the *inumPointsX*, *inumPointsY* and *inumPointsZ* arguments. In the case of a 3-dimensional HVS matrix you can define, for instance, 3 points for the X dimension, 5 for the Y dimension and 2 for the Z dimension. In this case, the total number of pivot-points is $3 * 5 * 2 = 30$. With this set of pivot points, the cube is divided into smaller cubed zones each one bounded by eight nearby points. Each point is numbered. The numeral order of these points is established in the following way: number zero is the first point, number 1 the second and so on. Assuming you are using a 3-dimensional HVS cube having the number of points above mentioned (i.e. 3, 5 and 2 respectively for the X, Y and Z axis), the first point (point zero) is the upper-left-front vertex of the cube, by facing the XY plane of the cube. The second point is the middle point of the upper front edge of the cube and so on. You can refer to the figure below in order to understand how the numeral order of the pivot-points proceeds:

For the 2-dimensional HVS, it is the same, by only omitting the rear cube face, so each zone is bounded by 4 pivot-points instead of 8. For the 1-dimensional HVS, the whole thing is even simpler because it is a line with the pivot-points proceeding from left to right. Each point is coupled with a snapshot.

Snapshot order, as stored into the *iSnapTab*, can or cannot follow the order of the pivot-points numbers. In fact it is possible to alter this order by means the *iPositionsTab*, a table that remaps the position of each snapshot in relation to the pivot points. The *iPositionsTab* is made up of the positions of the snapshots (contained in the *iSnapTab*) in the two-dimensional grid. Each subsequent element is actually a pointer representing the position in the *iSnapTab*. For example, in a 2-dimensional HVS matrix such as the following (in this case having *inumPointsX* = 3 and *inumPointsY* = 5:

Table 12.

5	7	1
3	4	9
6	2	0
4	1	3
8	2	7

These numbers (to be stored in the *iSnapTab* table by using, for instance, the GEN02 function generator) represents the snapshot position within the grid (in this case a 3x5 matrix). So, the first element 5, has index zero and represents the 6th (element zero is the first) snapshot contained in the *iSnapTab*, the second element 7 represents the 8th element of *iSnapTab* and so on. Summing up, the vertices of each zone (a cubed zone is delimited by 8 vertices; a squared zone by 4 vertices and a linear zone by 2 points) are coupled with a determinate snapshot, whose number is remapped by the *iSnapTab*.

Output values of the HVS are influenced by the motion pointer, a point whose position, in the HVS cube (or square or segment) is determined by the *kx*, *ky* and *kz* arguments. The values of these arguments, which must be in the 0 to 1 range, are externally set by the user. The output values, whose amount is equal to the *inumParms* argument, are stored in the *iOutTab*, a table that must be already allocated by the user, and must be at least *inumParms* size. In what way the motion pointer influences the output? Well, when the motion pointer falls in a determinate cubed zone, delimited, for instance, by 8 vertices (or pivot points), we assume that each vertex has associated a different snapshot (i.e. a set of *inumParms* values), well, the output will be the weighted average value of the 8 vertices, calculated according on the distance of the motion pointer from each of the 8 vertices. In the case of a default behavior, when the *iConfigTab* argument is not set, the exact output is calculated by using linear interpolation which is applied to each different parameter of the HVS. Anyway,

it is possible to influence this behavior by setting the *iConfigTab* argument to a number of a table whose contents can affect one or more HVS parameters. The *iConfigTab* table elements are associated to each HVS parameter and their values affect the HVS output in the following way:

- If *iConfigTab* is equal to -1, corresponding output is skipped, i.e. the element is not calculated, leaving corresponding element value in the *iOutTab* unchanged;
- If *iConfigTab* is equal to zero, then the normal HVS output is calculated (by using weighted average of the nearest vertex of current zone where it falls the motion pointer);
- If *iConfigTab* element is equal to an integer number > zero, then the contents of a table having that number is used as a shape of a table-based interpolation.

See Also

hvs1, *hvs2*, *vphaseseg*

Credits

Author: Gabriel Maldonado

New in version 5.06

i

i — Returns an init-type equivalent of a k-rate argument.

Description

Returns an init-type equivalent of a k-rate argument.

Syntax

`i(x)` (control-rate args only)

Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.



Note

Using `i()` with a k-rate expression argument is not recommended, and can produce unexpected results.

See Also

a, k, abs, exp, frac, int, log, log10, sqrt

ibetarand

ibetarand — Deprecated.

Description

Deprecated as of version 3.49. Use the *betarand* opcode instead.

ibexprnd

ibexprnd — Deprecated.

Description

Deprecated as of version 3.49. Use the *bexprnd* opcode instead.

icauchy

icauchy — Deprecated.

Description

Deprecated as of version 3.49. Use the *cauchy* opcode instead.

ictrl14

ictrl14 — Deprecated.

Description

Deprecated as of version 3.52. Use the *ctrl14* opcode instead.

ictrl21

ictrl21 — Deprecated.

Description

Deprecated as of version 3.52. Use the *ctrl21* opcode instead.

ictrl7

ictrl7 — Deprecated.

Description

Deprecated as of version 3.52. Use the *ctrl7* opcode instead.

iexprand

iexprand — Deprecated.

Description

Deprecated as of version 3.49. Use the *exprand* opcode instead.

if

if — Branches conditionally at initialization or during performance time.

Description

if...igoto -- conditional branch at initialization time, depending on the truth value of the logical expression *ia R ib*. The branch is taken only if the result is true.

if...kgoto -- conditional branch during performance time, depending on the truth value of the logical expression *ka R kb*. The branch is taken only if the result is true.

if...goto -- combination of the above. Condition tested on every pass.

if...then -- allows the ability to specify conditional *if/else/endif* blocks. All *if...then* blocks must end with an *endif* statement. *elseif* and *else* statements are optional. Any number of *elseif* statements are allowed. Only one *else* statement may occur and it must be the last conditional statement before the *endif* statement. Nested *if...then* blocks are allowed.



Note

Note that if the condition uses a k-rate variable (for instance, “if kval > 0”), the *if...goto* or *if...then* statement will be ignored during the i-time pass. This allows for opcode initialization, even if the k-rate variable has already been assigned an appropriate value by an earlier init statement.

Syntax

```
if ia R ib igoto label
```

```
if ka R kb kgoto label
```

```
if xa R xb goto label
```

```
if xa R xb then
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

If *goto* or *then* is used instead of *kgoto* or *igoto*, the behavior is determined by the type being compared. If the comparison used k-type variables, *kgoto* is used and viceversa.



Note

If/then/goto statements cannot do audio-type comparisons. You can't put a-type variables in the comparison expressions for these opcodes. The reason for this is that audio variables are actually vectors, which can't be compared in the same way as scalars. If you need to compare individual audio samples, use *kr = 1* or *Comparators*

Examples

Here is an example of the *if...igoto* combination. It uses the file *igoto.csd* [examples/igoto.csd].

Example 240. Example of the *if...igoto* combination.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o igoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Get the value of the 4th p-field from the score.
iparam = p4

; If iparam is 1 then play the high note.
; If not then play the low note.
if (iparam == 1) igoto highnote
    igoto lownote

highnote:
    ifreq = 880
    goto playit

lownote:
    ifreq = 440
    goto playit

playit:
; Print the values of iparam and ifreq.
print iparam
print ifreq

    a1 oscil 10000, ifreq, 1
    out a1
endin

</CsInstruments>
<CsScore>

; Table #1: a simple sine wave.
f 1 0 32768 10 1

; p4: 1 = high note, anything else = low note
; Play Instrument #1 for one second, a low note.
i 1 0 1 0
; Play a Instrument #1 for one second, a high note.
i 1 1 1 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1: iparam = 0.000
instr 1: ifreq = 440.000
instr 1: iparam = 1.000
instr 1: ifreq = 880.000
```

Here is an example of the if...kgoto combination. It uses the file *kgoto.csd* [examples/kgoto.csd].

Example 241. Example of the if...kgoto combination.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o kgoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Change kval linearly from 0 to 2 over
; the period set by the third p-field.
kval line 0, p3, 2

; If kval is greater than or equal to 1 then play the high note.
; If not then play the low note.
if (kval >= 1) kgoto highnote
    kgoto lownote

highnote:
    kfreq = 880
    goto playit

lownote:
    kfreq = 440
    goto playit

playit:
; Print the values of kval and kfreq.
printks "kval = %f, kfreq = %f\\n", 1, kval, kfreq

    al oscil 10000, kfreq, 1
    out al
endin

</CsInstruments>
<CsScore>

; Table #1: a simple sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
kval = 0.000000, kfreq = 440.000000
kval = 0.999732, kfreq = 440.000000
kval = 1.999639, kfreq = 880.000000
```

Examples

Here is an example of the if...then combo. It uses the file *if.csd* [examples/ifthen.csd].

Example 242. Example of the if...then combo.

See Also

elseif, else, endif, goto, igoto, kgoto, tigoto, timeout

Credits

Examples written by Kevin Conder.

Added a note by Jim Aikin.

February 2004. Added a note by Matt Ingalls.

igauss

igauss — Deprecated.

Description

Deprecated as of version 3.49. Use the *gauss* opcode instead.

igoto

igoto — Transfer control during the i-time pass.

Description

During the i-time pass only, unconditionally transfer control to the statement labeled by *label*.

Syntax

```
igoto label
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

Examples

Here is an example of the igoto opcode. It uses the file *igoto.csd* [examples/igoto.csd].

Example 243. Example of the igoto opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o igoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Get the value of the 4th p-field from the score.
iparam = p4

; If iparam is 1 then play the high note.
; If not then play the low note.
if (iparam == 1) igoto highnote
    igoto lownote

highnote:
    ifreq = 880
    goto playit

lownote:
    ifreq = 440
    goto playit

playit:
; Print the values of iparam and ifreq.
print iparam
print ifreq

    a1 oscil 10000, ifreq, 1
    out a1
endin
```



```
</CsInstruments>
<CsScore>

; Table #1: a simple sine wave.
f 1 0 32768 10 1

; p4: 1 = high note, anything else = low note
; Play Instrument #1 for one second, a low note.
i 1 0 1 0
; Play a Instrument #1 for one second, a high note.
i 1 1 1 1
e

</CsScore>
</CsSoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  iparam = 0.000
instr 1:  ifreq = 440.000
instr 1:  iparam = 1.000
instr 1:  ifreq = 880.000
```

See Also

cggoto, cigoto, ckgoto, goto, if, kgoto, rigoto, tigoto, timeout

Credits

Example written by Kevin Conder.

Added a note by Jim Aikin.

ihold

ihold — Creates a held note.

Description

Causes a finite-duration note to become a “held” note

Syntax

ihold

Performance

ihold -- this i-time statement causes a finite-duration note to become a “held” note. It thus has the same effect as a negative p3 (see score *i Statement*), except that p3 here remains positive and the instrument reclassifies itself to being held indefinitely. The note can be turned off explicitly with *turnoff*, or its space taken over by another note of the same instrument number (i.e. it is tied into that note). Effective at i-time only; no-op during a *reinit* pass.

Examples

Here is an example of the *ihold* opcode. It uses the file *ihold.csd* [examples/ihold.csd].

Example 244. Example of the *ihold* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ihold.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; A simple oscillator with its note held indefinitely.
a1 oscil 10000, 440, 1
ihold

; If p4 equals 0, turn the note off.
if (p4 == 0) kgoto offnow
           kgoto playit

offnow:
; Turn the note off now.
turnoff

playit:
; Play the note.
out a1
endin
```

```
</CsInstruments>
<CsScore>

; Table #1: an ordinary sine wave.
f 1 0 32768 10 1

; p4 = turn the note off (if it is equal to 0).
; Start playing Instrument #1.
i 1 0 1 1
; Turn Instrument #1 off after 3 seconds.
i 1 3 1 0
e

</CsScore>
</CsoundSynthesizer>
```

See Also

i Statement, turnoff

Credits

Example written by Kevin Conder.

ilinrand

ilinrand — Deprecated.

Description

Deprecated as of version 3.49. Use the *linrand* opcode instead.

imagecreate

imagecreate — Create an empty image of a given size.

Description

Create an empty image of a given size. Individual pixel values can then be set with *imagegetpixel*.

Syntax

```
iimagenum imagecreate iwidth, iheight
```

Initialization

iimagenum -- number assigned to the created image.

iwidth -- desired image width.

iheight -- desired image height.

Examples

Here is an example of the imagecreate opcode. It uses the file *imageopcodes.csd* [examples/imageopcodes.csd].

Example 245. Example of the imageload opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
;-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
-o imageopcodes.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr=48000
ksmps=1
nchnls=2

; this test .csd copies image.png into a new file 'imageout.png'

giimage1 imageload "image.png"
giimagew, giimageh imagesize giimage1
giimage2 imagecreate giimagew, giimageh

    instr 1

kndx = 0
kx_ linseg 0, p3, 1

myloop:
ky_ = kndx/(giimageh)
kr_ kg_ kb_ imagegetpixel giimage1, kx_, ky_
imagesetpixel giimage2, kx_, ky_, kr_, kg_, kb_
loop_lt kndx, 0.5, (giimageh), myloop
    endin

    instr 2

imagesave giimage2, "imageout.png"
    endin

    instr 3
imagefree giimage1
imagefree giimage2
```

```
    endin

</CsInstruments>
<CsScore>

i1 1 1
i2 2 1
i3 3 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

imageload, imagesize, imagesave, imagegetpixel, imagesetpixel, imagefree

Credits

Author: Cesare Marilungo

New in version 5.08

imagefree

imagecreate — Frees memory allocated for a previously loaded or created image.

Description

Frees memory allocated for a previously loaded or created image.

Syntax

```
imagefree iimagenum
```

Initialization

iimagenum -- reference of the image to free.

Examples

Here is an example of the imagefree opcode. It uses the file *imageopcodes.csd* [examples/imageopcodes.csd].

Example 246. Example of the imagefree opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
;-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
-o imageopcodes.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr=48000
ksmps=1
nchnls=2

; this test .csd copies image.png into a new file 'imageout.png'

giimage1 imageload "image.png"
giimagew, giimageh imagesize giimage1
giimage2 imagecreate giimagew,giimageh

instr 1

kndx = 0
kx_ linseg 0, p3, 1

myloop:
ky_ = kndx/(giimageh)
kr_ kg_ kb_ imagegetpixel giimage1, kx_, ky_
imagesetpixel giimage2, kx_, ky_, kr_, kg_, kb_
loop_lt kndx, 0.5, (giimageh), myloop
endin

instr 2

imagesave giimage2, "imageout.png"
endin

instr 3
imagefree giimage1
imagefree giimage2
endin

</CsInstruments>
<CsScore>
```

```
i1 1 1  
i2 2 1  
i3 3 1  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

imageload, imagecreate, imagesize, imagesave, imagegetpixel, imagesetpixel

Credits

Author: Cesare Marilungo

New in version 5.08

imagegetpixel

`imagegetpixel` — Return the RGB pixel values of a previously opened or created image.

Description

Return the RGB pixel values of a previously opened or created image. An image can be loaded with *imageload*. An empty image can be created with *imagecreate*.

Syntax

```
ared agreen ablue imagegetpixel iimagenum, ax, ay
```

```
kred kgreen kblue imagegetpixel iimagenum, kx, ky
```

Initialization

iimagenum -- the reference of the image.. It should be a value returned by *imageload* or *imagecreate*.

Performance

ax (*kx*) -- horizontal pixel position (must be a float from 0 to 1).

ay (*ky*) -- vertical pixel position (must be a float from 0 to 1).

ared (*kred*) -- red value of the pixel (mapped to a float from 0 to 1).

agreen (*kgreen*) -- green value of the pixel (mapped to a float from 0 to 1).

ablue (*kblue*) -- blue value of the pixel (mapped to a float from 0 to 1).

Examples

Here is an example of the `imagegetpixel` opcode. It uses the files *imageopcodesdemo2.csd* [examples/imageopcodesdemo2.csd] *test1.png* [examples/test1.png] and *test2.png* [examples/test2.png].

Example 247. Example of the `imagegetpixel` opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o imageopcodesdemo2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr          =          48000
ksmps      =          100
nchnls     = 2

;By Cesare Marilungo 2008
zakinit 10,1

;Load the image - should be 512x512 pixels
giimage imageload "test1.png"
;giimage imageload "test2.png" ;--try this too
```

```
giimagew, giimageh imagesize giimage

giwave ftgen 1, 0, 1024, 10, 1
gifrqs ftgen 2,0,512,-5, 1,512,10
giamps ftgen 3, 0, 512, 10, 1

    instr 100

kindex = 0
icnt = giimageh
kx_ linseg 0, p3, 1
kenv linseg 0, .2, 500, p3 - .4, 500, .2, 0

; Read a column of pixels and store the red values
; inside the table 'giamps'
loop:
    ky_ = kindex/giimageh

    ;Get the pixel color values at kx_, ky_
    kred, kgreen, kblue imagegetpixel giimage, kx_, ky_

    ;Write the red values inside the table 'giamps'
    tablew kred, kindex, giamps
    kindex = kindex+1

if (kindex < icnt) kgoto loop

; Use an oscillator bank (additive synthesis) to generate sound
; setting amplitudes for each partial according to the image
asig adsynt kenv, 220, giwave, gifrqs, giamps, icnt, 2
outs asig, asig

    endin

    instr 101
; Free memory used by the image
imagefree giimage
    endin

</CsInstruments>
<CsScore>

t 0 60

i100 1 20
i101 21 1

e

</CsScore>
</CsoundSynthesizer>
```

See Also

imageload, imagecreate, imagesize, imagesave, imagesetpixel, imagefree

Credits

Author: Cesare Marilungo

New in version 5.08

imageload

imageload — Load an image.

Description

Load an image and return a reference to it. Individual pixel values can then be accessed with *image-getpixel*.

Syntax

```
iimagenum imageload filename
```

Initialization

iimagenum -- number assigned to the loaded image.

filename -- The filename of the image to load (should be a valid PNG image file).

Examples

Here is an example of the imageload opcode. It uses the file *imageopcodes.csd* [examples/imageopcodes.csd].

Example 248. Example of the imageload opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
;-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
-o imageopcodes.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr=48000
ksmps=1
nchnls=2

; this test .csd copies image.png into a new file 'imageout.png'

giimage1 imageload "image.png"
giimagew, giimageh imagesize giimage1
giimage2 imagecreate giimagew,giimageh

    instr 1

kndx = 0
kx_ linseg 0, p3, 1

myloop:
ky_ = kndx/(giimageh)
kr_ kg_ kb_ imagegetpixel giimage1, kx_, ky_
imagesetpixel giimage2, kx_, ky_, kr_, kg_, kb_
loop_lt kndx, 0.5, (giimageh), myloop
    endin

    instr 2

imagesave giimage2, "imageout.png"
    endin

    instr 3
imagefree giimage1
imagefree giimage2
    endin
```

```
</CsInstruments>
<CsScore>

i1 1 1
i2 2 1
i3 3 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

imagecreate, imagesize, imagesave, imagegetpixel, imagesetpixel, imagefree

Credits

Author: Cesare Marilungo

New in version 5.08

imagesave

imagesave — Save a previously created image.

Description

Save a previously created image. An empty image can be created with *imagecreate* and its pixel RGB values can be set with *imagesetpixel*. The image will be saved in PNG format.

Syntax

```
imagesave iimagenum, filename
```

Initialization

iimagenum -- the reference of the image to be save. It should be a value returned by *imagecreate*.

filename -- The filename to use to save the image.

Examples

Here is an example of the imagesave opcode. It uses the file *imageopcodes.csd* [examples/imageopcodes.csd].

Example 249. Example of the imagesave opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
;-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
-o imageopcodes.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr=48000
ksmps=1
nchnls=2

; this test .csd copies image.png into a new file 'imageout.png'

giimage1 imageload "image.png"
giimagew, giimageh imagesize giimage1
giimage2 imagecreate giimagew,giimageh

instr 1
kndx = 0
kx_ linseg 0, p3, 1

myloop:
ky_ = kndx/(giimageh)
kr_ kg_ kb_ imagegetpixel giimage1, kx_, ky_
imagesetpixel giimage2, kx_, ky_, kr_, kg_, kb_
loop_lt kndx, 0.5, (giimageh), myloop
endin

instr 2

imagesave giimage2, "imageout.png"
endin

instr 3
imagefree giimage1
imagefree giimage2
endin
```

```
</CsInstruments>
<CsScore>

i1 1 1
i2 2 1
i3 3 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

imageload, imagecreate, imagesize, imagegetpixel, imagesetpixel, imagefree

Credits

Author: Cesare Marilungo

New in version 5.08

imagesetpixel

imagesetpixel — Set the RGB value of a pixel inside a previously opened or created image.

Description

Set the RGB value of a pixel inside a previously opened or created image. An image can be loaded with *imageload*. An empty image can be created with *imagecreate* and saved with *imagesave*.

Syntax

```
imagegetpixel iimagenum, ax, ay, ared, agreen, ablue
```

```
imagegetpixel iimagenum, kx, ky, kred, kgreen, kblue
```

Initialization

iimagenum -- the reference of the image.. It should be a value returned by *imageload* or *imagecreate*.

Performance

ax (*kx*) -- horizontal pixel position (must be a float from 0 to 1).

ay (*ky*) -- vertical pixel position (must be a float from 0 to 1).

ared (*kred*) -- red value of the pixel (mapped to a float from 0 to 1).

agreen (*kgreen*) -- green value of the pixel (mapped to a float from 0 to 1).

ablue (*kblue*) -- blue value of the pixel (mapped to a float from 0 to 1).

Examples

Here is an example of the imagesetpixel opcode. It uses the file *imageopcodes.csd* [examples/imageopcodes.csd].

Example 250. Example of the imagesetpixel opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
;-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
-o imageopcodes.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr=48000
ksmps=1
nchnls=2

; this test .csd copies image.png into a new file 'imageout.png'

giimager1 imageload "image.png"
giimagerw, giimagerh imagesize giimager1
giimager2 imagecreate giimagerw,giimagerh

instr 1
```

```
kndx = 0
kx_ linseg 0, p3, 1

myloop:
ky_ = kndx/(giimageh)
kr_ kg_ kb_ imagegetpixel giimage1, kx_, ky_
imagesetpixel giimage2, kx_, ky_, kr_, kg_, kb_
loop_lt kndx, 0.5, (giimageh), myloop
    endin

instr 2

imagesave giimage2, "imageout.png"
    endin

instr 3
imagefree giimage1
imagefree giimage2
    endin

</CsInstruments>
<CsScore>

i1 1 1
i2 2 1
i3 3 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

imagedload, imagecreate, imagesize, imagesave, imagegetpixel, imagefree

Credits

Author: Cesare Marilungo

New in version 5.08

imagesize

`imagesize` — Return the width and height of a previously opened or created image.

Description

Return the width and height of a previously opened or created image. An image can be loaded with *imageload*. An empty image can be created with *imagecreate*.

Syntax

```
iwidth iheight imagesize iimagenum
```

Initialization

iimagenum -- the reference of the image.. It should be a value returned by *imageload* or *imagecreate*.

iwidth -- image width.

iheight -- image height.

Examples

Here is an example of the `imagesize` opcode. It uses the file *imageopcodes.csd* [examples/imageopcodes.csd].

Example 251. Example of the `imagesize` opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
;-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
-o imageopcodes.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr=48000
ksmps=1
nchnls=2

; this test .csd copies image.png into a new file 'imageout.png'

giimage1 imageload "image.png"
giimagew, giimageh imagesize giimage1
giimage2 imagecreate giimagew,giimageh

    instr 1

kndx = 0
kx_ linseg 0, p3, 1

myloop:
ky_ = kndx/(giimageh)
kr_ kg_ kb_ imagegetpixel giimage1, kx_, ky_
imagesetpixel giimage2, kx_, ky_, kr_, kg_, kb_
loop_lt kndx, 0.5, (giimageh), myloop
    endin

    instr 2

imagesave giimage2, "imageout.png"
    endin
```

```
instr 3
imagefree giimage1
imagefree giimage2
endin

</CsInstruments>
<CsScore>

i1 1 1
i2 2 1
i3 3 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

imageload, imagecreate, imagesave, imagegetpixel, imagesetpixel, imagefree

Credits

Author: Cesare Marilungo

New in version 5.08

imidic14

imidic14 — Deprecated.

Description

Deprecated as of version 3.52. Use the *midic14* opcode instead.

imidic21

imidic21 — Deprecated.

Description

Deprecated as of version 3.52. Use the *midic21* opcode instead.

imidic7

imidic7 — Deprecated.

Description

Deprecated as of version 3.52. Use the *midic7* opcode instead.

in

in — Reads mono audio data from an external device or stream.

Description

Reads mono audio data from an external device or stream.



Warning

This opcode is designed to be used only with orchestras that have `nchnls=1`. Doing so with orchestras with `nchnls > 1` will cause incorrect audio input.

Syntax

```
ar1 in
```

Performance

Reads mono audio data from an external device or stream. If the command-line `-i` flag is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer. Any number of these opcodes can read freely from this buffer.

See Also

diskin, inh, inh, ino, inq, ins, soundin

Credits

Authors: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

Already in version 3.30

in32

in32 — Reads a 32-channel audio signal from an external device or stream.

Description

Reads a 32-channel audio signal from an external device or stream.



Warning

This opcode is designed to be used only with orchestras that have `nchnls=32`. Doing so with orchestras with `nchnls > 32` will cause incorrect audio input.

Syntax

```
ar1, ar2, ar3, ar4, ar5, ar6, ar7, ar8, ar9, ar10, ar11, ar12, ar13, ar14, \  
    ar15, ar16, ar17, ar18, ar19, ar20, ar21, ar22, ar23, ar24, ar25, ar26, \  
    ar27, ar28, ar29, ar30, ar31, ar32 in32
```

Performance

in32 reads a 32-channel audio signal from an external device or stream. If the command-line *-i* flag is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer.

Credits

inch, inx, inz

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

inch

inch — Reads from a numbered channel in an external audio signal or stream.

Description

Reads from a numbered channel in an external audio signal or stream.

Syntax

```
ar1 inch ksig1
```

Performance

inch reads from a numbered channel determined by *ksig1* into *a1*. If the command-line *-i* flag is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer.

Credits

in32, inx, inz

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

inh

inh — Reads six-channel audio data from an external device or stream.

Description

Reads six-channel audio data from an external device or stream.



Warning

This opcode is designed to be used only with orchestras that have `nchnls=6`. Doing so with orchestras with `nchnls > 6` will cause incorrect audio input.

Syntax

`ar1, ar2, ar3, ar4, ar5, ar6 inh`

Performance

Reads six-channel audio data from an external device or stream. If the command-line `-i` flag is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer. Any number of these opcodes can read freely from this buffer.

See Also

diskin, in, ino, inq, ins, soundin

Credits

Author: John ffitch

init

init — Puts the value of the i-time expression into a k- or a-rate variable.

Syntax

```
ares init iarg
```

```
ires init iarg
```

```
kres init iarg
```

Description

Put the value of the i-time expression into a k- or a-rate variable.

Initialization

Puts the value of the i-time expression *iarg* into a k- or a-rate variable, i.e., initialize the result. Note that **init** provides the only case of an init-time statement being permitted to write into a perf-time (k- or a-rate) result cell; the statement has no effect at perf-time.

See Also

`=`, *divz*, *tival*

initc14

initc14 — Initializes the controllers used to create a 14-bit MIDI value.

Description

Initializes the controllers used to create a 14-bit MIDI value.

Syntax

```
initc14 ichan, ictlno1, ictlno2, ivalue
```

Initialization

ichan -- MIDI channel (1-16)

ictlno1 -- most significant byte controller number (0-127)

ictlno2 -- least significant byte controller number (0-127)

ivalue -- floating point value (must be within 0 to 1)

Performance

initc14 can be used together with both *midic14* and *ctrl14* opcodes for initializing the first controller's value. *ivalue* argument must be set with a number within 0 to 1. An error occurs if it is not. Use the following formula to set *ivalue* according with *midic14* and *ctrl14* min and max range:

$$\text{ivalue} = (\text{initial_value} - \text{min}) / (\text{max} - \text{min})$$

See Also

ctrl7, *ctrl14*, *ctrl21*, *ctrlinit*, *initc7*, *initc21*, *midic7*, *midic14*, *midic21*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

initc21

initc21 — Initializes the controllers used to create a 21-bit MIDI value.

Description

Initializes the controllers used to create a 21-bit MIDI value.

Syntax

```
initc21 ichan, ictlno1, ictlno2, ictlno3, ivalue
```

Initialization

ichan -- MIDI channel (1-16)

ictlno1 -- most significant byte controller number (0-127)

ictlno2 -- medium significant byte controller number (0-127)

ictlno3 -- least significant byte controller number (0-127)

ivalue -- floating point value (must be within 0 to 1)

Performance

initc21 can be used together with both *midic21* and *ctrl21* opcodes for initializing the first controller's value. *ivalue* argument must be set with a number within 0 to 1. An error occurs if it is not. Use the following formula to set *ivalue* according with *midic21* and *ctrl21* min and max range:

$$ivalue = (initial_value - min) / (max - min)$$

See Also

ctrl7, *ctrl14*, *ctrl21*, *ctrlinit*, *initc7*, *initc14*, *midic7*, *midic14*, *midic21*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

initc7

initc7 — Initializes the controller used to create a 7-bit MIDI value.

Description

Initializes MIDI controller *ictlno* with *ivalue*

Syntax

```
initc7 ichan, ictlno, ivalue
```

Initialization

ichan -- MIDI channel (1-16)

ictlno -- controller number (0-127)

ivalue -- floating point value (must be within 0 to 1)

Performance

initc7 can be used together with both *midic7* and *ctrl7* opcodes for initializing the first controller's value. *ivalue* argument must be set with a number within 0 to 1. An error occurs if it is not. Use the following formula to set *ivalue* according with *midic7* and *ctrl7* min and max range:

$$\text{ivalue} = (\text{initial_value} - \text{min}) / (\text{max} - \text{min})$$

See Also

ctrl7, ctrl14, ctrl21, ctrlinit, initc14, initc21, midic7, midic14, midic21

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

ino

ino — Reads eight-channel audio data from an external device or stream.

Description

Reads eight-channel audio data from an external device or stream.



Warning

This opcode is designed to be used only with orchestras that have `nchnls=8`. Doing so with orchestras with `nchnls > 8` will cause incorrect audio input.

Syntax

```
ar1, ar2, ar3, ar4, ar5, ar6, ar7, ar8 ino
```

Performance

Reads eight-channel audio data from an external device or stream. If the command-line `-i` flag is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer. Any number of these opcodes can read freely from this buffer.

See Also

diskin, in, inh, inh, inq, ins, soundin

Credits

Author: John ffitch

inq

inq — Reads quad audio data from an external device or stream.

Description

Reads quad audio data from an external device or stream.



Warning

This opcode is designed to be used only with orchestras that have `nchnls=4`. Doing so with orchestras with `nchnls > 4` will cause incorrect audio input.

Syntax

```
ar1, ar2, ar3, a4 inq
```

Performance

Reads quad audio data from an external device or stream. If the command-line `-i` flag is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer. Any number of these opcodes can read freely from this buffer.

See Also

diskin, in, inh, inh, ino, ins, soundin

Credits

Authors: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

inrg

inrg — Allow input from a range of adjacent audio channels from the audio input device

Description

inrg reads audio from a range of adjacent audio channels from the audio input device.

Syntax

```
inrg kstart, ain1 [,ain2, ain3, ..., ainN]
```

Performance

kstart - the number of the first channel of the input device to be accessed (channel numbers starts with 1, which is the first channel)

ain1, ain2, ... ainN - the output arguments filled with the incoming audio coming from corresponding channels.

inrg allows input from a range of adjacent channels from the input device. *kstart* indicates the first channel to be accessed (channel 1 is the first channel). The user must be sure that the number obtained by summing *kstart* plus the number of accessed channels -1 is $\leq nchnls$.



Note

Note that this opcode is exceptional in that it produces its “output” on the parameters to the right.

Credits

Author: Gabriel Maldonado

New in version 5.06

ins

ins — Reads stereo audio data from an external device or stream.

Description

Reads stereo audio data from an external device or stream.



Warning

This opcode is designed to be used only with orchestras that have `nchnls=2`. Doing so with orchestras with `nchnls > 2` will cause incorrect audio input.

Syntax

```
ar1, ar2 ins
```

Performance

Reads stereo audio data from an external device or stream. If the command-line `-i` flag is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer. Any number of these opcodes can read freely from this buffer.

See Also

diskin, in, inh, inh, ino, inq, soundin

Credits

Authors: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

insremot

insremot — An opcode which can be used to implement a remote orchestra. This opcode will send note events from a source machine to one destination.

Description

With the **insremot** and **insglobal** opcodes you are able to perform instruments on remote machines and control them from a master machine. The remote opcodes are implemented using the master/client model. All the machines involved contain the same orchestra but only the master machine contains the information of the score. During the performance the master machine sends the note events to the clients. The **insremot** opcode will send events from a source machine to one destination if you want to send events to many destinations (broadcast) use the **insglobal** opcode instead. These two opcodes can be used in combination.

Syntax

```
insremot idestination, isource, instrnum [,instrnum...]
```

Initialization

idestination -- a string that is the intended host computer (e.g. 192.168.0.100). This is the destination host which receives the events from the given instrument.

isource -- a string that is the intended host computer (e.g. 192.168.0.100). This is the source host which generates the events of the given instrument and sends it to the address given by *idestination*.

instrnum -- a list of instrument numbers which will be played on the destination machine

Examples

Here is an example of the **insremot** opcode. It uses the files *insremot.csd* [examples/insremot.csd] and *insremotM.csd* [examples/insremotM.csd].

Example 252. Example of the insremot opcode.

The simple example below shows the *bilbar* example played on a remote machine. The master machine is named "192.168.1.100" and the client "192.168.1.101". Start the client on the machine (it will wait to receive the events from the master machine) and then start the master. Here is the command on linux to start a client (`csound -+rtaudio=alsa -odac -dm0 insremot.csd`), and the command on the master machine will look like this (`csound -+rtaudio=alsa -odac -dm0 insremotM.csd`).

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o insremot.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
nchnls = 1

insremot "192.168.1.100", "192.168.1.101", 1

instr 1
```

```
    aq barmodel 1, 1, p4, 0.001, 0.23, 5, p5, p6, p7
    out      aq
endin

</CsInstruments>
<CsScore>
f0 360

e
</CsScore>
</CsoundSynthesizer>

<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o insremotM.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
nchnls = 1

insremot "192.168.1.100", "192.168.1.101", 1

instr 1
    aq barmodel 1, 1, p4, 0.001, 0.23, 5, p5, p6, p7
    out      aq
endin

</CsInstruments>
<CsScore>
i1 0 0.5 3 0.2 500 0.05
i1 0.5 0.5 -3 0.3 1000 0.05
i1 1.0 0.5 -3 0.4 1000 0.1
i1 1.5 4.0 -3 0.5 800 0.05
e
</CsScore>
</CsoundSynthesizer>
```

See also

insglobal, midglobal, midremot, remoteport

Credits

Author: Simon Schampijer
2006

New in version 5.03

insglobal

insglobal — An opcode which can be used to implement a remote orchestra. This opcode will send note events from a source machine to many destinations.

Description

With the *insremot* and *insglobal* opcodes you are able to perform instruments on remote machines and control them from a master machine. The remote opcodes are implemented using the master/client model. All the machines involved contain the same orchestra but only the master machine contains the information of the score. During the performance the master machine sends the note events to the clients. The *insglobal* opcode sends the events to all the machines involved in the remote concert. These machines are determined by the *insremot* definitions made above the *insglobal* command. To send events to only one machine use *insremot*.

Syntax

```
insglobalsource, instrnum [,instrnum...]
```

Initialization

source -- a string that is the intended host computer (e.g. 192.168.0.100). This is the source host which generates the events of the given instrument(s) and sends it to all the machines involved in the remote concert.

instrnum -- a list of instrument numbers which will be played on the destination machines

Examples

See the entry for *insremot* for an example of usage.

See also

insremot, *midglobal*, *midremot*, *remoteport*

Credits

Author: Simon Schampijer
2006

New in version 5.03

instimek

instimek — Deprecated.

Description

Deprecated as of version 3.62. Use the *timeinstk* opcode instead.

Credits

David M. Boothe originally pointed out this deprecated name.

instimes

instimes — Deprecated.

Description

Deprecated as of version 3.62. Use the *timeinsts* opcode instead.

Credits

David M. Boothe originally pointed out this deprecated name.

instr

instr — Starts an instrument block.

Description

Starts an instrument block.

Syntax

```
instr i, j, ...
```

Initialization

Starts an instrument block defining instruments *i, j, ...*

i, j, ... must be numbers, not expressions. Any positive integer is legal, and in any order, but excessively high numbers are best avoided.



Note

There may be any number of instrument blocks in an orchestra.

Instruments can be defined in any order (but they will always be both initialized and performed in ascending instrument number order, with the exception of notes triggered by real time events that are initialized in the order of being received but still performed in ascending instrument number order). Instrument blocks cannot be nested (i.e. one block cannot contain another).

Performance

Calling an Instrument within an Instrument

You can call an instrument within an instrument as if it were an opcode either with the *subinstr* opcode or by specifying an instrument with a text name:

```
instr MyOscil  
...  
endin
```

If an instrument is defined with a name, you simply call it directly like an opcode:

```
asig MyOscil iamp, ipitch, iftable
```

By default, all output parameters correspond to the called instrument's output with the *signal output* opcodes. All input parameters are mapped to the called instrument's p-fields starting with the fourth one, p4. The values of the called instrument's second and third p-fields, p2 and p3, are automatically set to those of the calling instrument's.

A named instrument must be defined before any instrument that calls it.



Hint

If you use the *outc* opcode, you can create an instrument that will compile and function in any orchestra of any number of channels greater than or equal to the output channels of the instrument.

A nice feature to use with named instruments is the *#include* feature. You can then define your named instruments in separate files, using *#include* when you need to use one.

Examples

Here is an example of the *instr* opcode. It uses the file *instr.csd* [examples/instr.csd].

Example 253. Example of the *instr* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o instr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  iamp = 10000
  icps = 440
  iphs = 0

  al oscils iamp, icps, iphs
  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

endin, *in*, *out*, *opcode*, *endop*, *setksmps*, *xin*, *xout*, *subinstr*, *subinstrinit*

Credits

Example written by Kevin Conder.

int

int — Extracts an integer from a decimal number.

Description

Returns the integer part of x .

Syntax

`int(x)` (init-rate or control-rate; also works at audio rate in Csound5)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

Examples

Here is an example of the int opcode. It uses the file *int.csd* [examples/int.csd].

Example 254. Example of the int opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o int.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = 16 / 5
  i2 = int(i1)

  print i2
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i2 = 3.000
```

See Also

abs, exp, frac, log, log10, i, sqrt

Credits

Example written by Kevin Conder.

integ

integ — Modify a signal by integration.

Description

Modify a signal by integration.

Syntax

```
ares integ asig [, iskip]
```

```
kres integ ksig [, iskip]
```

Initialization

iskip (optional) -- initial disposition of internal save space (see *reson*). The default value is 0.

Performance

integ and *diff* perform integration and differentiation on an input control signal or audio signal. Each is the converse of the other, and applying both will reconstruct the original signal. Since these units are special cases of low-pass and high-pass filters, they produce a scaled (and phase shifted) output that is frequency-dependent. Thus *diff* of a sine produces a cosine, with amplitude $2 * \sin(\pi * Hz / sr)$ that of the original (for each component partial); *integ* will inversely affect the magnitudes of its component inputs. With this understanding, these units can provide useful signal modification.

Examples

Here is an example of the integ opcode. It uses the file *integ.csd* [examples/integ.csd].

Example 255. Example of the integ opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o integ.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 -- a differentiated signal.
instr 1
; Generate a band-limited pulse train.
asrc buzz 20000, 440, 20, 1

; Differentiate the signal.
adiff diff asrc

out adiff
```

```
endin

; Instrument #2 -- a re-integrated signal.
instr 2
; Generate a band-limited pulse train.
asrc buzz 20000, 440, 20, 1

; Differentiate the signal.
adiff diff asrc

; Re-integrate the previously differentiated signal.
al integ adiff

out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 1 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

diff, downsamp, interp, samphold, upsamp

Credits

Example written by Kevin Conder.

interp

interp — Converts a control signal to an audio signal using linear interpolation.

Description

Converts a control signal to an audio signal using linear interpolation.

Syntax

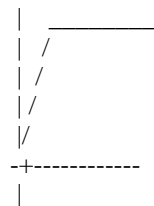
```
ares interp ksig [, iskip] [, imode]
```

Initialization

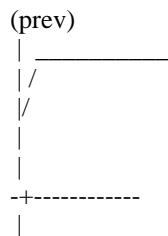
iskip (optional, default=0) -- initial disposition of internal save space (see *reson*). The default value is 0.

imode (optional, default=0) -- sets the initial output value to the first k-rate input instead of zero. The following graphs show the output of interp with a constant input value, in the original, when skipping init, and in the new mode:

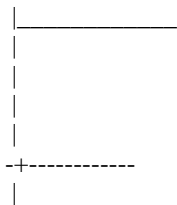
Example 256. iskip=0, imode=0



Example 257. iskip=1, imode=0



Example 258. iskip=0, imode=1



Performance

ksig -- input k-rate signal.

interp converts a control signal to an audio signal. It uses linear interpolation between successive kvals.

Examples

Here is an example of the *interp* opcode. It uses the file *interp.csd* [examples/interp.csd].

Example 259. Example of the *interp* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsSoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o interp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 8000
kr = 8
ksmps = 1000
nchnls = 1

; Instrument #1 - a simple instrument.
instr 1
; Create an amplitude envelope.
kamp linseg 0, p3/2, 20000, p3/2, 0

; The amplitude envelope will sound rough because it
; jumps every ksmps period, 1000.
a1 oscil kamp, 440, 1
out a1
endin

; Instrument #2 - a smoother sounding instrument.
instr 2
; Create an amplitude envelope.
kamp linseg 0, p3/2, 25000, p3/2, 0
aamp interp kamp

; The amplitude envelope will sound smoother due to
; linear interpolation at the higher a-rate, 8000.
a1 oscil aamp, 440, 1
out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 256 10 1
```

```
; Play Instrument #1 for two seconds.  
i 1 0 2  
; Play Instrument #2 for two seconds.  
i 2 2 2  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

diff, downsamp, integ, samphold, upsamp

Credits

Example written by Kevin Conder.

Updated November 2002, thanks to a note from both Rasmus Ekman and Istvan Varga.

invalue

invalue — Reads a k-rate signal from a user-defined channel.

Description

Reads a k-rate signal or string from a user-defined channel.

Syntax

```
kvalue invalue "channel name"
```

```
Sname invalue "channel name"
```

Performance

kvalue -- The k-rate value that is read from the channel.

Sname -- The string variable that is read from the channel.

"channel name" -- An integer, string (in double-quotes), or string variable identifying the channel.

See Also

outvalue

Credits

Author: Matt Ingalls

inx

inx — Reads a 16-channel audio signal from an external device or stream.

Description

Reads a 16-channel audio signal from an external device or stream.



Warning

This opcode is designed to be used only with orchestras that have `nchnls=16`. Doing so with orchestras with `nchnls > 16` will cause incorrect audio input.

Syntax

```
ar1, ar2, ar3, ar4, ar5, ar6, ar7, ar8, ar9, ar10, ar11, ar12, \  
ar13, ar14, ar15, ar16 inx
```

Performance

inx reads a 16-channel audio signal from an external device or stream. If the command-line *-i* flag is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer.

Credits

in32, inch, inz

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

inz

inz — Reads multi-channel audio samples into a ZAK array from an external device or stream.

Description

Reads multi-channel audio samples into a ZAK array from an external device or stream.

Syntax

```
inz ksigl
```

Performance

inz reads audio samples in *nchnls* into a ZAK array starting at *ksigl*. If the command-line *-i* flag is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer.

Credits

in32, *inch*, *inx*

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

ioff

ioff — Deprecated.

Description

Deprecated as of version 3.52. Use the *noteoff* opcode instead.

ion

ion — Deprecated.

Description

Deprecated as of version 3.52. Use the *noteon* opcode instead.

iondur

iondur — Deprecated.

Description

Deprecated as of version 3.52. Use the *noteondur* opcode instead.

iondur2

iondur2 — Deprecated.

Description

Deprecated as of version 3.52. Use the *noteondur2* opcode instead.

ioutat

ioutat — Deprecated.

Description

Deprecated as of version 3.52. Use the *outiat* opcode instead.

ioutc

ioutc — Deprecated.

Description

Deprecated as of version 3.52. Use the *outic* opcode instead.

ioutc14

ioutc14 — Deprecated.

Description

Deprecated as of version 3.52. Use the *outic14* opcode instead.

ioutpat

ioutpat — Deprecated.

Description

Deprecated as of version 3.52. Use the *outipat* opcode instead.

ioutpb

ioutpb — Deprecated.

Description

Deprecated as of version 3.52. Use the *outipb* opcode instead.

ioutpc

ioutpc — Deprecated.

Description

Deprecated as of version 3.52. Use the *outipc* opcode instead.

ipcauchy

ipcauchy — Deprecated.

Description

Deprecated as of version 3.49. Use the *pcauchy* opcode instead.

ipoisson

ipoisson — Deprecated.

Description

Deprecated as of version 3.49. Use the *poisson* opcode instead.

ipow

ipow — Deprecated.

Description

Deprecated as of version 3.48. Use the *pow* opcode instead.

is16b14

is16b14 — Deprecated.

Description

Deprecated as of version 3.52. Use the *s16b14* opcode instead.

is32b14

is32b14 — Deprecated.

Description

Deprecated as of version 3.52. Use the *s32b14* opcode instead.

islider16

islider16 — Deprecated.

Description

Deprecated as of version 3.52. Use the *slider16* opcode instead.

islider32

islider32 — Deprecated.

Description

Deprecated as of version 3.52. Use the *slider32* opcode instead.

islider64

islider64 — Deprecated.

Description

Deprecated as of version 3.52. Use the *slider64* opcode instead.

islider8

islider8 — Deprecated.

Description

Deprecated as of version 3.52. Use the *slider8* opcode instead.

itablecopy

itablecopy — Deprecated.

Description

Deprecated as of version 3.52. Use the *tablecopy* opcode instead.

itablegpw

itablegpw — Deprecated.

Description

Deprecated as of version 3.52. Use the *tableigpw* opcode instead.

itablemix

itablemix — Deprecated.

Description

Deprecated as of version 3.52. Use the *tableimix* opcode instead.

itablew

itablew — Deprecated.

Description

Deprecated as of version 3.52. Use the *tableiw* opcode instead.

itrirand

itrirand — Deprecated.

Description

Deprecated as of version 3.49. Use the *trirand* opcode instead.

iunirand

iunirand — Deprecated.

Description

Deprecated as of version 3.49. Use the *unirand* opcode instead.

iweibull

iweibull — Deprecated.

Description

Deprecated as of version 3.49. Use the *weibull* opcode instead.

jacktransport

jacktransport — Start/stop jack_transport and can optionally relocate the playback head.

Description

Start/stop jack_transport and can optionally relocate the playback head.

Syntax

```
jacktransport icommand [, ilocation]
```

Initialization

icommand -- 1 to start playing, 0 to stop.

ilocation -- optional location in seconds to specify where the playback head should be moved. If omitted, the transport is started from current location.



Note

Since *jacktransport* depends on jack audio connection kit, it will work only on Linux or Mac OS X systems which have the jack server running.

Examples

Here is a simple example of the jacktransport opcode. It uses the file *jacktransport.csd* [examples/jacktransport.csd].

Example 260. Simple example of the jacktransport opcode.

```
<CsoundSynthesizer>
<CsOptions>

+rtaudio=JACK -b 64 --sched -o dac:system:playback_
</CsOptions>
<CsInstruments>

sr      =                44100
ksmps   =                =    16
nchnls  = 2

    instr 1
jacktransport p4, p5
    endin

    instr 2
jacktransport p4
    endin

</CsInstruments>
<CsScore>

i2 0 5 1; play
i2 5 1 0; stop
i1 6 5 1 2 ; move at 2 seconds and start playing back
i1 11 1 0 0 ; stop and rewind

e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Cesare Marilungo

New in version 5.08

jitter

`jitter` — Generates a segmented line whose segments are randomly generated.

Description

Generates a segmented line whose segments are randomly generated.

Syntax

```
kout jitter kamp, kcpsMin, kcpsMax
```

Performance

kamp -- Amplitude of jitter deviation

kcpsMin -- Minimum speed of random frequency variations (expressed in cps)

kcpsMax -- Maximum speed of random frequency variations (expressed in cps)

jitter generates a segmented line whose segments are randomly generated inside the +kamp and -kamp interval. Duration of each segment is a random value generated according to kcpsmin and kcpsmax values.

jitter can be used to make more natural and “analog-sounding” some static, dull sound. For best results, it is suggested to keep its amplitude moderate.

Examples

Here is an example of the jitter opcode. It uses the file *jitter.csd* [examples/jitter.csd].

Example 261. Example of the jitter opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o jitter.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Instrument #1 -- plain instrument.
instr 1
  aplain vco 20000, 220, 2, 0.83

  outs aplain, aplain
endin

; Instrument #2 -- instrument with jitter.
instr 2
; Create a signal modulated the jitter opcode.
  kamp init 2
```

```
kcpsmin init 4
kcpsmax init 6
kj jitter kamp, kcpsmin, kcpsmax

aplain vco 20000, 220, 2, 0.83
ajitter vco 20000, 220+kj, 2, 0.83

outs aplain, ajitter
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 3 seconds.
i 1 0 3
; Play Instrument #2 for 3 seconds.
i 2 3 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

jitter2, vibr, vibrato

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

New in Version 4.15

jitter2

jitter2 — Generates a segmented line with user-controllable random segments.

Description

Generates a segmented line with user-controllable random segments.

Syntax

```
kout jitter2 ktotamp, kamp1, kcps1, kamp2, kcps2, kamp3, kcps3
```

Performance

ktotamp -- Resulting amplitude of jitter2

kamp1 -- Amplitude of the first jitter component

kcps1 -- Speed of random variation of the first jitter component (expressed in cps)

kamp2 -- Amplitude of the second jitter component

kcps2 -- Speed of random variation of the second jitter component (expressed in cps)

kamp3 -- Amplitude of the third jitter component

kcps3 -- Speed of random variation of the third jitter component (expressed in cps)

jitter2 also generates a segmented line such as *jitter*, but in this case the result is similar to the sum of three *randi* opcodes, each one with a different amplitude and frequency value (see *randi* for more details), that can be varied at k-rate. Different effects can be obtained by varying the input arguments.

jitter2 can be used to make more natural and “analog-sounding” some static, dull sound. For best results, it is suggested to keep its amplitude moderate.

Examples

Here is an example of the jitter2 opcode. It uses the file *jitter2.csd* [examples/jitter2.csd].

Example 262. Example of the jitter2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc       -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o jitter2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
```

```
nchnls = 2

; Instrument #1 -- plain instrument.
instr 1
  aplain vco 20000, 220, 2, 0.83
  outs aplain, aplain
endin

; Instrument #2 -- instrument with jitter.
instr 2
  ; Create a signal modulated with the jitter2 opcode.
  ktotamp init 2
  kamp1 init 0.66
  kcps1 init 3
  kamp2 init 0.66
  kcps2 init 3
  kamp3 init 0.66
  kcps3 init 3
  kj jitter2 ktotamp, kamp1, kcps1, kamp2, kcps2, \
    kamp3, kcps3

  aplain vco 20000, 220, 2, 0.83
  ajitter vco 20000, 220+kj, 2, 0.83

  outs aplain, ajitter
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 3 seconds.
i 1 0 3
; Play Instrument #2 for 3 seconds.
i 2 3 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

jitter, vibr, vibrato

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

New in Version 4.15

jspline

jspline — A jitter-spline generator.

Description

A jitter-spline generator.

Syntax

```
ares jspline xamp, kcpsMin, kcpsMax
```

```
kres jspline kamp, kcpsMin, kcpsMax
```

Performance

kres, ares -- Output signal

xamp -- Amplitude factor

kcpsMin, kcpsMax -- Range of point-generation rate. Min and max limits are expressed in cps.

jspline (jitter-spline generator) generates a smooth curve based on random points generated at [*cpsMin*, *cpsMax*] rate. This opcode is similar to *randomi* or *randi* or *jitter*, but segments are not straight lines, but cubic spline curves. Output value range is approximately $> -xamp$ and $< xamp$. Actually, real range could be a bit greater, because of interpolating curves between each pair of random-points.

At present time generated curves are quite smooth when *cpsMin* is not too different from *cpsMax*. When *cpsMin-cpsMax* interval is big, some little discontinuity could occur, but it should not be a problem, in most cases. Maybe the algorithm will be improved in next versions.

These opcodes are often better than *jitter* when user wants to “naturalize” or “analogize” digital sounds. They could be used also in algorithmic composition, to generate smooth random melodic lines when used together with *samphold* opcode.

Note that the result is quite different from the one obtained by filtering white noise, and they allow the user to obtain a much more precise control.

Credits

Author: Gabriel Maldonado

New in Version 4.15

k

k — Converts a i-rate parameter to an k-rate value.

Description

Converts an i-rate value to control rate, for example to be used with *rnd()* and *birnd()* to generate random numbers at k-rate.

Syntax

```
k(x) (i-rate args only)
```

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

See Also

i a

Credits

Author: Istvan Varga

New in version Csound 5.00

kbetarand

kbetarand — Deprecated.

Description

Deprecated as of version 3.49. Use the *betarand* opcode instead.

kbexprnd

kbexprnd — Deprecated.

Description

Deprecated as of version 3.49. Use the *bexprnd* opcode instead.

kcauchy

kcauchy — Deprecated.

Description

Deprecated as of version 3.49. Use the *cauchy* opcode instead.

kdump

kdump — Deprecated.

Description

Deprecated as of version 3.49. Use the *dumpk* opcode instead.

kdump2

kdump2 — Deprecated.

Description

Deprecated as of version 3.49. Use the *dumpk2* opcode instead.

kdump3

kdump3 — Deprecated.

Description

Deprecated as of version 3.49. Use the *dumpk3* opcode instead.

kdump4

kdump4 — Deprecated.

Description

Deprecated as of version 3.49. Use the *dumpk4* opcode instead.

kexprand

kexprand — Deprecated.

Description

Deprecated as of version 3.49. Use the *exprand* opcode instead.

kfilter2

kfilter2 — Deprecated.

Description

Deprecated as of version 3.49. Use the *filter2* opcode instead.

Credits

Author: Michael A. Casey
M.I.T.
Cambridge, Mass.
1997

New in version 3.47

kgauss

kgauss — Deprecated.

Description

Deprecated as of version 3.49. Use the *gauss* opcode instead.

kgoto

kgoto — Transfer control during the p-time passes.

Description

During the p-time passes only, unconditionally transfer control to the statement labeled by *label*.

Syntax

```
kgoto label
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

Examples

Here is an example of the kgoto opcode. It uses the file *kgoto.csd* [examples/kgoto.csd].

Example 263. Example of the kgoto opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o kgoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Change kval linearly from 0 to 2 over
; the period set by the third p-field.
kval line 0, p3, 2

; If kval is greater than or equal to 1 then play the high note.
; If not then play the low note.
if (kval >= 1) kgoto highnote
kgoto lownote

highnote:
kfreq = 880
goto playit

lownote:
kfreq = 440
goto playit

playit:
; Print the values of kval and kfreq.
printks "kval = %f, kfreq = %f\\n", 1, kval, kfreq

a1 oscil 10000, kfreq, 1
out a1
endin
```

```
</CsInstruments>
<CsScore>

; Table #1: a simple sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
kval = 0.000000, kfreq = 440.000000
kval = 0.999732, kfreq = 440.000000
kval = 1.999639, kfreq = 880.000000
```

See Also

cggoto, cigoto, ckgoto, goto, if, igoto, tigoto, timeout

Credits

Example written by Kevin Conder.

Added a note by Jim Aikin.

klinrand

klinrand — Deprecated.

Description

Deprecated as of version 3.49. Use the *linrand* opcode instead.

kon

kon — Deprecated.

Description

Deprecated as of version 3.49. Use the *midion* opcode instead.

koutat

koutat — Deprecated.

Description

Deprecated as of version 3.52. Use the *outkat* opcode instead.

koutc

koutc — Deprecated.

Description

Deprecated as of version 3.52. Use the *outkc* opcode instead.

koutc14

koutc14 — Deprecated.

Description

Deprecated as of version 3.52. Use the *outkc14* opcode instead.

koutpat

koutpat — Deprecated.

Description

Deprecated as of version 3.52. Use the *outkpat* opcode instead.

koutpb

koutpb — Deprecated.

Description

Deprecated as of version 3.52. Use the *outkpb* opcode instead.

koutpc

koutpc — Deprecated.

Description

Deprecated as of version 3.52. Use the *outkpc* opcode instead.

kpcauchy

kpcauchy — Deprecated.

Description

Deprecated as of version 3.49. Use the *pcauchy* opcode instead.

kpoisson

kpoisson — Deprecated.

Description

Deprecated as of version 3.49. Use the *poisson* opcode instead.

kpow

kpow — Deprecated.

Description

Deprecated as of version 3.48. Use the *pow* opcode instead.

kr

kr — Sets the control rate.

Description

These statements are global value *assignments*, made at the beginning of an orchestra, before any instrument block is defined. Their function is to set certain *reserved symbol variables* that are required for performance. Once set, these reserved symbols can be used in expressions anywhere in the orchestra.

Syntax

```
kr = iarg
```

Initialization

kr = (optional) -- set control rate to *iarg* samples per second. The default value is 1000.

In addition, any *global variable* [48] can be initialized by an *init-time assignment* anywhere before the first *instr statement*. All of the above assignments are run as instrument 0 (i-pass only) at the start of real performance.

Beginning with Csound version 3.46, *kr* can be omitted. Csound will use the default values, or calculate *kr* from defined *ksmps* and *sr*. It is usually better to just specify *ksmps* and *sr* and let csound calculate *kr*.

Examples

```
sr = 10000  
kr = 500  
ksmps = 20  
gil = sr/2.  
ga init 0  
itranspose = octpch(.01)
```

See Also

ksmps, *nchnls*, *sr*

kread

kread — Deprecated.

Description

Deprecated as of version 3.52. Use the *readk* opcode instead.

kread2

kread2 — Deprecated.

Description

Deprecated as of version 3.52. Use the *readk2* opcode instead.

kread3

kread3 — Deprecated.

Description

Deprecated as of version 3.52. Use the *readk3* opcode instead.

kread4

kread4 — Deprecated.

Description

Deprecated as of version 3.52. Use the *readk4* opcode instead.

ksmps

ksmps — Sets the number of samples in a control period.

Description

These statements are global value *assignments*, made at the beginning of an orchestra, before any instrument block is defined. Their function is to set certain *reserved symbol variables* that are required for performance. Once set, these reserved symbols can be used in expressions anywhere in the orchestra.

Syntax

```
ksmps = iarg
```

Initialization

ksmps = (optional) -- set the number of samples in a control period. This value must equal *sr*/*kr*. The default value is 10.

In addition, any *global variable* [48] can be initialized by an *init-time assignment* anywhere before the first *instr statement*. All of the above assignments are run as instrument 0 (i-pass only) at the start of real performance.

Beginning with Csound version 3.46, either *ksmps* may be omitted. Csound will attempt to calculate the omitted value from the specified *sr* and *kr* values, but it should evaluate to an integer.



Warning

ksmps must be an integer value.

Examples

```
sr = 10000
kr = 500
ksmps = 20
gil = sr/2.
ga init 0
itranspose = octpch(.01)
```

See Also

kr, *nchnls*, *sr*

Credits

Thanks to a note from Gabriel Maldonado, added a warning about integer values.

htableseg

htableseg — Deprecated.

Description

Deprecated. Use the *htableseg* opcode instead.

Syntax

```
htableseg ifn1, idur1, ifn2 [, idur2] [, ifn3] [...]
```

ktrirand

ktrirand — Deprecated.

Description

Deprecated as of version 3.49. Use the *trirand* opcode instead.

kunirand

kunirand — Deprecated.

Description

Deprecated as of version 3.49. Use the *unirand* opcode instead.

kweibull

kweibull — Deprecated.

Description

Deprecated as of version 3.49. Use the *weibull* opcode instead.

lfo

lfo — A low frequency oscillator of various shapes.

Description

A low frequency oscillator of various shapes.

Syntax

```
kres lfo kamp, kcps [, itype]
```

```
ares lfo kamp, kcps [, itype]
```

Initialization

itype (optional, default=0) -- determine the waveform of the oscillator. Default is 0.

- *itype* = 0 - sine
- *itype* = 1 - triangles
- *itype* = 2 - square (bipolar)
- *itype* = 3 - square (unipolar)
- *itype* = 4 - saw-tooth
- *itype* = 5 - saw-tooth(down)

The sine wave is implemented as a 4096 table and linear interpolation. The others are calculated.

Performance

kamp -- amplitude of output

kcps -- frequency of oscillator

Examples

Here is an example of the lfo opcode. It uses the file *lfo.csd* [examples/lfo.csd].

Example 264. Example of the lfo opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o lfo.wav -W ;;; for file output any platform
</CsOptions>
```

```
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 10
  kcps = 5
  itype = 4

  k1 lfo kamp, kcps, itype
  ar oscil p4, p5+k1, 1
  out ar
endin

</CsInstruments>
<CsScore>

; Table #1: an ordinary sine wave.
f 1 0 32768 10 1

; p4 = amplitude of the output signal.
; p5 = frequency (in cycles per second) of the output signal.
; Play Instrument #1 for two seconds.
i 1 0 2 10000 220
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John fitch
University of Bath/Codemist Ltd.
Bath, UK
November 1998

New in Csound version 3.491

limit

limit — Sets the lower and upper limits of the value it processes.

Description

Sets the lower and upper limits of the value it processes.

Syntax

```
ares limit asig, klow, khigh
```

```
ires limit isig, ilow, ihigh
```

```
kres limit ksig, klow, khigh
```

Initialization

isig -- input signal

ilow -- low threshold

ihigh -- high threshold

Performance

xsig -- input signal

klow -- low threshold

khigh -- high threshold

limit sets the lower and upper limits on the *xsig* value it processes. If *xhigh* is lower than *xlow*, then the output will be the average of the two - it will not be affected by *xsig*.

This opcode is useful in several situations, such as table indexing or for clipping and modeling a-rate, i-rate or k-rate signals.

See Also

mirror, *wrap*

Credits

Author: Robin Whittle
Australia

New in Csound version 3.46

line

line — Trace a straight line between specified points.

Description

Trace a straight line between specified points.

Syntax

```
ares line ia, idur, ib
```

```
kres line ia, idur, ib
```

Initialization

ia -- starting value.

ib -- value after *idur* seconds.

idur -- duration in seconds of segment. A zero or negative value will cause all initialization to be skipped.

Performance

line generates control or audio signals whose values move linearly from an initial value to a final one.



Note

A common error with this opcode is to assume that the value of *ib* is held after the time *idur*. *line* does not automatically end or stop at the end of the duration given. If your note length is longer than *idur* seconds, *kres* (or *ares*) will not come to rest at *ib*, but will instead continue to rise or fall with the same rate. If a rise (or fall) and then hold is required that the *linseg* opcode should be considered instead.

Examples

Here is an example of the line opcode. It uses the file *line.csd* [examples/line.csd].

Example 265. Example of the line opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o line.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
```

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Define kcps as a frequency value that linearly declines
; from 880 to 220. It declines over the period set by p3.
kcps line 880, p3, 220

al oscil 20000, kcps, 1
out al
endin

instr 2
kcps line 880, 1, 660 ; kcps won't stop at 660 if p3 > 1
al oscil 20000, kcps, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2

; Play Instrument #2 for two seconds.
i 2 3 2

e

</CsScore>
</CsoundSynthesizer>
```

See Also

expon, expseg, expsegr, linseg, linsegr

Credits

Example written by Kevin Conder.

linen

linen — Applies a straight line rise and decay pattern to an input amp signal.

Description

linen -- apply a straight line rise and decay pattern to an input amp signal.

Syntax

```
ares linen xamp, irise, idur, idec
```

```
kres linen kamp, irise, idur, idec
```

Initialization

irise -- rise time in seconds. A zero or negative value signifies no rise modification.

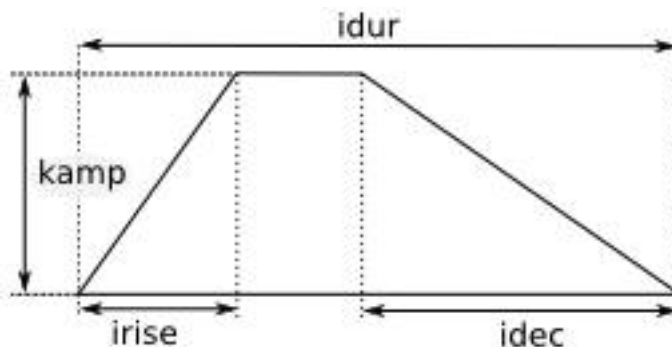
idur -- overall duration in seconds. A zero or negative value will cause initialization to be skipped.

idec -- decay time in seconds. Zero means no decay. An *idec* > *idur* will cause a truncated decay.

Performance

kamp, *xamp* -- input amplitude signal.

Rise modifications are applied for the first *irise* seconds, and decay from time *idur* - *idec*. If these periods are separated in time there will be a steady state during which *amp* will be unmodified. If *linen* rise and decay periods overlap then both modifications will be in effect for that time. If the overall duration *idur* is exceeded in performance, the final decay will continue on in the same direction, going negative.



Envelope generated by the *linen* opcode



Note

A common error with this opcode is to assume that the value of 0 is the held after the envelope has finished at *idur*. *linen* does not automatically end or stop at the end of the duration given. If your note length is longer than *idur* seconds, *kres* (or *ares*) will not come to rest at 0, but will instead continue to fall with the same rate. If a decay and then hold is required then the *linseg* opcode should be considered instead.

See Also

envlpx, envlpxr, linenr

linenr

linenr — The *linen* opcode extended with a final release segment.

Description

linenr -- same as *linen* except that the final segment is entered only on sensing a MIDI note release. The note is then extended by the decay time.

Syntax

```
ares linenr xamp, irise, idec, iatdec
```

```
kres linenr kamp, irise, idec, iatdec
```

Initialization

irise -- rise time in seconds. A zero or negative value signifies no rise modification.

idec -- decay time in seconds. Zero means no decay. An *idec* > *idur* will cause a truncated decay.

iatdec -- attenuation factor by which the closing steady state value is reduced exponentially over the decay period. This value must be positive and is normally of the order of .01. A large or excessively small value is apt to produce a cutoff which is audible. A zero or negative value is illegal.

Performance

kamp, *xamp* -- input amplitude signal.

linenr is unique within Csound in containing a *note-off sensor* and *release time extender*. When it senses either a score event termination or a MIDI noteoff, it will immediately extend the performance time of the current instrument by *idec* seconds, then execute an exponential decay towards the factor *iatdec*. For two or more units in an instrument, extension is by the greatest *idec*.

You can use other pre-made envelopes which start a release segment upon receiving a note off message, like *linsegr* and *expsegr*, or you can construct more complex envelopes using *xtratim* and *release*. Note that you don't need to use *xtratim* if you are using *linenr*, since the time is extended automatically.

These “r” units can also be modified by MIDI noteoff velocities (see *veloffs*).

See Also

linsegr, *expsegr*, *envlpxr*, *mxadsr*, *madsr*, *envlpx*, *linen*, *xtratim*

lineto

lineto — Generate glissandos starting from a control signal.

Description

Generate glissandos starting from a control signal.

Syntax

```
kres lineto ksig, ktime
```

Performance

kres -- Output signal.

ksig -- Input signal.

ktime -- Time length of glissando in seconds.

lineto adds glissando (i.e. straight lines) to a stepped input signal (for example, produced by *randh* or *lpshold*). It generates a straight line starting from previous step value, reaching the new step value in *ktime* seconds. When the new step value is reached, such value is held until a new step occurs. Be sure that *ktime* argument value is smaller than the time elapsed between two consecutive steps of the original signal, otherwise discontinuities will occur in output signal.

When used together with the output of *lpshold* it emulates the glissando effect of old analog sequencers.



Note

No new value for *ksig* or *ktime* will have effect until the previous *ktime* has elapsed.

See Also

tlineto

Credits

Author: Gabriel Maldonado

New in Version 4.13

linrand

linrand — Linear distribution random number generator (positive values only).

Description

Linear distribution random number generator (positive values only). This is an x-class noise generator.

Syntax

```
ares linrand krange
```

```
ires linrand krange
```

```
kres linrand krange
```

Performance

krange -- the range of the random numbers (0 - *krange*). Outputs only positive numbers.

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the linrand opcode. It uses the file *linrand.csd* [examples/linrand.csd].

Example 266. Example of the linrand opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o linrand.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number between 0 and 1.
; krange = 1

il linrand 1
```

```
    print i1
  endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 0.394
```

See Also

seed, betarand, bexprnd, cauchy, exprand, gauss, pcauchy, poisson, trirand, unirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder.

linseg

linseg — Trace a series of line segments between specified points.

Description

Trace a series of line segments between specified points.

Syntax

```
ares linseg ia, idur1, ib [, idur2] [, ic] [...]
```

```
kres linseg ia, idur1, ib [, idur2] [, ic] [...]
```

Initialization

ia -- starting value.

ib, *ic*, etc. -- value after *dur1* seconds, etc.

idur1 -- duration in seconds of first segment. A zero or negative value will cause all initialization to be skipped.

idur2, *idur3*, etc. -- duration in seconds of subsequent segments. A zero or negative value will terminate the initialization process with the preceding point, permitting the last-defined line or curve to be continued indefinitely in performance. The default is zero.

Performance

These units generate control or audio signals whose values can pass through 2 or more specified points. The sum of *dur* values may or may not equal the instrument's performance time: a shorter performance will truncate the specified pattern, while a longer one will cause the last-defined segment to continue on in the same direction.



Note

A common error with this opcode is to assume that the last value is held after the total duration. *linseg* does not automatically end or stop at the end of the total duration. If your note length is longer than the sum of all *idur* values, *kres* (or *ares*) will not come to rest at the last given value, but will instead continue to rise or fall with the current rate. You can add a final segment at the same previous value to create a held final value.

Examples

Here is an example of the *linseg* opcode. It uses the file *linseg.csd* [examples/linseg.csd].

Example 267. Example of the *linseg* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o linseg.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; p4 = frequency in pitch-class notation.
kcps = cpspch(p4)

; Create an amplitude envelope.
kenv linseg 0, p3*0.25, 1, p3*0.75, 0
kamp = kenv * 30000

al oscil kamp, kcps, 1
out al
endin

instr 2
; p4 = frequency in pitch-class notation.
kcps = cpspch(p4)

; Create an amplitude envelope.
kenv linseg 0, 0.25, 1, 0.75, 0 ; kenv will go into negative if p3 > 1
kamp = kenv * 30000

al oscil kamp, kcps, 1
out al
endin

instr 3
; p4 = frequency in pitch-class notation.
kcps = cpspch(p4)

; Create an amplitude envelope.
kenv linseg 0, 0.25, 1, 0.75, 0, 1, 0 ; kenv will stay at 0 indefinitely at the end
kamp = kenv * 30000

al oscil kamp, kcps, 1
out al
endin
</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for a half-second, p4=8.00
i 1 0 0.5 8.00
; Play Instrument #1 for a half-second, p4=8.01
i 1 1 0.5 8.01
; Play Instrument #1 for a half-second, p4=8.02
i 1 2 0.5 8.02
; Play Instrument #1 for a half-second, p4=8.03
i 1 3 0.5 8.03

i 2 4 1.5 8.00 ; Notice the problem with linseg
i 3 6 1.5 8.00 ; this is the solution (instr 3)
e

</CsScore>
</CsoundSynthesizer>
```

See Also

expon, expseg, expsegr, line, linsegr transeg

Credits

Example written by Kevin Conder.

linsegr

linsegr — Trace a series of line segments between specified points including a release segment.

Description

Trace a series of line segments between specified points including a release segment.

Syntax

```
ares linsegr ia, idur1, ib [, idur2] [, ic] [...], irel, iz
```

```
kres linsegr ia, idur1, ib [, idur2] [, ic] [...], irel, iz
```

Initialization

ia -- starting value.

ib, *ic*, etc. -- value after *dur1* seconds, etc.

idur1 -- duration in seconds of first segment. A zero or negative value will cause all initialization to be skipped.

idur2, *idur3*, etc. -- duration in seconds of subsequent segments. A zero or negative value will terminate the initialization process with the preceding point, permitting the last-defined line or curve to be continued indefinitely in performance. The default is zero.

irel, *iz* -- duration in seconds and final value of a note releasing segment.

For Csound versions prior to 5.00, the release time cannot be longer than $32767/kr$ seconds. This limit has been extended to $(2^{31}-1)/kr$.

Performance

These units generate control or audio signals whose values can pass through 2 or more specified points. The sum of *dur* values may or may not equal the instrument's performance time: a shorter performance will truncate the specified pattern, while a longer one will cause the last-defined segment to continue on in the same direction.

linsegr is amongst the Csound “r” units that contain a note-off sensor and release time extender. When each senses an event termination or MIDI noteoff, it immediately extends the performance time of the current instrument by *irel* seconds, and sets out to reach the value *iz* by the end of that period (no matter which segment the unit is in). “r” units can also be modified by MIDI noteoff velocities. For two or more extenders in an instrument, extension is by the greatest period.

You can use other pre-made envelopes which start a release segment upon receiving a note off message, like *linenr* and *expsegr*, or you can construct more complex envelopes using *xtratim* and *release*. Note that you don't need to use *xtratim* if you are using *linsegr*, since the time is extended automatically.

Examples

Here is an example of the *linsegr* opcode. It uses the file *linsegr.csd* [examples/linsegr.csd].

Example 268. Example of the *linsegr* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc       -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o linsegr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; p4 = frequency in pitch-class notation.
kcps = cpspch(p4)

; Use an amplitude envelope with second-long release.
kenv linsegr 1, p3, 0.25, 1, 0
kamp = kenv * 30000

al oscil kamp, kcps, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Make sure the score lasts for four seconds.
f 0 4

; p4 = frequency (in pitch-class notation).
; Play Instrument #1 for a half-second, p4=8.00
i 1 0 0.5 8.00
; Play Instrument #1 for a half-second, p4=8.01
i 1 1 0.5 8.01
; Play Instrument #1 for a half-second, p4=8.02
i 1 2 0.5 8.02
; Play Instrument #1 for a half-second, p4=8.03
i 1 3 0.5 8.03
e

</CsScore>
</CsoundSynthesizer>
```

See Also

linsegr, *expsegr*, *envlpxr*, *mxadsr*, *madsr* *expon*, *expseg*, *expsega* *line*, *linseg*, *xtratim*

Credits

Author: Barry L. Vercoe

Example written by Kevin Conder.

December 2002, December 2006. Thanks to Istvan Varga, added documentation about the maximum release time.

New in Csound 3.47

locsend

locsend — Distributes the audio signals of a previous *locsig* opcode.

Description

locsend depends upon the existence of a previously defined *locsig*. The number of output signals must match the number in the previous *locsig*. The output signals from *locsend* are derived from the values given for distance and reverb in the *locsig* and are ready to be sent to local or global reverb units (see example below). The reverb amount and the balance between the 2 or 4 channels are calculated in the same way as described in the Dodge book (an essential text!).

Syntax

```
a1, a2 locsend
```

```
a1, a2, a3, a4 locsend
```

Examples

```
asig some audio signal
kdegree      line 0, p3, 360
kdistance    line 1, p3, 10
a1, a2, a3, a4 locsig asig, kdegree, kdistance, .1
ar1, ar2, ar3, ar4 locsend
ga1 = ga1+ar1
ga2 = ga2+ar2
ga3 = ga3+ar3
ga4 = ga4+ar4
                                outq  a1, a2, a3, a4
endin

instr 99 ; reverb instrument
a1      reverb2 ga1, 2.5, .5
a2      reverb2 ga2, 2.5, .5
a3      reverb2 ga3, 2.5, .5
a4      reverb2 ga4, 2.5, .5
                                outq  a1, a2, a3, a4

ga1=0
ga2=0
ga3=0
ga4=0
```

In the above example, the signal, *asig*, is sent around a complete circle once during the duration of a note while at the same time it becomes more and more “distant” from the listeners' location. *locsig* sends the appropriate amount of the signal internally to *locsend*. The outputs of the *locsend* are added to global accumulators in a common Csound style and the global signals are used as inputs to the reverb units in a separate instrument.

locsig is useful for quad and stereo panning as well as fixed placed of sounds anywhere between two loudspeakers. Below is an example of the fixed placement of sounds in a stereo field.

```
instr 1
a1, a2      locsig asig, p4, p5, .1
ar1, ar2    locsend
ga1=ga1+ar1
ga2=ga2+ar2
                                outs a1, a
endin
```

```
instr 99
  i reverb....
endin
```

A few notes:

```
;place the sound in the left speaker and near:
il 0 1 0 1

;place the sound in the right speaker and far:
il 1 1 90 25

;place the sound equally between left and right and in the middle ground distance:
il 2 1 45 12
e
```

The next example shows a simple intuitive use of the distance value to simulate Doppler shift. The same value is used to scale the frequency as is used as the distance input to *locsig*.

```
kdistance      line      1, p3, 10
kfreq = (ifreq * 340) / (340 + kdistance)
asig           oscili    iamp, kfreq, 1
kdegree        line      0, p3, 360
a1, a2, a3, a4  locsig    asig, kdegree, kdistance, .1
ar1, ar2, ar3, ar4 locsend
```

See Also

locsig

Credits

Author: Richard Karpen
Seattle, WA USA
1998

New in Csound version 3.48

locsig

locsig — Takes and input signal and distributes between 2 or 4 channels.

Description

locsig takes an input signal and distributes it among 2 or 4 channels using values in degrees to calculate the balance between adjacent channels. It also takes arguments for distance (used to attenuate signals that are to sound as if they are some distance further than the loudspeaker itself), and for the amount the signal that will be sent to reverberators. This unit is based upon the example in the Charles Dodge/Thomas Jerse book, *Computer Music*, page 320.

Syntax

```
a1, a2 locsig asig, kdegree, kdistance, kreverbSEND
```

```
a1, a2, a3, a4 locsig asig, kdegree, kdistance, kreverbSEND
```

Performance

kdegree -- value between 0 and 360 for placement of the signal in a 2 or 4 channel space configured as: a1=0, a2=90, a3=180, a4=270 (kdegree=45 would balanced the signal equally between a1 and a2). *locsig* maps *kdegree* to sin and cos functions to derive the signal balances (ie.: asig=1, kdegree=45, a1=a2=.707).

kdistance -- value >= 1 used to attenuate the signal and to calculate reverb level to simulate distance cues. As *kdistance* gets larger the sound should get softer and somewhat more reverberant (assuming the use of *locSEND* in this case).

kreverbSEND -- the percentage of the direct signal that will be factored along with the distance and degree values to derive signal amounts that can be sent to a reverb unit such as reverb, or reverb2.

Examples

```
asig some audio signal
kdegree      line    0, p3, 360
kdistance    line    1, p3, 10
a1, a2, a3, a4 locsig asig, kdegree, kdistance, .1
ar1, ar2, ar3, ar4 locSEND
ga1 = ga1+ar1
ga2 = ga2+ar2
ga3 = ga3+ar3
ga4 = ga4+ar4
                                outq    a1, a2, a3, a4
endin

instr 99 ; reverb instrument
a1      reverb2 ga1, 2.5, .5
a2      reverb2 ga2, 2.5, .5
a3      reverb2 ga3, 2.5, .5
a4      reverb2 ga4, 2.5, .5
                                outq    a1, a2, a3, a4

ga1=0
ga2=0
ga3=0
ga4=0
```

In the above example, the signal, *asig*, is sent around a complete circle once during the duration of a

note while at the same time it becomes more and more "distant" from the listeners' location. *locsig* sends the appropriate amount of the signal internally to *locsend*. The outputs of the *locsend* are added to global accumulators in a common Csound style and the global signals are used as inputs to the reverb units in a separate instrument.

locsig is useful for quad and stereo panning as well as fixed placed of sounds anywhere between two loudspeakers. Below is an example of the fixed placement of sounds in a stereo field.

```
instr 1
  a1, a2          locsig asig, p4, p5, .1
  ar1, ar2        locsend
  gal=gal+ar1
  ga2=ga2+ar2
                  outs a1, a
endin
instr 99
  ; reverb....
endin
```

A few notes:

```
;place the sound in the left speaker and near:
il 0 1 0 1

;place the sound in the right speaker and far:
il 1 1 90 25

;place the sound equally between left and right and in the middle ground distance:
il 2 1 45 12
e
```

The next example shows a simple intuitive use of the distance value to simulate Doppler shift. The same value is used to scale the frequency as is used as the distance input to *locsig*.

```
kdistance      line 1, p3, 10
kfreq = (ifreq * 340) / (340 + kdistance)
asig           oscili iamp, kfreq, 1
kdegree        line 0, p3, 360
a1, a2, a3, a4  locsig asig, kdegree, kdistance, .1
ar1, ar2, ar3, ar4 locsend
```

See Also

locsend

Credits

Author: Richard Karpen
Seattle, WA USA
1998

New in Csound version 3.48

log

log — Returns a natural log.

Description

Returns the natural log of x (x positive only).

The argument value is restricted for *log*, *log10*, and *sqrt*.

Syntax

`log(x)` (no rate restriction)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

Examples

Here is an example of the log opcode. It uses the file *log.csd* [examples/log.csd].

Example 269. Example of the log opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o log.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = log(8)
  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 2.079
```

See Also

abs, exp, frac, int, log10, i, sqrt

Credits

Written by John ffitch.

New in version 3.47

Example written by Kevin Conder.

log10

log10 — Returns a base 10 log.

Description

Returns the base 10 log of x (x positive only).

The argument value is restricted for *log*, *log10*, and *sqrt*.

Syntax

`log10(x)` (no rate restriction)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

Examples

Here is an example of the log10 opcode. It uses the file *log10.csd* [examples/log10.csd].

Example 270. Example of the log10 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o log10.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = log10(8)
  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 0.903
```

See Also

abs, exp, frac, int, log, i, sqrt

Credits

Written by John ffitch.

New in version 3.47

Example written by Kevin Conder.

logbtwo

logbtwo — Performs a logarithmic base two calculation.

Description

Performs a logarithmic base two calculation.

Syntax

`logbtwo(x)` (init-rate or control-rate args only)

Performance

`logbtwo()` returns the logarithm base two of x . The range of values admitted as argument is .25 to 4 (i.e. from -2 octave to +2 octave response). This function is the inverse of `powoftwo()`.

These functions are fast, because they read values stored in tables. Also they are very useful when working with tuning ratios. They work at i- and k-rate.

Examples

Here is an example of the logbtwo opcode. It uses the file `logbtwo.csd` [examples/logbtwo.csd].

Example 271. Example of the logbtwo opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o logbtwo.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = logbtwo(3)
  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 1.585
```

See Also

powoftwo

Credits

Author: Gabriel Maldonado
Italy
June 1998

Author: John ffitch
University of Bath, Codemist, Ltd.
Bath, UK
July 1999

Example written by Kevin Conder.

New in Csound version 3.57

logcurve

logcurve — This opcode implements a formula for generating a normalised logarithmic curve in range 0 - 1. It is based on the Max / MSP work of Eric Singer (c) 1994.

Description

Generates a logarithmic curve in range 0 to 1 of arbitrary steepness. Steepness index equal to or lower than 1.0 will result in Not-a-Number errors and cause unstable behavior.

The formula used to calculate the curve is:

$$\log(x * (y-1)+1) / (\log(y))$$

where x is equal to *kindex* and y is equal to *ksteepness*.

Syntax

```
kout logcurve kindex, ksteepness
```

Performance

kindex -- Index value. Expected range 0 to 1.

ksteepness -- Steepness of the generated curve. Values closer to 1.0 result in a straighter line while larger values steepen the curve.

kout -- Scaled output.

Examples

Here is an example of the logcurve opcode. It uses the file *logcurve.csd* [examples/logcurve.csd].

Example 272. Example of the logcurve opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  Silent
-odac        -idac     -d     ;;realtime output
</CsOptions>
<CsInstruments>

sr = 48000
ksmps = 100
nchnls = 2

/*--- */

instr 1 ; logcurve test

kmod phasor 1/200

kout logcurve kmod, 2

printk2 kmod
printk2 kout

endin
```



```
/*--- ---*/  
</CsInstruments>  
<CsScore>  
  
i1 0 8888  
  
e  
</CsScore>  
</CsoundSynthesizer>
```

See Also

scale, gainslider, expcurve

Credits

Author: David Akbari
October
2006

loop_ge

loop_ge — Looping constructions.

Description

Construction of looping operations.

Syntax

```
loop_ge    indx, idecr, imin, label
```

```
loop_ge    kndx, kdecr, kmin, label
```

Initialization

indx -- i-rate variable to count loop.

idecr -- value to decrement the loop.

imin -- minimum value of loop index.

Performance

kndx -- k-rate variable to count loop.

kdecr -- value to decrement the loop.

kmin -- minimum value of loop index.

The actions of **loop_ge** are equivalent to the code

```
indx = indx - idecr
if (indx >= imin) igoto label
```

or

```
kndx = kndx - kdecr
if (kndx >= kmin) kgoto label
```

See Also

loop_gt, *loop_le* and *loop_lt*.

Credits

Istvan Varga. 2006

New in Csound version 5.01

loop_gt

loop_gt — Looping constructions.

Description

Construction of looping operations.

Syntax

```
loop_gt    indx, idecr, imin, label
```

```
loop_gt    kndx, kdecr, kmin, label
```

Initialization

indx -- i-rate variable to count loop.

idecr -- value to decrement the loop.

imin -- minimum value of loop index.

Performance

kndx -- k-rate variable to count loop.

kdecr -- value to decrement the loop.

kmin -- minimum value of loop index.

The actions of **loop_gt** are equivalent to the code

```
indx = indx - idecr
if (indx > imin) igoto label
```

or

```
kndx = kndx - kdecr
if (kndx > kmin) kgoto label
```

See Also

loop_ge, *loop_le* and *loop_lt*.

Credits

Istvan Varga.

New in Csound version 5.01

loop_le

loop_le — Looping constructions.

Description

Construction of looping operations.

Syntax

```
loop_le    indx, incr, imax, label
```

```
loop_le    kndx, kncr, kmax, label
```

Initialization

indx -- i-rate variable to count loop.

incr -- value to increment the loop.

imax -- maximum value of loop index.

Performance

kndx -- k-rate variable to count loop.

kncr -- value to increment the loop.

kmax -- maximum value of loop index.

The actions of **loop_le** are equivalent to the code

```
indx = indx + incr
if (indx <= imax) igoto label
```

or

```
kndx = kndx + kncr
if (kndx <= kmax) kgoto label
```

See Also

loop_ge, *loop_gt* and *loop_lt*.

Credits

Istvan Varga.

New in Csound version 5.01

loop_lt

loop_lt — Looping constructions.

Description

Construction of looping operations.

Syntax

```
loop_lt    indx, incr, imax, label
```

```
loop_lt    kndx, kncr, kmax, label
```

Initialization

indx -- i-rate variable to count loop.

incr -- value to increment the loop.

imax -- maximum value of loop index.

Performance

kndx -- k-rate variable to count loop.

kncr -- value to increment the loop.

kmax -- maximum value of loop index.

The actions of **loop_lt** are equivalent to the code

```
indx = indx + incr  
if (indx < imax) igoto label
```

or

```
kndx = kndx + kncr  
if (kndx < kmax) kgoto label
```

See Also

loop_ge, *loop_gt* and *loop_le*.

Credits

Istvan Varga.

New in Csound version 5.01

loopseg

`loopseg` — Generate control signal consisting of linear segments delimited by two or more specified points.

Description

Generate control signal consisting of linear segments delimited by two or more specified points. The entire envelope is looped at *kfreq* rate. Each parameter can be varied at k-rate.

Syntax

```
ksig loopseg kfreq, ktrig, ktime0, kvalue0 [, ktime1] [, kvalue1] \  
      [, ktime2] [, kvalue2] [...]
```

Performance

ksig -- Output signal.

kfreq -- Repeat rate in Hz or fraction of Hz.

ktrig -- If non-zero, retriggers the envelope from start (see *trigger opcode*), before the envelope cycle is completed.

ktime0...ktimeN -- Times of points; expressed in fraction of a cycle.

kvalue0...kvalueN -- Values of points

loopseg opcode is similar to *linseg*, but the entire envelope is looped at *kfreq* rate. Notice that times are not expressed in seconds but in fraction of a cycle. Actually each duration represent is proportional to the other, and the entire cycle duration is proportional to the sum of all duration values.

The sum of all duration is then rescaled according to *kfreq* argument. For example, considering an envelope made up of 3 segments, each segment having 100 as duration value, their sum will be 300. This value represents the total duration of the envelope, and is actually divided into 3 equal parts, a part for each segment.

Actually, the real envelope duration in seconds is determined by *kfreq*. Again, if the envelope is made up of 3 segments, but this time the first and last segments have a duration of 50, whereas the central segment has a duration of 100 again, their sum will be 200. This time 200 represent the total duration of the 3 segments, so the central segment will be twice as long as the other segments.

All parameters can be varied at k-rate. Negative frequency values are allowed, reading the envelope backward. *ktime0* should always be set to 0, except if the user wants some special effect.

Examples

Here is an example of the *loopseg* opcode. It uses the file *loopseg.csd* [examples/loopseg.csd].

Example 273. Example of the *loopseg* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>
```

```
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-o dac      -i adc      -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o loopseg.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1
instr 1
  kfreq line 1, p3, 20

  klp loopseg kfreq, 0, 0, 0, 0.5, 30000, 1, 0

  al oscil klp, 440, 1
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for five seconds.
i 1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

See Also

lpshold

Credits

Author: Gabriel Maldonado

New in Version 4.13

loopsegg

loopsegg — Control signals based on linear segments.

Description

Generate control signal consisting of linear segments delimited by two or more specified points. The entire envelope can be looped at time-variant rate. Each segment coordinate can also be varied at k-rate.

Syntax

```
ksig loopsegg kphase, kvalue0, kdur0, kvalue1 \  
    [, kdur1, ... , kdurN-1, kvalueN]
```

Performance

ksig - output signal

kphase - point of the sequence read, expressed as a fraction of a cycle (0 to 1)

kvalue0 ...kvalueN - values of points

kdur0 ...kdurN-1 - duration of points expressed in fraction of a cycle

loopsegg opcode is similar to *loopseg*; the only difference is that, instead of frequency, a time-variant phase is required. If you use *phasor* to get the phase value, you will have a behaviour identical to *loopseg*, but interesting results can be achieved when using phases having non-linear motions, making *loopsegg* more powerful and general than *loopseg*.

Examples

Here is an example of the *loopsegg* opcode. It uses the file *loopsegg.csd* [examples/loopsegg.csd].

Example 274. Example of the loopsegg opcode.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  
-odac          -iadc      ;;RT audio I/O  
; For Non-realtime output leave only the line below:  
; -o loopsegg.wav -W ;; for file output any platform  
</CsOptions>  
<CsInstruments>  
sr=44100  
ksmps=1  
nchnls=2  
  
; By Mark Van Peteghem 2008  
  
instr 1  
  iphase = p4  
  
  kenv    linen 1, 0.1, p3, 0.1  
  kph_amp phasor 2, 0  
  kamp    loopsegg kph_amp, 60, 1, 30, 1, 60  
  kamp    = ampdb(kamp)*kenv  
  
  kph_freq phasor 2, iphase  
  klow_freq line 200, p3, 100  
  kfreq    loopsegg kph_freq, 400, 1, klow_freq, 1, 400
```



```
asig      vco2 kamp, kfreq, 2, 0.5
          outs asig, asig

        endin

</CsInstruments>
<CsScore>
i1 0 3 0
i1 + . 0.50
</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado.

New in Csound 5. (Previously available only on CsoundAV)

lorenz

lorenz — Implements the Lorenz system of equations.

Description

Implements the Lorenz system of equations. The Lorenz system is a chaotic-dynamic system which was originally used to simulate the motion of a particle in convection currents and simplified weather systems. Small differences in initial conditions rapidly lead to diverging values. This is sometimes expressed as the butterfly effect. If a butterfly flaps its wings in Australia, it will have an effect on the weather in Alaska. This system is one of the milestones in the development of chaos theory. It is useful as a chaotic audio source or as a low frequency modulation source.

Syntax

```
ax, ay, az lorenz ksv, krν, kbν, kh, ix, iy, iz, iskip [, iskipinit]
```

Initialization

ix, iy, iz -- the initial coordinates of the particle.

iskip -- used to skip generated values. If *iskip* is set to 5, only every fifth value generated is output. This is useful in generating higher pitched tones.

iskipinit (optional, default=0) -- if non zero skip the initialisation of the filter. (New in Csound version 4.23f13 and 5.0)

Performance

ksv -- the Prandtl number or sigma

krν -- the Rayleigh number

kbν -- the ratio of the length and width of the box in which the convection currents are generated

kh -- the step size used in approximating the differential equation. This can be used to control the pitch of the systems. Values of .1-.001 are typical.

The equations are approximated as follows:

$$\begin{aligned}x &= x + h*(s*(y - x)) \\y &= y + h*(-x*z + r*x - y) \\z &= z + h*(x*y - b*z)\end{aligned}$$

The historical values of these parameters are:

$$\begin{aligned}ks &= 10 \\kr &= 28 \\kb &= 8/3\end{aligned}$$

Examples

Here is an example of the lorenz opcode. It uses the file *lorenz.csd* [examples/lorenz.csd].

Example 275. Example of the lorenz opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o lorenz.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 2

; Instrument #1 - a lorenz system in 3D space.
instr 1
; Create a basic tone.
kamp init 25000
kcps init 220
ifn = 1
asnd oscil kamp, kcps, ifn

; Figure out its X, Y, Z coordinates.
ksv init 10
krv init 28
kbv init 2.667
kh init 0.0003
ix = 0.6
iy = 0.6
iz = 0.6
iskip = 1
ax1, ay1, az1 lorenz ksv, krv, kbv, kh, ix, iy, iz, iskip

; Place the basic tone within 3D space.
kx downsamp ax1
ky downsamp ay1
kz downsamp az1
idist = 1
ift = 0
imode = 1
imdel = 1.018853416
iovr = 2
aw2, ax2, ay2, az2 spat3d asnd, kx, ky, kz, idist, \
                           ift, imode, imdel, iovr

; Convert the 3D sound to stereo.
aleft = aw2 + ay2
aright = aw2 - ay2

outs aleft, aright
endin

</CsInstruments>
<CsScore>

; Table #1 a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 5 seconds.
i 1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Hans Mikelson
February 1999

New in Csound version 3.53

lorisread

lorisread — Imports a set of bandwidth-enhanced partials from a SDIF-format data file, applying control-rate frequency, amplitude, and bandwidth scaling envelopes, and stores the modified partials in memory.

Syntax

```
lorisread ktmpnt, ifilcod, istoreidx, kfreqenv, kampenv, kbwenv[, ifadetime]
```

Description

lorisread imports a set of bandwidth-enhanced partials from a SDIF-format data file, applying control-rate frequency, amplitude, and bandwidth scaling envelopes, and stores the modified partials in memory.

Initialization

ifilcod - integer or character-string denoting a control-file derived from reassigned bandwidth-enhanced analysis of an audio signal. An integer denotes the suffix of a file *loris.sdif* (e.g. *loris.sdif.1*); a character-string (in double quotes) gives a filename, optionally a full pathname. If not a full pathname, the file is sought first in the current directory, then in the one given by the environment variable *SADIR* (if defined). The reassigned bandwidth-enhanced data file contains breakpoint frequency, amplitude, noisiness, and phase envelope values organized for bandwidth-enhanced additive resynthesis. The control data must conform to one of the SDIF formats that can be

Loris stores partials in SDIF RBEP frames. Each RBEP frame contains one RBEP matrix, and each row in a RBEP matrix describes one breakpoint in a Loris partial. A RBEL frame containing one RBEL matrix describing the labeling of the partials may precede the first RBEP frame in the SDIF file. The RBEP and RBEL frame and matrix definitions are included in the SDIF file's header. In addition to RBEP frames, Loris can also read and write SDIF ITRC frames. Since ITRC frames do not represent bandwidth-enhancement or the exact timing of Loris breakpoints, their use is not recommended. ITRC capabilities are provided to allow interchange with programs that are unable to handle RBEP frames.

istoreidx, *ireadidx*, *isrcidx*, *itgtidx* are labels that identify a stored set of bandwidth-enhanced partials. **lorisread** imports partials from a SDIF file and stores them with the integer label *istoreidx*. **lorismorph** morphs sets of partials labeled *isrcidx* and *itgtidx*, and stores the resulting partials with the integer label *istoreidx*. **lorisplay** renders the partials stored with the label *ireadidx*. The labels are used only at initialization time, and may be reused without any cost or benefit in efficiency, and without introducing any interaction between instruments or instances.

ifadetime (*optional*) - In general, partials exported from Loris begin and end at non-zero amplitude. In order to prevent artifacts, it is very often necessary to fade the partials in and out, instead of turning them abruptly on and off. Specification of a non-zero *ifadetime* causes partials to fade in at their onsets and to fade out at their terminations. This is achieved by adding two more breakpoints to each partial: one *ifadetime* seconds before the start time and another *ifadetime* seconds after the end time. (However, no breakpoint will be introduced at a time less than zero. If necessary, the onset fade time will be shortened.) The additional breakpoints at the partial onset and termination will have the same frequency and bandwidth as the first and last breakpoints in the partial, respectively, but their amplitudes will be zero. The phase of the new breakpoints will be extrapolated to preserve phase correctness. If no value is specified, *ifadetime* defaults to zero. Note that the *ifadetime* may not be exact, since the partial parameter envelopes are sampled at the control rate (*krate*) and indexed by *ktmpnt* (see below), and not by real time.

Performance

`lorisread` reads pre-computed Reassigned Bandwidth-Enhanced analysis data from a file stored in SDIF format, as described above. The passage of time through this file is specified by `ktimpnt`, which represents the time in seconds. `ktimpnt` must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the analysis file. `kfreqenv` is a control-rate transposition factor: a value of 1 incurs no transposition, 1.5 transposes up a perfect fifth, and .5 down an octave. `kampenv` is a control-rate scale factor that is applied to all partial amplitude envelopes. `kbwenv` is a control-rate scale factor that is applied to all partial bandwidth or noisiness envelopes. The bandwidth-enhanced partial data is stored in memory with a specified label for future access by another generator.

Credits

This implementation of the Loris unit generators was written by Kelly Fitz (loris@cerlsoundgroup.org [mailto:loris@cerlsoundgroup.org]). It is patterned after a prototype implementation of the *lorisplay* unit generator written by Corbin Champion, and based on the method of Bandwidth-Enhanced Additive Synthesis and on the sound morphing algorithms implemented in the Loris library for sound modeling and manipulation. The opcodes were further adapted as a plugin for Csound 5 by Michael Gogins.

lorismorph

lorismorph — Morphs two stored sets of bandwidth-enhanced partials and stores a new set of partials representing the morphed sound. The morph is performed by linearly interpolating the parameter envelopes (frequency, amplitude, and bandwidth, or noisiness) of the bandwidth-enhanced partials according to control-rate frequency, amplitude, and bandwidth morphing functions.

Syntax

```
lorismorph isrcidx, itgtidx, istoreidx, kfreqmorphenv, kampmorphenv, kbwmorphenv
```

Description

lorismorph morphs two stored sets of bandwidth-enhanced partials and stores a new set of partials representing the morphed sound. The morph is performed by linearly interpolating the parameter envelopes (frequency, amplitude, and bandwidth, or noisiness) of the bandwidth-enhanced partials according to control-rate frequency, amplitude, and bandwidth morphing functions.

Initialization

istoreidx, *ireadidx*, *isrcidx*, *itgtidx* are labels that identify a stored set of bandwidth-enhanced partials. *lorisread* imports partials from a SDIF file and stores them with the integer label *istoreidx*. *lorismorph* morphs sets of partials labeled *isrcidx* and *itgtidx*, and stores the resulting partials with the integer label *istoreidx*. *lorisplay* renders the partials stored with the label *ireadidx*. The labels are used only at initialization time, and may be reused without any cost or benefit in efficiency, and without introducing any interaction between instruments or instances.

Performance

lorismorph generates a set of bandwidth-enhanced partials by morphing two stored sets of partials, the source and target partials, which may have been imported using *lorisread*, or generated by another unit generator, including another instance of *lorismorph*. The morph is performed by interpolating the parameters of corresponding (labeled) partials in the two source sounds. The sound morph is described by three control-rate morphing envelopes. *kfreqmorphenv* describes the interpolation of partial frequency values in the two source sounds. When *kfreqmorphenv* is 0, partial frequencies are obtained from the partials stored at *isrcidx*. When *kfreqmorphenv* is 1, partial frequencies are obtained from the partials at *itgtidx*. When *kfreqmorphenv* is between 0 and 1, the partial frequencies are interpolated between corresponding source and target partials. Interpolation of partial amplitudes and bandwidth (noisiness) coefficients are similarly described by *kampmorphenv* and *kbwmorphenv*.

Credits

This implementation of the Loris unit generators was written by Kelly Fitz (loris@cerlsoundgroup.org [<mailto:loris@cerlsoundgroup.org>]). It is patterned after a prototype implementation of the *lorisplay* unit generator written by Corbin Champion, and based on the method of Bandwidth-Enhanced Additive Synthesis and on the sound morphing algorithms implemented in the Loris library for sound modeling and manipulation. The opcodes were further adapted as a plugin for Csound 5 by Michael gogins.

lorisplay

lorisplay — renders a stored set of bandwidth-enhanced partials using the method of Bandwidth-Enhanced Additive Synthesis implemented in the Loris software, applying control-rate frequency, amplitude, and bandwidth scaling envelopes.

Syntax

```
ar lorisplay ireadidx, kfrequenv, kampenv, kbwenv
```

Description

lorisplay renders a stored set of bandwidth-enhanced partials using the method of Bandwidth-Enhanced Additive Synthesis implemented in the Loris software, applying control-rate frequency, amplitude, and bandwidth scaling envelopes.

Initialization

istoreidx, *ireadidx*, *isrcidx*, *itgtidx* are labels that identify a stored set of bandwidth-enhanced partials. *lorisread* imports partials from a SDIF file and stores them with the integer label *istoreidx*. *lorismorph* morphs sets of partials labeled *isrcidx* and *itgtidx*, and stores the resulting partials with the integer label *istoreidx*. *lorisplay* renders the partials stored with the label *ireadidx*. The labels are used only at initialization time, and may be reused without any cost or benefit in efficiency, and without introducing any interaction between instruments or instances.

Performance

lorisplay implements signal reconstruction using Bandwidth-Enhanced Additive Synthesis. The control data is obtained from a stored set of bandwidth-enhanced partials imported from an SDIF file using *lorisread* or constructed by another unit generator such as *lorismorph*. *kfrequenv* is a control-rate transposition factor: a value of 1 incurs no transposition, 1.5 transposes up a perfect fifth, and .5 down an octave. *kampenv* is a control-rate scale factor that is applied to all partial amplitude envelopes. *kbwenv* is a control-rate scale factor that is applied to all partial bandwidth or noisiness envelopes. The bandwidth-enhanced partial data is stored in memory with a specified label for future access by another generator.

Credits

This implementation of the Loris unit generators was written by Kelly Fitz (loris@cerlsoundgroup.org [mailto:loris@cerlsoundgroup.org]). It is patterned after a prototype implementation of the *lorisplay* unit generator written by Corbin Champion, and based on the method of Bandwidth-Enhanced Additive Synthesis and on the sound morphing algorithms implemented in the Loris library for sound modeling and manipulation. The opcodes were further adapted as a plugin for Csound 5 by Michael Gogins.

loscil

loscil — Read sampled sound from a table.

Description

Read sampled sound (mono or stereo) from a table, with optional sustain and release looping.

Syntax

```
ar1 [,ar2] loscil xamp, kcps, ifn [, ibas] [, imod1] [, ibeg1] [, iend1] \  
[, imod2] [, ibeg2] [, iend2]
```

Initialization

ifn -- function table number, typically denoting an sampled sound segment with prescribed looping points loaded using *GEN01*. The source file may be mono or stereo.

ibas (optional) -- base frequency in *Hz* of the recorded sound. This optionally overrides the frequency given in the audio file, but is required if the file did not contain one. The default value is 261.626 Hz, i.e. middle C. (New in Csound 4.03). If this value is not known or not present, use 1 here and in *kcps*.

imod1, *imod2* (optional, default=-1) -- play modes for the sustain and release loops. A value of 1 denotes normal looping, 2 denotes forward & backward looping, 0 denotes no looping. The default value (-1) will defer to the mode and the looping points given in the source file. Make sure you select an appropriate mode if the file does not contain this information.

ibeg1, *iend1*, *ibeg2*, *iend2* (optional, dependent on *mod1*, *mod2*) -- begin and end points of the sustain and release loops. These are measured in *sample frames* from the beginning of the file, so will look the same whether the sound segment is monaural or stereo. If no loop points are specified, and a looping mode (*imod1*, *imod2*) is given, the file will be looped for the whole length.

Performance

ar1, *ar2* -- the output at audio-rate. There is just *ar1* for mono output. However, there is both *ar1* and *ar2* for stereo output.

xamp -- the amplitude of the output signal.

kcps -- the frequency of the output signal in cycles per second.

loscil samples the *fable* audio at a-rate determined by *kcps*, then multiplies the result by *xamp*. The sampling increment for *kcps* is dependent on the table's base-note frequency *ibas*, and is automatically adjusted if the orchestra *sr* value differs from that at which the source was recorded. In this unit, *fable* is always sampled with interpolation.

If sampling reaches the *sustain loop* endpoint and looping is in effect, the point of sampling will be modified and *loscil* will continue reading from within that loop segment. Once the instrument has received a *turnoff* signal (from the score or from a MIDI noteoff event), the next sustain endpoint encountered will be ignored and sampling will continue towards the *release loop* end-point, or towards the last sample (henceforth to zeros).

loscil is the basic unit for building a sampling synthesizer. Given a sufficient set of recorded piano tones, for example, this unit can resample them to simulate the missing tones. Locating the sound source nearest a desired pitch can be done via table lookup. Once a sampling instrument has begun, its *turnoff* point may be unpredictable and require an external *release* envelope; this is often done by gating the sampled audio with *linenr*, which will extend the duration of a turned-off instrument by a

specific period while it implements a decay.

If you want to loop the whole file, specify a looping mode in `imod1` and do not enter any values for `ibeg` and `iend`.



Note to Windows users

Windows users typically use back-slashes, “\”, when specifying the paths of their files. As an example, a Windows user might use the path “c:\music\samples\loop001.wav”. This is problematic because back-slashes are normally used to specify special characters.

To correctly specify this path in Csound, one may alternately:

- *Use forward slashes:* c:/music/samples/loop001.wav
- *Use back-slash special characters, “\\”:* c:\\music\\samples\\loop001.wav



Note

This is mono loscil:

```
a1 loscil 10000, 1, 1, 1 ,1
```

...and this is stereo loscil:

```
a1, a2 loscil 10000, 1, 1, 1 ,1
```

Examples

Here is an example of the `loscil` opcode. It uses the file `loscil.csd` [examples/loscil.csd], and `beats.wav` [examples/beats.wav].

Example 276. Example of the `loscil` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o loscil.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
```

```
kamp = 30000
; If you don't know the frequency of your audio file,
; set both the kcps and ibas parameters equal to 1.
kcps = 1
ifn = 1
ibas = 1

a1 loscil kamp, kcps, ifn, ibas
out a1
endin

</CsInstruments>
<CsScore>

; Table #1: an audio file.
; Its table size is deferred,
; and format taken from the soundfile header.
f 1 0 0 1 "beats.wav" 0 0 0

; Play Instrument #1 for 6 seconds.
; This will loop the audio file several times.
i 1 0 6
e

</CsScore>
</CsSoundSynthesizer>
```

See Also

loscil3 and *GEN01*

Credits

Note about the mono/stereo difference was contributed by Rasmus Ekman.

Example written by Kevin Conder.

loscil3

loscil3 — Read sampled sound from a table using cubic interpolation.

Description

Read sampled sound (mono or stereo) from a table, with optional sustain and release looping, using cubic interpolation.

Syntax

```
ar1 [,ar2] loscil3 xamp, kcps, ifn [, ibas] [, imod1] [, ibeg1] [, iend1] \  
[, imod2] [, ibeg2] [, iend2]
```

Initialization

ifn -- function table number, typically denoting an sampled sound segment with prescribed looping points loaded using *GEN01*. The source file may be mono or stereo.

ibas (optional) -- base frequency in *Hz* of the recorded sound. This optionally overrides the frequency given in the audio file, but is required if the file did not contain one. The default value is 261.626 Hz, i.e. middle C. (New in Csound 4.03). If this value is not known or not present, use 1 here and in *kcps*.

imod1, *imod2* (optional, default=-1) -- play modes for the sustain and release loops. A value of 1 denotes normal looping, 2 denotes forward & backward looping, 0 denotes no looping. The default value (-1) will defer to the mode and the looping points given in the source file. Make sure you select an appropriate mode if the file does not contain this information.

ibeg1, *iend1*, *ibeg2*, *iend2* (optional, dependent on *mod1*, *mod2*) -- begin and end points of the sustain and release loops. These are measured in *sample frames* from the beginning of the file, so will look the same whether the sound segment is monaural or stereo. If no loop points are specified, and a looping mode (*imod1*, *imod2*) is given, the file will be looped for the whole length.

Performance

ar1, *ar2* -- the output at audio-rate. There is just *ar1* for mono output. However, there is both *ar1* and *ar2* for stereo output.

xamp -- the amplitude of the output signal.

kcps -- the frequency of the output signal in cycles per second.

loscil3 is identical to *loscil* except that it uses cubic interpolation. New in Csound version 3.50.



Note to Windows users

Windows users typically use back-slashes, “\”, when specifying the paths of their files. As an example, a Windows user might use the path “c:\music\samples\loop001.wav”. This is problematic because back-slashes are normally used to specify special characters.

To correctly specify this path in Csound, one may alternately:

- Use forward slashes: c:/music/samples/loop001.wav
- Use back-slash special characters, “\\”: c:\\music\\samples\\loop001.wav



Note

This is mono loscil3:

```
a1 loscil3 10000, 1, 1, 1, 1
```

...and this is stereo loscil3:

```
a1, a2 loscil3 10000, 1, 1, 1, 1
```

Examples

Here is an example of the loscil3 opcode. It uses the file *loscil3.csd* [examples/loscil3.csd], and *beats.wav* [examples/beats.wav].

Example 277. Example of the loscil3 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc     -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o loscil3.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  ; If you don't know the frequency of your audio file,
  ; set both the kcps and ibas parameters equal to 1.
  kcps = 1
  ifn = 1
  ibas = 1

  a1 loscil3 kamp, kcps, ifn, ibas
  out a1
endin

</CsInstruments>
<CsScore>

; Table #1: an audio file.
; Its table size is deferred,
; and format taken from the soundfile header.
f 1 0 0 1 "beats.wav" 0 0 0

; Play Instrument #1 for 6 seconds.
; This will loop the drum pattern several times.
i 1 0 6
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

loscil and *GEN01*

Credits

Note about the mono/stereo difference was contributed by Rasmus Ekman.

Example written by Kevin Conder.

loscilx

loscilx — Loop oscillator.

Description

This file is currently a stub, but the syntax should be correct.

Syntax

```
ar1 [, ar2, ar3, ar4, ar5, ar6, ar7, ar8, ar9, ar10, ar11, ar12, ar13, ar14, \  
      ar15, ar16] loscilx xamp, kcps, ifn \  
      [, iwsiz, ibas, istr, imod, ibeg, iend]
```

See Also

sndload

loscil

Credits

Written by Istvan Varga.

2006

New in Csound 5.03

lowpass2

lowpass2 — A resonant lowpass filter.

Description

Implementation of a resonant second-order lowpass filter.

Syntax

```
ares lowpass2 asig, kcf, kq [, iskip]
```

Initialization

iskip -- initial disposition of internal data space. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- input signal to be filtered

kcf -- cutoff or resonant frequency of the filter, measured in Hz

kq -- Q of the filter, defined, for bandpass filters, as bandwidth/cutoff. *kq* should be between 1 and 500

lowpass2 is a second order IIR lowpass filter, with k-rate controls for cutoff frequency (*kcf*) and Q (*kq*). As *kq* is increased, a resonant peak forms around the cutoff frequency, transforming the low-pass filter response into a response that is similar to a bandpass filter, but with more low frequency energy. This corresponds to an increase in the magnitude and "sharpness" of the resonant peak. For high values of *kq*, a scaling function such as *balance* may be required. In practice, this allows for the simulation of the voltage-controlled filters of analog synthesizers, or for the creation of a pitch of constant amplitude while filtering white noise.

Examples

Here is an example of the lowpass2 opcode. It uses the file *lowpass2.csd* [examples/lowpass2.csd].

Example 278. Example of the lowpass2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o lowpass2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Sean Costello */
; Orchestra file for resonant filter sweep of a sawtooth-like waveform.
sr = 44100
kr = 2205
ksmps = 20
nchnls = 1
```



```
instr 1
  idur    =      p3
  ifreq   =      p4
  iamp    =      p5 * .5
  iharms  =      (sr*.4) / ifreq

; Sawtooth-like waveform
  asig    gbuzz 1, ifreq, iharms, 1, .9, 1

; Envelope to control filter cutoff
  kfreq   linseg 1, idur * 0.5, 5000, idur * 0.5, 1

  afilt   lowpass2 asig, kfreq, 30

; Simple amplitude envelope
  kenv    linseg 0, .1, iamp, idur -.2, iamp, .1, 0
  out     asig * kenv

endin

</CsInstruments>
<CsScore>

/* Written by Sean Costello */
f1 0 8192 9 1 1 .25

i1 0 5 100 1000
i1 5 5 200 1000
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Sean Costello
Seattle, Washington
August 1999

New in Csound version 4.0

lowres

lowres — Another resonant lowpass filter.

Description

lowres is a resonant lowpass filter.

Syntax

```
ares lowres asig, kcutoff, kresonance [, iskip]
```

Initialization

iskip -- initial disposition of internal data space. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- input signal

kcutoff -- filter cutoff frequency point

kresonance -- resonance amount

lowres is a resonant lowpass filter derived from a Hans Mikelson orchestra. This implementation is much faster than implementing it in Csound language, and it allows *kr* lower than *sr*. *kcutoff* is not in Hz and *kresonance* is not in dB, so experiment for the finding best results.

Examples

Here is an example of the *lowres* opcode. It uses the file *lowres.csd* [examples/lowres.csd] and *beats.wav* [examples/beats.wav].

Example 279. Example of the lowres opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o lowres.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a nice sawtooth waveform.
asig vco 5000, 440, 1
```

```
; Vary the cutoff frequency from 30 to 300 Hz.
kcutoff line 30, p3, 300
kresonance = 10

; Apply the filter.
a1 lowres asig, kcutoff, kresonance

out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave for the vco opcode.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

lowresx

Credits

Author: Gabriel Maldonado (adapted by John ffitch)
Italy

Example written by Kevin Conder.

New in Csound version 3.49

lowresx

lowresx — Simulates layers of serially connected resonant lowpass filters.

Description

lowresx is equivalent to more layers of *lowres* with the same arguments serially connected.

Syntax

```
ares lowresx asig, kcutoff, kresonance [, inumlayer] [, iskip]
```

Initialization

inumlayer -- number of elements in a *lowresx* stack. Default value is 4. There is no maximum.

iskip -- initial disposition of internal data space. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- input signal

kcutoff -- filter cutoff frequency point

kresonance -- resonance amount

lowresx is equivalent to more layer of *lowres* with the same arguments serially connected. Using a stack of a larger number of filters allows a sharper cutoff. This is faster than using a larger number of instances of *lowres* in a Csound orchestra because only one initialization and k cycle are needed at time and the audio loop falls entirely inside the cache memory of processor. Based on an orchestra by Hans Mikelson

Examples

Here is an example of the *lowresx* opcode. It uses the file *lowresx.csd* [examples/lowresx.csd], and *beats.wav* [examples/beats.wav].

Example 280. Example of the *lowresx* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o lowresx.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

```
; Instrument #1 - play the sawtooth waveform through a
; stack of filters.
instr 1
  ; Use a nice sawtooth waveform.
  asig vco 5000, 440, 1

  ; Vary the cutoff frequency from 30 to 300 Hz.
  kcutoff line 30, p3, 300
  kresonance = 3
  inumlayer = 2

  alr lowresx asig, kcutoff, kresonance, inumlayer

  ; It gets loud, so clip the output amplitude to 30,000.
  al clip alr, 1, 30000
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave for the vco opcode.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

lowres

Credits

Author: Gabriel Maldonado (adapted by John ffitich)
Italy

New in Csound version 3.49

lpf18

lpf18 — A 3-pole sweepable resonant lowpass filter.

Description

Implementation of a 3 pole sweepable resonant lowpass filter.

Syntax

```
ares lpf18 asig, kfco, kres, kdist
```

Performance

kfco -- the filter cutoff frequency in Hz. Should be in the range 0 to sr/2.

kres -- the amount of resonance. Self-oscillation occurs when *kres* is approximately 1. Should usually be in the range 0 to 1, however, values slightly greater than 1 are possible for more sustained oscillation and an “overdrive” effect.

kdist -- amount of distortion. *kdist* = 0 gives a clean output. *kdist* > 0 adds *tanh()* distortion controlled by the filter parameters, in such a way that both low cutoff and high resonance increase the distortion amount. Some experimentation is encouraged.

lpf18 is a digital emulation of a 3 pole (18 dB/oct.) lowpass filter capable of self-oscillation with a built-in distortion unit. It is really a 3-pole version of *moogvcf*, retuned, recalibrated and with some performance improvements. The tuning and feedback tables use no more than 6 adds and 6 multiplies per control rate. The distortion unit, itself, is based on a modified *tanh* function driven by the filter controls.



Note

This filter requires that the input signal be normalized to one.

Examples

Here is an example of the lpf18 opcode. It uses the file *lpf18.csd* [examples/lpf18.csd].

Example 281. Example of the lpf18 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o lpf18.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

```
; Instrument #1.
instr 1
; Generate a sine waveform.
; Note that its amplitude (kamp) ranges from 0 to 1.
kamp init 1
kcps init 440
knh init 3
ifn = 1
asine buzz kamp, kcps, knh, ifn

; Filter the sine waveform.
; Vary the cutoff frequency (kfco) from 300 to 3,000 Hz.
kfco line 300, p3, 3000
kres init 0.8
kdist init 0.3
aout lpf18 asine, kfco, kres, kdist

out aout * 30000
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for four seconds.
i 1 0 4
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Josep M Comajuncosas
Spain
December 2000

Example written by Kevin Conder with help from Iain Duncan. Thanks goes to Iain for helping with the example.

New in Csound version 4.10

lpfreson

lpfreson — Resynthesises a signal from the data passed internally by a previous *lpread*, applying formant shifting.

Description

Resynthesises a signal from the data passed internally by a previous *lpread*, applying formant shifting.

Syntax

```
ares lpfreson asig, kfrqratio
```

Performance

asig -- an audio driving function for resynthesis.

kfrqratio -- frequency ratio. Must be greater than 0.

lpfreson receives values internally produced by a leading *lpread*. *lpread* gets its values from the control file according to the input value *ktimpnt* (in seconds). If *ktimpnt* proceeds at the analysis rate, time-normal synthesis will result; proceeding at a faster, slower, or variable rate will result in time-warped synthesis. At each k-period, *lpread* interpolates between adjacent frames to more accurately determine the parameter values (presented as output) and the filter coefficient settings (passed internally to a subsequent *lpreson*).

The error signal *kerr* (between 0 and 1) derived during predictive analysis reflects the deterministic/random nature of the analyzed source. This will emerge low for pitched (periodic) material and higher for noisy material. The transition from voiced to unvoiced speech, for example, produces an error signal value of about .3. During synthesis, the error signal value can be used to determine the nature of the *lpreson* driving function: for example, by arbitrating between pitched and non-pitched input, or even by determining a mix of the two. In normal speech resynthesis, the pitched input to *lpreson* is a wideband periodic signal or pulse train derived from a unit such as *buzz*, and the nonpitched source is usually derived from *rand*. However, any audio signal can be used as the driving function, the only assumption of the analysis being that it has a flat response.

lpfreson is a formant shifted *lpreson*, in which *kfrqratio* is the (cps) ratio of shifted to original formant positions. This permits synthesis in which the source object changes its apparent acoustic size. *lpfreson* with *kfrqratio* = 1 is equivalent to *lpreson*.

Generally, *lpreson* provides a means whereby the time-varying content and spectral shaping of a composite audio signal can be controlled by the dynamic spectral content of another. There can be any number of *lpread*/*lpreson* (or *lpfreson*) pairs in an instrument or in an orchestra; they can read from the same or different control files independently.

See Also

lpread, *lpreson*

lphasor

lphasor — Generates a table index for sample playback

Description

This opcode can be used to generate table index for sample playback (e.g. tablexkt).

Syntax

```
ares lphasor xtrns [, ilps] [, ilpe] [, imode] [, istrtr] [, istor]
```

Initialization

ilps -- loop start.

ilpe -- loop end (must be greater than *ilps* to enable looping). The default value of *ilps* and *ilpe* is zero.

imode (optional: default = 0) -- loop mode. Allowed values are:

- 0: no loop
- 1: forward loop
- 2: backward loop
- 3: forward-backward loop

istrtr (optional: default = 0) -- The initial output value (phase). It must be less than *ilpe* if looping is enabled, but is allowed to be greater than *ilps* (i.e. you can start playback in the middle of the loop).

istor (optional: default = 0) -- skip initialization if set to any non-zero value.

Performance

ares -- a raw table index in samples (same unit for loop points). Can be used as index with the table opcodes.

xtrns -- transpose factor, expressed as a playback ratio. *ares* is incremented by this value, and wraps around loop points. For example, 1.5 means a fifth above, 0.75 means fourth below. It is not allowed to be negative.

Examples

Here is an example of the lphasor opcode. It uses the file *lphasor.csd* [examples/lphasor.csd].

Example 282. Example of the lphasor opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
```

```
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o lphashor.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
; Example by Jonathan Murphy Dec 2006

sr      = 44100
ksmps   = 10
nchnls  = 1

instr 1

ifn      = 1 ; table number
ilen     = nsamp(ifn) ; return actual number of samples in table
itrns    = 1 ; no transposition
ilps     = 0 ; loop starts at index 0
ilpe     = ilen ; ends at value returned by nsamp above
imode    = 3 ; loop forwards & backwards
istrt    = 10000 ; commence playback at index 10000 samples
; lphasor provides index into f1
alphs    lphasor itrns, ilps, ilpe, imode, istrt
atab     tablei alphs, ifn
; amplify signal
atab     = atab * 10000

out      atab

endin

</CsInstruments>
<CsScore>
f 1 0 262144 1 "beats.wav" 0 4 1
i1 0 60
e
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Istvan Varga
January 2002
Example by: Jonathan Murphy

New in version 4.18

Updated April 2002 and November 2002 by Istvan Varga

lpinterp

lpslot, lpinterp — Computes a new set of poles from the interpolation between two analysis.

Description

Computes a new set of poles from the interpolation between two analysis.

Syntax

```
lpinterp islot1, islot2, kmix
```

Initialization

islot1 -- slot where the first analysis was stored

islot2 -- slot where the second analysis was stored

kmix -- mix value between the two analysis. Should be between 0 and 1. 0 means analysis 1 only. 1 means analysis 2 only. Any value in between will produce interpolation between the filters.

lpinterp computes a new set of poles from the interpolation between two analysis. The poles will be stored in the current *lpslot* and used by the next *lpreson* opcode.

Examples

Here is a typical orc using the opcodes:

```
ipower init 50000 ; Define sound generator
ifreq  init 440
asrc  buzz ipower,ifreq,10,1

ktime  line 0,p3,p3          ; Define time lin
       lpslot 0             ; Read square data poles
krmsr,krms0,kerr,kcps lpread ktime,"square.pol"
       lpslot 1             ; Read triangle data poles
krmsr,krms0,kerr,kcps lpread ktime,"triangle.pol"
kmix  line 0,p3,1           ; Compute result of mixing
       lpinterp 0,1,kmix    ; and balance power
ares  lpreson asrc
aout  balance ares,asrc
       out aout
```

See Also

lpslot

Credits

Author: Gabriel Maldonado

lposcil

lposcil, lposcil3 — Read sampled sound from a table with optional looping and high precision.

Description

Read sampled sound (mono or stereo) from a table, with optional sustain and release looping, and high precision.

Syntax

```
ares lposcil kamp, kfregratio, kloop, kend, ifn [, iphs]
```

Initialization

ifn -- function table number

Performance

kamp -- amplitude

kfregratio -- multiply factor of table frequency (for example: 1 = original frequency, 1.5 = a fifth up, .5 = an octave down)

kloop -- loop point (in samples)

kend -- end loop point (in samples)

lposcil (looping precise oscillator) allows varying at k-rate, the starting and ending point of a sample contained in a table (*GEN01*). This can be useful when reading a sampled loop of a wavetable, where repeat speed can be varied during the performance.

See Also

lposcil3, *lposcila*, *lposcilsa*, *lposcilsa2*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.52

lposcil3

lposcil3 — Read sampled sound from a table with high precision and cubic interpolation.

Description

Read sampled sound (mono or stereo) from a table, with optional sustain and release looping, and high precision. *lposcil3* uses cubic interpolation.

Syntax

```
ares lposcil3 kamp, kfregratio, kloop, kend, ifn [, iphs]
```

Initialization

ifn -- function table number

Performance

kamp -- amplitude

kfregratio -- multiply factor of table frequency (for example: 1 = original frequency, 1.5 = a fifth up, .5 = an octave down)

kloop -- loop point (in samples)

kend -- end loop point (in samples)

lposcil (looping precise oscillator) allows varying at k-rate, the starting and ending point of a sample contained in a table (*GEN01*). This can be useful when reading a sampled loop of a wavetable, where repeat speed can be varied during the performance.

See Also

lposcil

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.52

lposcila

lposcila — Read sampled sound from a table with optional looping and high precision.

Description

lposcila reads sampled sound from a table with optional looping and high precision.

Syntax

```
ar lposcila aamp, kfregratio, kloop, kend, ift [,iphs]
```

Initialization

ift - function table number

iphs - initial phase (in samples)

Performance

ar - output signal

aamp - amplitude

kfregratio - multiply factor of table frequency (for example: 1 = original frequency, 1.5 = a fifth up , .5 = an octave down)

kloop - loop point (in samples)

kend - end loop point (in samples)

lposcila is the same as *lposcil*, but has an audio-rate amplitude argument (instead of k-rate) to allow fast envelope transients.

See Also

lposcil, *lposcilsa*, *lposcilsa2*

Credits

Author: Gabriel Maldonado

New in version 5.06

lposcilsa

lposcilsa — Read stereo sampled sound from a table with optional looping and high precision.

Description

lposcilsa reads stereo sampled sound from a table with optional looping and high precision.

Syntax

```
ar1, ar2 lposcilsa aamp, kfregratio, kloop, kend, ift [,iphs]
```

Initialization

ift - function table number

iphs - initial phase (in samples)

Performance

ar1, ar2 - output signal

aamp - amplitude

kfregratio - multiply factor of table frequency (for example: 1 = original frequency, 1.5 = a fifth up , .5 = an octave down)

kloop - loop point (in samples)

kend - end loop point (in samples)

lposcilsa is the same as *lposcila*, but works with stereo files loaded with *GEN01*.

See Also

lposcil, lposcila, lposcilsa2

Credits

Author: Gabriel Maldonado

New in version 5.06

lposcilsa2

lposcilsa2 — Read stereo sampled sound from a table with optional looping and high precision.

Description

lposcilsa2 reads stereo sampled sound from a table with optional looping and high precision.

Syntax

```
ar1, ar2 lposcilsa2 aamp, kfreqratio, kloop, kend, ift [,iphs]
```

Initialization

ift - function table number

iphs - initial phase (in samples)

Performance

ar1, ar2 - output signal

aamp - amplitude

kfreqratio - multiply factor of table frequency (for example: 1 = original frequency, 2 = an octave up). Only integers are allowed

kloop - loop point (in samples)

kend - end loop point (in samples)

lposcilsa2 is the same as *lposcilsa*, but no interpolation is implemented and only works with integer *kfreqratio* values. Much faster than *lposcilsa*, it is mainly intended to be used with *kfreqratio* = 1, being in this case a fast substitute of *soundin*, since the soundfile must be entirely loaded in memory.

See Also

lposcil, *lposcila*, *lposcilsa*

Credits

Author: Gabriel Maldonado

New in version 5.06

lpread

lpread — Reads a control file of time-ordered information frames.

Description

Reads a control file of time-ordered information frames.

Syntax

```
krmsr, krms0, kerr, kcps lpread ktimpnt, ifilcod [, inpoles] [, ifrmrate]
```

Initialization

ifilcod -- integer or character-string denoting a control-file (reflection coefficients and four parameter values) derived from n-pole linear predictive spectral analysis of a source audio signal. An integer denotes the suffix of a file *lp.m*; a character-string (in double quotes) gives a filename, optionally a full pathname. If not fullpath, the file is sought first in the current directory, then in that of the environment variable SADIR (if defined). Memory usage depends on the size of the file, which is held entirely in memory during computation but shared by multiple calls (see also *adsyn*, *pvoc*).

inpoles (optional, default=0) -- number of poles in the lpc analysis. It is required only when the control file does not have a header; it is ignored when a header is detected.

ifrmrate (optional, default=0) -- frame rate per second in the lpc analysis. It is required only when the control file does not have a header; it is ignored when a header is detected.

Performance

lpread accesses a control file of time-ordered information frames, each containing n-pole filter coefficients derived from linear predictive analysis of a source signal at fixed time intervals (e.g. 1/100 of a second), plus four parameter values:

krmsr -- root-mean-square (rms) of the residual of analysis

krms0 -- rms of the original signal

kerr -- the normalized error signal

kcps -- pitch in Hz

ktimpnt -- The passage of time, in seconds, through the analysis file. *ktimpnt* must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the analysis file.

lpread gets its values from the control file according to the input value *ktimpnt* (in seconds). If *ktimpnt* proceeds at the analysis rate, time-normal synthesis will result; proceeding at a faster, slower, or variable rate will result in time-warped synthesis. At each k-period, *lpread* interpolates between adjacent frames to more accurately determine the parameter values (presented as output) and the filter coefficient settings (passed internally to a subsequent *lpreson*).

The error signal *kerr* (between 0 and 1) derived during predictive analysis reflects the deterministic/random nature of the analyzed source. This will emerge low for pitched (periodic) material and higher for noisy material. The transition from voiced to unvoiced speech, for example, produces an error signal value of about .3. During synthesis, the error signal value can be used to determine the nature of the *lpreson* driving function: for example, by arbitrating between pitched and non-pitched input, or even by determining a mix of the two. In normal speech resynthesis, the pitched input to *lpreson* is a wideband periodic signal or pulse train derived from a unit such as *buzz*, and the nonpitched

source is usually derived from *rand*. However, any audio signal can be used as the driving function, the only assumption of the analysis being that it has a flat response.

lpfreson is a formant shifted *lpreson*, in which *kfrqratio* is the (cps) ratio of shifted to original formant positions. This permits synthesis in which the source object changes its apparent acoustic size. *lpfreson* with *kfrqratio* = 1 is equivalent to *lpreson*.

Generally, *lpreson* provides a means whereby the time-varying content and spectral shaping of a composite audio signal can be controlled by the dynamic spectral content of another. There can be any number of *lpread/lpreson* (or *lpfreson*) pairs in an instrument or in an orchestra; they can read from the same or different control files independently.

See Also

lpfreson, *lpreson*, *LPANAL*

lpreson

lpreson — Resynthesises a signal from the data passed internally by a previous **lpread**.

Description

Resynthesises a signal from the data passed internally by a previous **lpread**.

Syntax

```
ares lpreson asig
```

Performance

asig -- an audio driving function for resynthesis.

lpreson receives values internally produced by a leading *lpread*. *lpread* gets its values from the control file according to the input value *ktimpnt* (in seconds). If *ktimpnt* proceeds at the analysis rate, time-normal synthesis will result; proceeding at a faster, slower, or variable rate will result in time-warped synthesis. At each k-period, *lpread* interpolates between adjacent frames to more accurately determine the parameter values (presented as output) and the filter coefficient settings (passed internally to a subsequent *lpreson*).

The error signal *kerr* (between 0 and 1) derived during predictive analysis reflects the deterministic/random nature of the analyzed source. This will emerge low for pitched (periodic) material and higher for noisy material. The transition from voiced to unvoiced speech, for example, produces an error signal value of about .3. During synthesis, the error signal value can be used to determine the nature of the *lpreson* driving function: for example, by arbitrating between pitched and non-pitched input, or even by determining a mix of the two. In normal speech resynthesis, the pitched input to *lpreson* is a wideband periodic signal or pulse train derived from a unit such as *buzz*, and the nonpitched source is usually derived from *rand*. However, any audio signal can be used as the driving function, the only assumption of the analysis being that it has a flat response.

lpfreson is a formant shifted *lpreson*, in which *kfrqratio* is the (cps) ratio of shifted to original formant positions. This permits synthesis in which the source object changes its apparent acoustic size. *lpfreson* with *kfrqratio* = 1 is equivalent to *lpreson*.

Generally, *lpreson* provides a means whereby the time-varying content and spectral shaping of a composite audio signal can be controlled by the dynamic spectral content of another. There can be any number of *lpread*/*lpreson* (or *lpfreson*) pairs in an instrument or in an orchestra; they can read from the same or different control files independently.

See Also

lpfreson, *lpread*

lpshold

lpshold — Generate control signal consisting of held segments.

Description

Generate control signal consisting of held segments delimited by two or more specified points. The entire envelope is looped at *kfreq* rate. Each parameter can be varied at k-rate.

Syntax

```
ksig lpshold kfreq, ktrig, ktime0, kvalue0 [, ktime1] [, kvalue1] \
      [, ktime2] [, kvalue2] [...]
```

Performance

ksig -- Output signal

kfreq -- Repeat rate in Hz or fraction of Hz

ktrig -- If non-zero, retriggers the envelope from start (see *trigger opcode*), before the envelope cycle is completed.

ktime0...ktimeN -- Times of points; expressed in fraction of a cycle

kvalue0...kvalueN -- Values of points

lpshold is similar to *loopseg*, but can generate only horizontal segments, i.e. holds values for each time interval placed between *ktimeN* and *ktimeN+1*. It can be useful, among other things, for melodic control, like old analog sequencers.

Examples

Here is an example of the lpshold opcode. It uses the file *lpshold.csd* [examples/lpshold.csd].

Example 283. Example of the lpshold opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o lpshold.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1
instr 1
  kfreq line 1, p3, 20

  klp lpshold kfreq, 0, 0, 0, p3*0.25, 20000, p3*0.75, 0
```

```
al oscil klp, 440, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for five seconds.
i 1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

See Also

loopseg

Credits

Author: Gabriel Maldonado

New in Version 4.13

lpsholdp

lpsholdp — Control signals based on held segments.

Description

Generate control signal consisting of held segments delimited by two or more specified points. The entire envelope can be looped at time-variant rate. Each segment coordinate can also be varied at k-rate.

Syntax

```
ksig lpsholdp kphase, ktrig, ktime0, kvalue0 [, ktime1] [, kvalue1] \  
      [, ktime2] [, kvalue2] [...]
```

Performance

ksig - output signal

kphase -

kvalue0 ...*kvalueN* - values of points

ktime0 ...*ktimeN* - times of points expressed in fraction of a cycle

lpsholdp opcode is similar to *lpshold*; the only difference is that, instead of frequency, a time-variant phase is required. If you use a phasor to get the phase value, you will have a behaviour identical to *lpshold*, but interesting results can be achieved when using phases having non-linear motions, making *lpsholdp* more powerful and general than *lpshold*.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

lpslot

lpslot — Selects the slot to be use by further lp opcodes.

Description

Selects the slot to be use by further lp opcodes.

Syntax

```
lpslot islot
```

Initialization

islot -- number of slot to be selected.

Performance

lpslot selects the slot to be use by further lp opcodes. This is the way to load and reference several analyses at the same time.

Examples

Here is a typical orc using the opcodes:

```
ipower init 50000 ; Define sound generator
ifreq  init 440
asrc   buzz ipower,ifreq,10,1

ktime  line 0,p3,p3 ; Define time lin
       lpslot 0 ; Read square data poles
krmsr,krms0,kerr,kcps lpread ktime,"square.pol"
       lpslot 1 ; Read triangle data poles
krmsr,krms0,kerr,kcps lpread ktime,"triangle.pol"
kmix   line 0,p3,1 ; Compute result of mixing
       lpinterp 0,1,kmix ; and balance power
ares   lpreson asrc
aout   balance ares,asrc
out    aout
```

See Also

lpinterp

Credits

Author: Mark Resibois
Brussels
1996

New in version 3.44

mac

mac — Multiplies and accumulates a- and k-rate signals.

Description

Multiplies and accumulates a- and k-rate signals.

Syntax

```
ares mac asig1, ksig1 [, asig2] [, ksig2] [, asig3] [, ksig3] [...]
```

Performance

ksig1, etc. -- k-rate input signals

asig1, etc. -- a-rate input signals

mac multiplies and accumulates a- and k-rate signals. It is equivalent to:

$$\text{ares} = \text{asig1} * \text{ksig1} + \text{asig2} * \text{ksig2} + \text{asig3} * \text{ksig3} + \dots$$

See Also

maca

Credits

Author: John ffitch
University of Bath, Codemist, Ltd.
Bath, UK
May 1999

New in Csound version 3.54

maca

maca — Multiply and accumulate a-rate signals only.

Description

Multiply and accumulate a-rate signals only.

Syntax

```
ares maca asig1 , asig2 [, asig3] [, asig4] [, asig5] [...]
```

Performance

asig1, asig2, ... -- a-rate input signals

maca multiplies and accumulates a-rate signals only. It is equivalent to:

$$\text{ares} = \text{asig1} * \text{asig2} + \text{asig3} * \text{asig4} + \text{asig5} * \text{asig6} + \dots$$

See Also

mac

Credits

Author: John ffitch
University of Bath, Codemist, Ltd.
Bath, UK
May 1999

New in Csound version 3.54

madsr

madsr — Calculates the classical ADSR envelope using the *linsegr* mechanism.

Description

Calculates the classical ADSR envelope using the *linsegr* mechanism.

Syntax

```
ares madsr iatt, idec, islev, irel [, idel] [, ireltim]
```

```
kres madsr iatt, idec, islev, irel [, idel] [, ireltim]
```

Initialization

iatt -- duration of attack phase

idec -- duration of decay

islev -- level for sustain phase

irel -- duration of release phase.

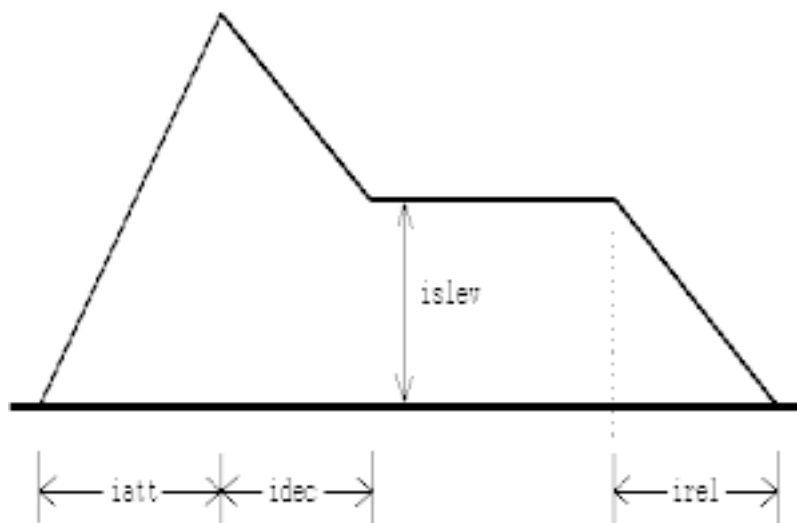
idel -- period of zero before the envelope starts

ireltim (optional, default=-1) -- Control release time after receiving a MIDI noteoff event. If less than zero, the longest release time given in the current instrument is used. If zero or more, the given value will be used for release time. Its default value is -1. (New in Csound 3.59 - not yet properly tested)

Please note that the release time cannot be longer than $32767/kr$ seconds.

Performance

The envelope is in the range 0 to 1 and may need to be scaled further. The envelope may be described as:



Picture of an ADSR envelope.

The length of the sustain is calculated from the length of the note. This means *adsr* is not suitable for use with MIDI events. The opcode *madsr* uses the *linsegr* mechanism, and so can be used in MIDI applications.

You can use other pre-made envelopes which start a release segment upon receiving a note off message, like *linsegr* and *expsegr*, or you can construct more complex envelopes using *xtratim* and *release*. Note that you don't need to use *xtratim* if you are using *madsr*, since the time is extended automatically.

Examples

Here is an example of the *madsr* opcode. It uses the file *madsr.csd* [examples/madsr.csd].

Example 284. Example of the *madsr* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
-o madsr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Iain McCurdy */
; Initialize the global variables.
sr = 44100
kr = 441
ksmps = 100
nchnls = 1

; Instrument #1.
instr 1
; Attack time.
iattack = 0.5
; Decay time.
idecay = 0
; Sustain level.
isustain = 1
; Release time.
irelease = 0.5
aenv madsr iattack, idecay, isustain, irelease

al oscili 10000, 440, 1
out al*aenv
endin

</CsInstruments>
<CsScore>

/* Written by Iain McCurdy */
; Table #1, a sine wave.
f 1 0 1024 10 1

; Leave the score running for 6 seconds.
f 0 6

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

linsegr, expsegr, envlpxr, mxadsr, madsr, xadsr expon, expseg, expsega line, linseg, xtratim

Credits

Author: John ffitch

November 2002. Thanks to Rasmus Ekman, added documentation for the *ireltim* parameter.

December 2002. Thanks to Iain McCurdy, added an example.

December 2002. Thanks to Istvan Varga, added documentation about the maximum release time.

New in Csound version 3.49.

mandel

mandel — Mandelbrot set

Description

Returns the number of iterations corresponding to a given point of complex plane by applying the Mandelbrot set formula.

Syntax

```
kiter, koutrig mandel ktrig, kx, ky, kmaxIter
```

Performance

kiter - number of iterations

koutrig - output trigger signal

ktrig - input trigger signal

kx, ky - coordinates of a given point belonging to the complex plane

kmaxIter - maximum iterations allowed

mandel is an opcode that allows the use of the Mandelbrot set formula to generate an output that can be applied to any musical (or non-musical) parameter. It has two output arguments: *kiter*, that contains the iteration number of a given point, and *koutrig*, that generates a trigger 'bang' each time *kiter* changes. A new number of iterations is evaluated only when *ktrig* is set to a non-zero value. The coordinates of the complex plane are set in *kx* and *ky*, while *kmaxIter* contains the maximum number of iterations. Output values, which are integer numbers, can be mapped in any sorts of ways by the composer.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

mandol

mandol — An emulation of a mandolin.

Description

An emulation of a mandolin.

Syntax

```
ares mandol kamp, kfreq, kpluck, kdetune, kgain, ksize, ifn [, iminfreq]
```

Initialization

ifn -- table number containing the pluck wave form. The file *mandpluk.aiff* [examples/mandpluk.aiff] is suitable for this. It is also available at <ftp://ftp.cs.bath.ac.uk/pub/dream/documentation/sounds/modelling/>.

iminfreq (optional, default=0) -- Lowest frequency to be played on the note. If it is omitted it is taken to be the same as the initial *kfreq*.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kpluck -- The pluck position, in range 0 to 1. Suggest 0.4.

kdetune -- The proportional detuning between the two strings. Suggested range 0.9 to 1.

kgain -- the loop gain of the model, in the range 0.97 to 1.

ksize -- The size of the body of the mandolin. Range 0 to 2.

Examples

Here is an example of the mandol opcode. It uses the file *mandol.csd* [examples/mandol.csd], and *mandpluk.aiff* [examples/mandpluk.aiff].

Example 285. Example of the mandol opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc          -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o mandol.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
```

```
kmps = 10
nchnls = 1

; Instrument #1.
instr 1
  ; kamp = 30000
  ; kfreq = 880
  ; kpluck = 0.4
  ; kdetune = 0.99
  ; kgain = 0.99
  ; ksize = 2
  ; ifn = 1
  ; ifreq = 220

  al mandol 30000, 880, 0.4, 0.99, 0.99, 2, 1, 220

  out al
endin

</CsInstruments>
<CsScore>

; Table #1: the "mandpluk.aiff" audio file
f 1 0 8192 1 "mandpluk.aiff" 0 0 0

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.47

marimba

marimba — Physical model related to the striking of a wooden block.

Description

Audio output is a tone related to the striking of a wooden block as found in a marimba. The method is a physical model developed from Perry Cook but re-coded for Csound.

Syntax

```
ares marimba kamp, kfreq, ihrd, ipos, imp, kvibf, kvamp, ivibfn, idec \  
      [, idoubles] [, itriples]
```

Initialization

ihrd -- the hardness of the stick used in the strike. A range of 0 to 1 is used. 0.5 is a suitable value.

ipos -- where the block is hit, in the range 0 to 1.

imp -- a table of the strike impulses. The file *marmstk1.wav* [examples/marmstk1.wav] is a suitable function from measurements and can be loaded with a *GEN01* table. It is also available at <ftp://ftp.cs.bath.ac.uk/pub/dream/documentation/sounds/modelling/>.

ivfn -- shape of vibrato, usually a sine table, created by a function

idec -- time before end of note when damping is introduced

idoubles (optional) -- percentage of double strikes. Default is 40%.

itriples (optional) -- percentage of triple strikes. Default is 20%.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kvibf -- frequency of vibrato in Hertz. Suggested range is 0 to 12

kvamp -- amplitude of the vibrato

Examples

Here is an example of the marimba opcode. It uses the file *marimba.csd* [examples/marimba.csd], and *marmstk1.wav* [examples/marmstk1.wav].

Example 286. Example of the marimba opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in    No messages
```



```
-odac          -iadc      -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o marimba.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
ksmps = 128
nchnls = 2

; Instrument #1.
instr 1
  ifreq = cpspch(p4)
  ihrd = 0.1
  ipos = 0.561
  imp = 1
  kvibf = 6.0
  kvamp = 0.05
  ivibfn = 2
  idec = 0.6

  al marimba 20000, ifreq, ihrd, ipos, imp, kvibf, kvamp, ivibfn, idec, 20, 10

  outs al, al
endin

</CsInstruments>
<CsScore>

; Table #1, the "marmstk1.wav" audio file.
f 1 0 256 1 "marmstk1.wav" 0 0 0
; Table #2, a sine wave for the vibrato.
f 2 0 128 10 1

; Play Instrument #1 for one second.
i 1 0 1 8.09
i 1 + 0.5 8.00
i 1 + 0.5 7.00
i 1 + 0.25 8.02
i 1 + 0.25 8.01
i 1 + 0.25 7.09
i 1 + 0.25 8.02
i 1 + 0.25 8.01
i 1 + 0.25 7.09
i 1 + 0.3333 8.09
i 1 + 0.3333 8.02
i 1 + 0.3334 8.01
i 1 + 0.25 8.00
i 1 + 0.3333 8.09
i 1 + 0.3333 8.02
i 1 + 0.25 8.01
i 1 + 0.3333 7.00
i 1 + 0.3334 6.00

e

</CsScore>
</CsoundSynthesizer>
```

See Also

vibes

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

massign

massign — Assigns a MIDI channel number to a Csound instrument.

Description

Assigns a MIDI channel number to a Csound instrument.

Syntax

```
massign ichnl, insnum[, ireset]
```

```
massign ichnl, "insname"[, ireset]
```

Initialization

ichnl -- MIDI channel number (1-16).

insnum -- Csound orchestra instrument number. If zero or negative, the channel is muted (i.e. it doesn't trigger a csound instrument, though information will still be received by opcodes like *midin*).

"insname" -- A string (in double-quotes) representing a named instrument.

ireset -- If non-zero resets the controllers; default is to reset.

Performance

Assigns a MIDI channel number to a Csound instrument. Also useful to make sure a certain instrument (if its number is from 1 to 16) will not be triggered by midi noteon messages (if using something *midin* to interpret midi information). In this case set *insnum* to 0 or a negative number.

If *ichan* is set to 0, the value of *insnum* is used for all channels. This way you can route all MIDI channels to a single Csound instrument. You can also disable triggering of instruments from MIDI note events from all channels with the following line:

```
massign 0,0
```

This can be useful if you are doing all MIDI evaluation within Csound with an always on instrument(e.g. using *midin* and *turnon*) to avoid doubling the instrument when a note is played.

See Also

ctrlinit

Credits

Author: Barry L. Vercoe - Mike Berry
MIT, Cambridge, Mass.

New in Csound version 3.47

ireset parameter new in Csound5

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel range.

max

max — Produces a signal that is the maximum of any number of input signals.

Description

The *max* opcode takes any number of a-rate or k-rate signals as input (all of the same rate), and outputs a signal at the same rate that is the maximum of all of the inputs. For a-rate signals, the inputs are compared one sample at a time (i.e. *max* does not scan an entire ksmps period of a signal for its local maximum as the *max_k* opcode does).

Syntax

```
amax max ain1 [, ain2] [, ain3] [, ain4] [...]
```

```
kmax max kin1 [, kin2] [, kin3] [, kin4] [...]
```

Performance

ain1, ain2, ... -- a-rate signals to be compared.

kin1, kin2, ... -- k-rate signals to be compared.

See Also

min, maxabs, minabs, maxaccum, minaccum, maxabsaccum, minabsaccum, max_k

Credits

Author: Anthony Kozar
March 2006

New in Csound version 5.01

maxabs

maxabs — Produces a signal that is the maximum of the absolute values of any number of input signals.

Description

The *maxabs* opcode takes any number of a-rate or k-rate signals as input (all of the same rate), and outputs a signal at the same rate that is the maximum of all of the inputs. It is identical to the *max* opcode except that it takes the absolute value of each input before comparing them. Therefore, the output is always non-negative. For a-rate signals, the inputs are compared one sample at a time (i.e. *maxabs* does not scan an entire ksmps period of a signal for its local maximum as the *max_k* opcode does).

Syntax

```
amax maxabs ain1 [, ain2] [, ain3] [, ain4] [...]
```

```
kmax maxabs kin1 [, kin2] [, kin3] [, kin4] [...]
```

Performance

ain1, ain2, ... -- a-rate signals to be compared.

kin1, kin2, ... -- k-rate signals to be compared.

See Also

minabs, max, min, maxaccum, minaccum, maxabsaccum, minabsaccum, max_k

Credits

Author: Anthony Kozar
March 2006

New in Csound version 5.01

maxabsaccum

maxabsaccum — Accumulates the maximum of the absolute values of audio signals.

Description

maxabsaccum compares two audio-rate variables and stores the maximum of their absolute values into the first.

Syntax

```
maxabsaccum aAccumulator, aInput
```

Performance

aAccumulator -- audio variable to store the maximum value

aInput -- signal that *aAccumulator* is compared to

The *maxabsaccum* opcode is designed to accumulate the maximum value from among many audio signals that may be in different note instances, different channels, or otherwise cannot all be compared at once using the *maxabs* opcode. *maxabsaccum* is identical to *maxaccum* except that it takes the absolute value of *aInput* before the comparison. Its semantics are similar to *vincr* since *aAccumulator* is used as both an input and an output variable, except that *maxabsaccum* keeps the maximum absolute value instead of adding the signals together. *maxabsaccum* performs the following operation on each pair of samples:

$$\text{if } (\text{abs}(\text{aInput}) > \text{aAccumulator}) \text{ aAccumulator} = \text{abs}(\text{aInput})$$

aAccumulator will usually be a global audio variable. At the end of any given computation cycle (k-period), after its value is read and used in some way, the accumulator variable should usually be reset to zero (perhaps by using the *clear* opcode). Clearing to zero is sufficient for *maxabsaccum*, unlike the *maxaccum* opcode.

See Also

minabsaccum, *maxaccum*, *minaccum*, *max*, *min*, *maxabs*, *minabs*, *vincr*, *clear*

Credits

Author: Anthony Kozar
March 2006

New in Csound version 5.01

maxaccum

maxaccum — Accumulates the maximum value of audio signals.

Description

maxaccum compares two audio-rate variables and stores the maximum value between them into the first.

Syntax

```
maxaccum aAccumulator, aInput
```

Performance

aAccumulator -- audio variable to store the maximum value

aInput -- signal that *aAccumulator* is compared to

The *maxaccum* opcode is designed to accumulate the maximum value from among many audio signals that may be in different note instances, different channels, or otherwise cannot all be compared at once using the *max* opcode. Its semantics are similar to *vincr* since *aAccumulator* is used as both an input and an output variable, except that *maxaccum* keeps the maximum value instead of adding the signals together. *maxaccum* performs the following operation on each pair of samples:

```
if (aInput > aAccumulator) aAccumulator = aInput
```

aAccumulator will usually be a global audio variable. At the end of any given computation cycle (k-period), after its value is read and used in some way, the accumulator variable should usually be reset to zero (perhaps by using the *clear* opcode). Care must be taken however if *aInput* is negative at any point, in which case the accumulator should be initialized and reset to some large enough negative value that will always be less than the input signals to which it is compared.

See Also

minaccum, *maxabsaccum*, *minabsaccum*, *max*, *min*, *maxabs*, *minabs*, *vincr*, *clear*

Credits

Author: Anthony Kozar
March 2006

New in Csound version 5.01

maxalloc

maxalloc — Limits the number of allocations of an instrument.

Description

Limits the number of allocations of an instrument.

Syntax

```
maxalloc insnum, icount
```

Initialization

insnum -- instrument number

icount -- number of instrument allocations

Performance

All instances of *maxalloc* must be defined in the header section, not in the instrument body.

Examples

Here is an example of the maxalloc opcode. It uses the file *maxalloc.csd* [examples/maxalloc.csd].

Example 287. Example of the maxalloc opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o maxalloc.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Limit Instrument #1 to three instances.
maxalloc 1, 3

; Instrument #1
instr 1
; Generate a waveform, get the cycles per second from the 4th p-field.
a1 oscil 6500, p4, 1
out a1
endin

</CsInstruments>
<CsScore>

; Just generate a nice, ordinary sine wave.
f 1 0 32768 10 1
```

```
; Play five instances of Instrument #1 for one second.  
; Note that 4th p-field contains cycles per second.  
i 1 0 1 220  
i 1 0 1 440  
i 1 0 1 880  
i 1 0 1 1320  
i 1 0 1 1760  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

Its output should contain a message like this:

```
WARNING: cannot allocate last note because it exceeds instr maxalloc
```

See Also

cpuprc, *prealloc*

Credits

Author: Gabriel Maldonado
Italy
July 1999

Example written by Kevin Conder.

New in Csound version 3.57

max_k

max_k — Local maximum (or minimum) value of an incoming asig signal

Description

max_k outputs the local maximum (or minimum) value of the incoming *asig* signal, checked in the time interval between *ktrig* has become true twice.

Syntax

```
knumkout max_k asig, ktrig, itype
```

Initialization

itype - itype determinates the behaviour of *max_k* (see below)

Performance

asig - incoming (input) signal

ktrig - trigger signal

max_k outputs the local maximum (or minimum) value of the incoming *asig* signal, checked in the time interval between *ktrig* has become true twice. *itype* determinates the behaviour of *max_k*:

- 1 - absolute maximum (sign of negative values is changed to positive before evaluation)
- 2 - actual maximum
- 3 - actual minimum
- 4 - calculate average value of *asig* in the time interval

This opcode can be useful in several situations, for example to implement a vu-meter.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

mclock

mclock — Sends a MIDI CLOCK message.

Description

Sends a MIDI CLOCK message.

Syntax

```
mclock ifreq
```

Initialization

ifreq -- clock message frequency rate in Hz

Performance

Sends a MIDI CLOCK message (0xF8) every $1/ifreq$ seconds. So *ifreq* is the frequency rate of CLOCK message in Hz.

See Also

mrtmsg

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

mdelay

mdelay — A MIDI delay opcode.

Description

A MIDI delay opcode.

Syntax

```
mdelay kstatus, kchan, kd1, kd2, kdelay
```

Performance

kstatus -- status byte of MIDI message to be delayed

kchan -- MIDI channel (1-16)

kd1 -- first MIDI data byte

kd2 -- second MIDI data byte

kdelay -- delay time in seconds

Each time that *kstatus* is other than zero, *mdelay* outputs a MIDI message to the MIDI out port after *kdelay* seconds. This opcode is useful in implementing MIDI delays. Several instances of *mdelay* can be present in the same instrument with different argument values, so complex and colorful MIDI echoes can be implemented. Further, the delay time can be changed at k-rate.

Examples

Here is an example of the mdelay opcode. It uses the file *mdelay.csd* [examples/mdelay.csd].

Example 288. Example of the mdelay opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d           -M0  -Q1;;;RT audio I/O with MIDI in
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Example by Giorgio Zucco 2007

instr 1 ;Triggered by MIDI notes on channel 1

  kstatus init 0
  ifund notnum
  ivel    veloc

  noteondur 1, ifund, ivel, 1

  kstatus = kstatus + 1
```

```
idel1 = .2
idel2 = .4
idel3 = .6
idel4 = .8

;make four delay lines

mdelay      kstatus,1,ifund+2, ivel,idel1
mdelay      kstatus,1,ifund+4, ivel,idel2
mdelay      kstatus,1,ifund+6, ivel,idel3
mdelay      kstatus,1,ifund+8, ivel,idel4

endin

</CsInstruments>
<CsScore>
; Dummy ftable
f 0 60
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Gabriel Maldonado

Italy

November 1998

New in Csound version 3.492

metro

metro — Trigger Metronome

Description

Generate a metronomic signal to be used in any circumstance an isochronous trigger is needed.

Syntax

```
ktrig metro kfreq [, initphase]
```

Initialization

initphase - initial phase value (in the 0 to 1 range)

Performance

ktrig - output trigger signal

kfreq - frequency of trigger bangs in cps

metro is a simple opcode that outputs a sequence of isochronous bangs (that is 1 values) each 1/kfreq seconds. Trigger signals can be used in any circumstance, mainly to temporize realtime algorithmic compositional structures.

Examples

Here is an example of the metro opcode. It uses the file *metro.csd* [examples/metro.csd]

Example 289. Example of the metro opcode.

```
<CsoundSynthesizer>
<CsOptions>
-odac -B441 -b441
</CsOptions>
<CsInstruments>

sr      =      44100
kr      =      100
kmps    =      441
nchnls  =      2

      instr      1
ktrig metro 0.2
printk2 ktrig
      endin

</CsInstruments>
<CsScore>
i 1 0 20

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado.

Example written by Andrés Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

midic14

midic14 — Allows a floating-point 14-bit MIDI signal scaled with a minimum and a maximum range.

Description

Allows a floating-point 14-bit MIDI signal scaled with a minimum and a maximum range.

Syntax

```
idest midic14 ictrlno1, ictrlno2, imin, imax [, ifn]
```

```
kdest midic14 ictrlno1, ictrlno2, kmin, kmax [, ifn]
```

Initialization

idest -- output signal

ictrlno -- most-significant byte controller number (0-127)

ictrlno2 -- least-significant byte controller number (0-127)

imin -- user-defined minimum floating-point value of output

imax -- user-defined maximum floating-point value of output

ifn (optional) -- table to be read when indexing is required. Table must be normalized. Output is scaled according to *imin* and *imax* values.

Performance

kdest -- output signal

kmin -- user-defined minimum floating-point value of output

kmax -- user-defined maximum floating-point value of output

midic14 (i- and k-rate 14 bit MIDI control) allows a floating-point 14-bit MIDI signal scaled with a minimum and a maximum range. The minimum and maximum values can be varied at k-rate. It can use optional interpolated table indexing. It requires two MIDI controllers as input.



Note

Please note that the *midic* family of opcodes are designed for MIDI triggered events, and don't require a channel number since they will respond to the same channel as the one that triggered the instrument (see *massign*). However they will crash if called from a score driven event.

See Also

ctrl7, *ctrl14*, *ctrl21*, *initc7*, *initc14*, *initc21*, *midic7*, *midic21*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct controller number range.

midic21

midic21 — Allows a floating-point 21-bit MIDI signal scaled with a minimum and a maximum range.

Description

Allows a floating-point 21-bit MIDI signal scaled with a minimum and a maximum range.

Syntax

```
idest midic21 ictlno1, ictlno2, ictlno3, imin, imax [, ifn]
```

```
kdest midic21 ictlno1, ictlno2, ictlno3, kmin, kmax [, ifn]
```

Initialization

idest -- output signal

ictlno1 -- most-significant byte controller number (0-127)

ictlno2 -- mid-significant byte controller number (0-127)

ictlno3 -- least-significant byte controller number (0-127)

imin -- user-defined minimum floating-point value of output

imax -- user-defined maximum floating-point value of output

ifn (optional) -- table to be read when indexing is required. Table must be normalized. Output is scaled according to the *imin* and *imax* values.

Performance

kdest -- output signal

kmin -- user-defined minimum floating-point value of output

kmax -- user-defined maximum floating-point value of output

midic21 (i- and k-rate 21 bit MIDI control) allows a floating-point 21-bit MIDI signal scaled with a minimum and a maximum range. Minimum and maximum values can be varied at k-rate. It can use optional interpolated table indexing. It requires three MIDI controllers as input.



Note

Please note that the *midic* family of opcodes are designed for MIDI triggered events, and don't require a channel number since they will respond to the same channel as the one that triggered the instrument (see *massign*). However they will crash if called from a score driven event.

See Also

ctrl7, ctrl14, ctrl21, initc7, initc14, initc21, midic7, midic14

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct controller number range.

midic7

midic7 — Allows a floating-point 7-bit MIDI signal scaled with a minimum and a maximum range.

Description

Allows a floating-point 7-bit MIDI signal scaled with a minimum and a maximum range.

Syntax

```
idest midic7 ictlno, imin, imax [, ifn]
```

```
kdest midic7 ictlno, kmin, kmax [, ifn]
```

Initialization

idest -- output signal

ictlno -- MIDI controller number (0-127)

imin -- user-defined minimum floating-point value of output

imax -- user-defined maximum floating-point value of output

ifn (optional) -- table to be read when indexing is required. Table must be normalized. Output is scaled according to the *imin* and *imax* values.

Performance

kdest -- output signal

kmin -- user-defined minimum floating-point value of output

kmax -- user-defined maximum floating-point value of output

midic7 (i- and k-rate 7 bit MIDI control) allows a floating-point 7-bit MIDI signal scaled with a minimum and a maximum range. It also allows optional non-interpolated table indexing. In *midic7* minimum and maximum values can be varied at k-rate.



Note

Please note that the *midic* family of opcodes are designed for MIDI triggered events, and don't require a channel number since they will respond to the same channel as the one that triggered the instrument (see *massign*). However they will crash if called from a score driven event.

See Also

ctrl7, *ctrl14*, *ctrl21*, *initc7*, *initc14*, *initc21*, *midic14*, *midic21*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct controller number range.

midichannelaftertouch

midichannelaftertouch — Gets a MIDI channel's aftertouch value.

Description

midichannelaftertouch is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midichannelaftertouch xchannelaftertouch [ , ilow] [ , ihigh]
```

Initialization

ilow (optional) -- optional low value after rescaling, defaults to 0.

ihigh (optional) -- optional high value after rescaling, defaults to 127.

Performance

xchannelaftertouch -- returns the MIDI channel aftertouch during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the value of *xchannelaftertouch* with the corresponding value from MIDI input. If the instrument was *NOT* activated by MIDI input, the value of *xchannelaftertouch* remains unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

Examples

Here is an example of the *midichannelaftertouch* opcode. It uses the file *midichannelaftertouch.csd* [examples/midichannelaftertouch.csd].

Example 290. Example of the midichannelaftertouch opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using com-

mand line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d          -M0    ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o midichannelaftertouch.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kaft init 0
  midichannelaftertouch kaft

  ; Display the aftertouch value when it changes.
  printk2 kaft
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
i1 127.00000
i1 20.00000
i1 44.00000
```

See Also

midicontrolchange, mididefault, midinoteoff, midinoteoncps, midinoteonkey, midinoteonoct, midinoteonpch, midipitchbend, midipolyaftertouch, midiprogramchange

Credits

Author: Michael Gogins

Example written by Kevin Conder.

New in version 4.20

midichn

midichn — Returns the MIDI channel number from which the note was activated.

Description

midichn returns the MIDI channel number (1 - 16) from which the note was activated. In the case of score notes, it returns 0.

Syntax

ichn **midichn**

Initialization

ichn -- channel number. If the current note was activated from score, it is set to zero.

Examples

Here is a simple example of the *midichn* opcode. It uses the file *midichn.csd* [examples/midichn.csd].

Example 291. Example of the *midichn* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime output leave only the line below:
; -o midichn.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  il midichn

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 12 seconds.
i 1 0 12
e

</CsScore>
</CsoundSynthesizer>
```

Here is an advanced example of the *midichn* opcode. It uses the file *midichn_advanced.csd* [ex-

amples/midichn_advanced.csd].

Don't forget that you must include the *-F flag* when using an external MIDI file like “midichn_advanced.mid”.

Example 292. An advanced example of the midichn opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d          -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o midichn_advanced.wav -W   ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 10
nchnls = 1

        massign 1, 1           ; all channels use instr 1
        massign 2, 1
        massign 3, 1
        massign 4, 1
        massign 5, 1
        massign 6, 1
        massign 7, 1
        massign 8, 1
        massign 9, 1
        massign 10, 1
        massign 11, 1
        massign 12, 1
        massign 13, 1
        massign 14, 1
        massign 15, 1
        massign 16, 1

gicnt = 0           ; note counter

        instr 1

gicnt = gicnt + 1 ; update note counter
kcnt init gicnt ; copy to local variable
ichn midichn           ; get channel number
istime times           ; note-on time

        if (ichn > 0.5) goto 12           ; MIDI note
printks "note %.0f (time = %.2f) was activated from the score\\n", \
        3600, kcnt, istime
        goto 11

12:
        printks "note %.0f (time = %.2f) was activated from channel %.0f\\n", \
        3600, kcnt, istime, ichn
11:
        endin

</CsInstruments>
<CsScore>

t 0 60
f 0 6 2 -2 0
i 1 1 0.5
i 1 4 0.5
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
note 7 (time = 0.00) was activated from channel 4
note 8 (time = 0.00) was activated from channel 2
```


See Also

pgmassign

Credits

Author: Istvan Varga
May 2002

The simple example was written by Kevin Conder.

New in version 4.20

midicontrolchange

midicontrolchange — Gets a MIDI control change value.

Description

midicontrolchange is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midicontrolchange xcontroller, xcontrollervalue [, ilow] [, ihigh]
```

Initialization

ilow (optional) -- optional low value after rescaling, defaults to 0.

ihigh (optional) -- optional high value after rescaling, defaults to 127.

Performance

xcontroller -- specifies a MIDI controller number (0-127).

xcontrollervalue -- returns the value of the MIDI controller during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the values of the *xcontroller* and *xcontrollervalue* with the corresponding values from MIDI input. If the instrument was *NOT* activated by MIDI input, the values of *xcontroller* and *xcontrollervalue* remain unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

See Also

midichannelaftertouch, *mididefault*, *midinoteoff*, *midinoteoncps*, *midinoteonkey*, *midinoteonoct*, *midinoteonpch*, *midipitchbend*, *midipolyaftertouch*, *midiprogramchange*

Credits

Author: Michael Gogins

New in version 4.20

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

midictrl

midictrl — Get the current value (0-127) of a specified MIDI controller.

Description

Get the current value (0-127) of a specified MIDI controller.

Syntax

```
ival midictrl inum [ , imin] [ , imax]
```

```
kval midictrl inum [ , imin] [ , imax]
```

Initialization

inum -- MIDI controller number (0-127)

imin, *imax* -- set minimum and maximum limits on values obtained.

Performance

Get the current value (0-127) of a specified MIDI controller.

Warning

midictrl should only be used in notes that were triggered from MIDI, so that an associated channel number is available. For notes activated from the score, line events, or orchestra, the *ctrl7* opcode that takes an explicit channel number should be used instead.

See Also

aftouch, *ampmidi*, *cpismidi*, *cpismidib*, *notnum*, *octmidi*, *octmidib*, *pchbend*, *pchmidi*, *pchmidib*, *veloc*

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Thanks goes to Rasmus Ekman for pointing out the correct controller number range.

mididefault

mididefault — Changes values, depending on MIDI activation.

Description

mididefault is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
mididefault xdefault, xvalue
```

Performance

xdefault -- specifies a default value that will be used during MIDI activation.

xvalue -- overwritten by *xdefault* during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode will overwrite the value of *xvalue* with the value of *xdefault*. If the instrument was *NOT* activated by MIDI input, *xvalue* will remain unchanged.

This enables score pfields to receive a default value during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

See Also

midichannelaftertouch, *midicontrolchange*, *midinoteoff*, *midinoteoncps*, *midinoteonkey*, *midinoteonoct*, *midinoteonpch*, *midipitchbend*, *midipolyaftertouch*, *midiprogramchange*

Credits

Author: Michael Gogins

New in version 4.20

midin

`midin` — Returns a generic MIDI message received by the MIDI IN port.

Description

Returns a generic MIDI message received by the MIDI IN port

Syntax

```
kstatus, kchan, kdata1, kdata2 midin
```

Performance

kstatus -- the type of MIDI message. Can be:

- 128 (note off)
- 144 (note on)
- 160 (polyphonic aftertouch)
- 176 (control change)
- 192 (program change)
- 208 (channel aftertouch)
- 224 (pitch bend)
- 0 if no MIDI message are pending in the MIDI IN buffer

kchan -- MIDI channel (1-16)

kdata1, kdata2 -- message-dependent data values

midin has no input arguments, because it reads at the MIDI in port implicitly. It works at k-rate. Normally (i.e., when no messages are pending) *kstatus* is zero, only when MIDI data are present in the MIDI IN buffer, is *kstatus* set to the type of the relevant messages.



Note

Be careful when using *midin* in low numbered instruments, since a MIDI note will launch additional instances of the instrument, resulting in duplicate events and weird behaviour. Use *massign* to direct MIDI note on messages to a different instrument or to disable triggering of instruments from MIDI.

Examples

Here is an example of the *midin* opcode. It uses the file *midin.csd* [examples/midin.csd].

Example 293. Example of the midin opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      -M0  --rtmidi=virtual  ;;RT audio I/O with MIDI in
; For Non-realtime output leave only the line below:
; -o midiin.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr          = 44100
ksmps       = 10
nchnls      = 1

; Example by schwaahed 2006

      massign          0, 130 ; make sure that all channels
      pgmassign        0, 130 ; and programs are assigned to test instr

instr       130

knotelength   init      0
knoteontime   init      0

kstatus, kchan, kdata1, kdata2          midiin

if (kstatus == 128) then
knoteofftime   times
knotelength    = knoteofftime - knoteontime
printks "kstatus= %d, kchan = %d, \\tnote# = %d, velocity = %d \\tNote OFF\\t%f %f\\n", 0, kstatus,
knoteontime    times
printks "kstatus= %d, kchan = %d, \\tnote# = %d, velocity = %d \\tNote ON\\t%f\\n", 0, kstatus, kchan,

elseif (kstatus == 144) then
printks "kstatus= %d, kchan = %d, \\tkdata1 = %d, kdata2 = %d \\tPolyphonic Aftertouch\\n", 0, ksta

elseif (kstatus == 160) then
printks "kstatus= %d, kchan = %d, \\t CC = %d, value = %d \\tControl Change\\n", 0, kstatus, kchan,

elseif (kstatus == 176) then
printks "kstatus= %d, kchan = %d, \\tkdata1 = %d, kdata2 = %d \\tProgram Change\\n", 0, kstatus, kchan,

elseif (kstatus == 192) then
printks "kstatus= %d, kchan = %d, \\tkdata1 = %d, kdata2 = %d \\tChannel Aftertouch\\n", 0, kstatu

elseif (kstatus == 208) then
printks "kstatus= %d, kchan = %d, \\tkdata1 = %d, kdata2 = %d \\tPitch Bend\\n", 0, kstatus,

elseif (kstatus == 224) then
printks "kstatus= %d, kchan = %d, \\t ( data1 , kdata2 ) = ( %d, %d )\\tPitch Bend\\n", 0, kstatus,

endif

endin

</CsInstruments>
<CsScore>
i130 0 3600
e
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Gabriel Maldonado
Italy
1998

New in Csound version 3.492

midinoteoff

midinoteoff — Gets a MIDI noteoff value.

Description

midinoteoff is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midinoteoff xkey, xvelocity
```

Performance

xkey -- returns MIDI key during MIDI activation, remains unchanged otherwise.

xvelocity -- returns MIDI velocity during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the values of the *xkey* and *xvelocity* with the corresponding values from MIDI input. If the instrument was *NOT* activated by MIDI input, the values of *xkey* and *xvelocity* remain unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

Examples

Here is an example of the midinoteoff opcode. It uses the file *midinoteoff.csd* [examples/midinoteoff.csd].

Example 294. Example of the midinoteoff opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
```



```
-odac          -iadc      -d          -M0    ;;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o midinoteoff.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kkey init 0
  kvelocity init 0

  midinoteoff kkey, kvelocity

  ; Display the key value when it changes.
  printk2 kkey
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
i1      60.00000
i1      76.00000
```

See Also

midichannelaftertouch, midicontrolchange, mididefault, midinoteoncps, midinoteonkey, midinoteonoct, midinoteonpch, midipitchbend, midipolyaftertouch, midiprogramchange

Credits

Author: Michael Gogins

Example written by Kevin Conder.

New in version 4.20

midinoteoncps

midinoteoncps — Gets a MIDI note number as a cycles-per-second frequency.

Description

midinoteoncps is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midinoteoncps xcps, xvelocity
```

Performance

xcps -- returns MIDI key translated to cycles per second during MIDI activation, remains unchanged otherwise.

xvelocity -- returns MIDI velocity during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the values of *xcps* and *xvelocity* with the corresponding values from MIDI input. If the instrument was *NOT* activated by MIDI input, the values of *xcps* and *xvelocity* remain unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

Examples

Here is an example of the midinoteoncps opcode. It uses the file *midinoteoncps.csd* [examples/midinoteoncps.csd].

Example 295. Example of the midinoteoncps opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
```

```
; Audio out   Audio in   No messages  MIDI in
-odac         -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o midinoteoncps.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kcps init 0
  kvelocity init 0

  midinoteoncps kcps, kvelocity

  ; Display the cycles-per-second value when it changes.
  printk2 kcps
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
i1 261.62561
i1 440.00006
```

See Also

midichannelaftertouch, midicontrolchange, mididefault, midinoteoff, midinoteonkey, midinoteonoct, midinoteonpch, midipitchbend, midipolyaftertouch, midiprogramchange

Credits

Author: Michael Gogins

Example written by Kevin Conder.

New in version 4.20

midinoteonkey

midinoteonkey — Gets a MIDI note number value.

Description

midinoteonkey is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midinoteonkey xkey, xvelocity
```

Performance

xkey -- returns MIDI key during MIDI activation, remains unchanged otherwise.

xvelocity -- returns MIDI velocity during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the values of *xkey* and *xvelocity* with the corresponding values from MIDI input. If the instrument was *NOT* activated by MIDI input, the values of *xkey* and *xvelocity* remain unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

Examples

Here is an example of the midinoteonkey opcode. It uses the file *midinoteonkey.csd* [examples/midinoteonkey.csd].

Example 296. Example of the midinoteonkey opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
```

```
-odac          -iadc      -d          -M0    ;;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o midinoteonkey.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kkey init 0
  kvelocity init 0

  midinoteonkey kkey, kvelocity

  ; Display the key value when it changes.
  printk2 kkey
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
i1      60.00000
i1      69.00000
```

See Also

midichannelaftertouch, midicontrolchange, mididefault, midinoteoff, midinoteoncps, midinoteonoct, midinoteonpch, midipitchbend, midipolyaftertouch, midiprogramchange

Credits

Author: Michael Gogins

Example written by Kevin Conder.

New in version 4.20

midinoteonoct

midinoteonoct — Gets a MIDI note number value as octave-point-decimal value.

Description

midinoteonoct is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midinoteonoct xoct, xvelocity
```

Performance

xoct -- returns MIDI key translated to linear octaves during MIDI activation, remains unchanged otherwise.

xvelocity -- returns MIDI velocity during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the values of *xoct* and *xvelocity* with the corresponding value from MIDI input. If the instrument was *NOT* activated by MIDI input, the values of *xoct* and *xvelocity* remain unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

Examples

Here is an example of the midinoteonoct opcode. It uses the file *midinoteonoct.csd* [examples/midinoteonoct.csd].

Example 297. Example of the midinoteonoct opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
```

```
; Audio out   Audio in   No messages  MIDI in
-odac         -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o midinoteonoct.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  koct init 0
  kvelocity init 0

  midinoteonoct koct, kvelocity

  ; Display the octave-point-decimal value when it changes.
  printk2 koct
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
i1      8.00000
i1      9.33333
```

See Also

midichannelaftertouch, midicontrolchange, mididefault, midinoteoff, midinoteoncps, midinoteonkey, midinoteonpch, midipitchbend, midipolyaftertouch, midiprogramchange

Credits

Author: Michael Gogins

Example written by Kevin Conder.

New in version 4.20

midinoteonpch

midinoteonpch — Gets a MIDI note number as a pitch-class value.

Description

midinoteonpch is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midinoteonpch xpch, xvelocity
```

Performance

xpch -- returns MIDI key translated to octave.pch during MIDI activation, remains unchanged otherwise.

xvelocity -- returns MIDI velocity during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the values of *xpch* and *xvelocity* with the corresponding value from MIDI input. If the instrument was *NOT* activated by MIDI input, the values of *xpch* and *xvelocity* remain unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

Examples

Here is an example of the midinoteonpch opcode. It uses the file *midinoteonpch.csd* [examples/midinoteonpch.csd].

Example 298. Example of the midinoteonpch opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
```



```
; Audio out   Audio in   No messages  MIDI in
-odac         -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o midinoteonpch.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kpch init 0
  kvelocity init 0

  midinoteonpch kpch, kvelocity

  ; Display the pitch-class value when it changes.
  printk2 kpch
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
i1      8.09000
i1      9.05000
```

See Also

midichannelaftertouch, midicontrolchange, mididefault, midinoteoff, midinoteoncps, midinoteonkey, midinoteonoct, midipitchbend, midipolyaftertouch, midiprogramchange

Credits

Author: Michael Gogins

Example written by Kevin Conder.

New in version 4.20

midion

midion — Generates MIDI note messages at k-rate.

Description

Generates MIDI note messages at k-rate.

Syntax

```
midion kchn, knum, kvel
```

Performance

kchn -- MIDI channel number (1-16)

knun -- note number (0-127)

kvel -- velocity (0-127)

midion (k-rate note on) plays MIDI notes with current *kchn*, *knun* and *kvel*. These arguments can be varied at k-rate. Each time the MIDI converted value of any of these arguments changes, last MIDI note played by current instance of *midion* is immediately turned off and a new note with the new argument values is activated. This opcode, as well as *moscil*, can generate very complex melodic textures if controlled by complex k-rate signals.

Any number of *midion* opcodes can appear in the same Csound instrument, allowing a counterpoint-style polyphony within a single instrument.

Examples

Here is a simple example of the *midion* opcode. It uses the file *midion_simple.csd* [examples/midion_simple.csd].

Example 299. Simple Example of the midion opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

This example generates a minor chord over every note received on the MIDI input. It generates MIDI notes on csound's MIDI output, so be sure to connect something.

```
<CsoundSynthesizer>
<CsOptions>

; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadac    -d          -M0  -Q1  ;;RT audio I/O with MIDI in
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Example by Giorgio Zucco 2007

instr 1 ;Triggered by MIDI notes on channel 1
```

```
ifund notnum
ivel veloc

knote1 init ifund
knote2 init ifund + 3
knote3 init ifund + 5

;minor chord on MIDI out channel 1
;Needs something plugged to csound's MIDI output
midion 1, knote1,ivel
midion 1, knote2,ivel
midion 1, knote3,ivel

endin

</CsInstruments>
<CsScore>
; Dummy ftable
f0 60
</CsScore>
</CsoundSynthesizer>
```

Here is another example of the midion opcode. It uses the file *midion_scale.csd* [examples/midion_scale.csd].

Example 300. Example of the midion opcode to generate random notes from a scale.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

This example generates random notes from a given scale for every note received on the MIDI input. It generates MIDI notes on csound's MIDI output, so be sure to connect something.

```
<CsoundSynthesizer>
<CsOptions>

; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadac     -d         -M0 -Q1 ;;RT audio I/O with MIDI in
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Example by Giorgio Zucco 2007

instr 1 ; Triggered by MIDI notes on channel 1

ivel          veloc

krate = 8
iscale = 100 ;f

; Random sequence from table f100
krnd randh int(14),krate,-1
knote table abs(krnd),iscale
; Generates random notes from the scale on ftable 100
; on channel 1 of csound's MIDI output
midion 1,knote,ivel

endin

</CsInstruments>
<CsScore>
f100 0 32 -2 40 50 60 70 80 44 54 65 74 84 39 49 69 69

; Dummy ftable
f0 60
</CsScore>
</CsoundSynthesizer>
```

See Also

moscil, midion2, noteon, noteoff, noteondur, noteondur2

Credits

Author: Gabriel Maldonado
Italy
May 1997

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

midion2

midion2 — Sends noteon and noteoff messages to the MIDI OUT port.

Description

Sends noteon and noteoff messages to the MIDI OUT port when triggered by a value different than zero.

Syntax

```
midion2 kchn, knum, kvel, ktrig
```

Performance

kchn -- MIDI channel (1-16)

knum -- MIDI note number (0-127)

kvel -- note velocity (0-127)

ktrig -- trigger input signal (normally 0)

Similar to *midion*, this opcode sends noteon and noteoff messages to the MIDI out port, but only when *ktrig* is non-zero. This opcode is can work together with the output of the *trigger* opcode.

See Also

moscil, *midion*, *noteon*, *noteoff*, *noteondur*, *noteondur2*

Credits

Author: Gabriel Maldonado
Italy
1998

New in Csound version 3.492

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

midout

midout — Sends a generic MIDI message to the MIDI OUT port.

Description

Sends a generic MIDI message to the MIDI OUT port.

Syntax

```
midout kstatus, kchan, kdata1, kdata2
```

Performance

kstatus -- the type of MIDI message. Can be:

- 128 (note off)
- 144 (note on)
- 160 (polyphonic aftertouch)
- 176 (control change)
- 192 (program change)
- 208 (channel aftertouch)
- 224 (pitch bend)
- 0 when no MIDI messages must be sent to the MIDI OUT port

kchan -- MIDI channel (1-16)

kdata1, *kdata2* -- message-dependent data values

midout has no output arguments, because it sends a message to the MIDI OUT port implicitly. It works at k-rate. It sends a MIDI message only when *kstatus* is non-zero.



Warning

Warning: Normally *kstatus* should be set to 0. Only when the user intends to send a MIDI message, can it be set to the corresponding message type number.

Credits

Author: Gabriel Maldonado
Italy
1998

New in Csound version 3.492

midipitchbend

midipitchbend — Gets a MIDI pitchbend value.

Description

midipitchbend is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midipitchbend xpitchbend [, ilow] [, ihigh]
```

Initialization

ilow (optional) -- optional low value after rescaling, defaults to 0.

ihigh (optional) -- optional high value after rescaling, defaults to 127.

Performance

xpitchbend -- returns the MIDI pitch bend during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the value of *xpitchbend* with the corresponding value from MIDI input. If the instrument was *NOT* activated by MIDI input, the value of *xpitchbend* remains unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

Examples

Here is an example of the midipitchbend opcode. It uses the file *midipitchbend.csd* [examples/midipitchbend.csd].

Example 301. Example of the midipitchbend opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d          -M0    ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o midipitchbend.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kpb init 0

  midipitchbend kpb

  ; Display the pitch-bend value when it changes.
  printk2 kpb
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
i1      0.12695
i1      0.00000
i1      -0.01562
```

See Also

midichannelaftertouch, midicontrolchange, mididefault, midinoteoff, midinoteoncps, midinoteonkey, midinoteonoct, midinoteonpch, midipolyaftertouch, midiprogramchange

Credits

Author: Michael Gogins

Example written by Kevin Conder.

New in version 4.20

midipolyaftertouch

midipolyaftertouch — Gets a MIDI polyphonic aftertouch value.

Description

midipolyaftertouch is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midipolyaftertouch xpolyaftertouch, xcontrollervalue [, ilow] [, ihigh]
```

Initialization

ilow (optional) -- optional low value after rescaling, defaults to 0.

ihigh (optional) -- optional high value after rescaling, defaults to 127.

Performance

xpolyaftertouch -- returns MIDI polyphonic aftertouch during MIDI activation, remains unchanged otherwise.

xcontrollervalue -- returns the value of the MIDI controller during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the values of *xpolyaftertouch* and *xcontrollervalue* with the corresponding values from MIDI input. If the instrument was *NOT* activated by MIDI input, the values of *xpolyaftertouch* and *xcontrollervalue* remain unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

See Also

midichannelaftertouch, *midicontrolchange*, *mididefault*, *midinoteoff*, *midinoteoncps*, *midinoteonkey*, *midinoteonoct*, *midinoteonpch*, *midipitchbend*, *midiprogramchange*

Credits

Author: Michael Gogins

New in version 4.20

midiprogramchange

midiprogramchange — Gets a MIDI program change value.

Description

midiprogramchange is designed to simplify writing instruments that can be used interchangeably for either score or MIDI input, and to make it easier to adapt instruments originally written for score input to work with MIDI input.

In general, it should be possible to write instrument definitions that work identically with both scores and MIDI, including both MIDI files and real-time MIDI input, without using any conditional statements, and that take full advantage of MIDI voice messages.

Note that correlating Csound instruments with MIDI channel numbers is done using the *massign* opcode for real-time performance,. For file-driven performance, instrument numbers default to MIDI channel number + 1, but the defaults are overridden by any MIDI program change messages in the file.

Syntax

```
midiprogramchange xprogram
```

Performance

xprogram -- returns the MIDI program change value during MIDI activation, remains unchanged otherwise.

If the instrument was activated by MIDI input, the opcode overwrites the value of *xprogram* with the corresponding value from MIDI input. If the instrument was *NOT* activated by MIDI input, the value of *xprogram* remains unchanged.

This enables score p-fields to receive MIDI input data during MIDI activation, and score values otherwise.



Adapting a score-activated Csound instrument.

See the *MIDI interop opcodes* section for details on adapting score driven instruments for MIDI or vice-versa.

See Also

midichannelaftertouch, *midicontrolchange*, *mididefault*, *midinoteoff*, *midinoteoncps*, *midinoteonkey*, *midinoteonoct*, *midinoteonpch*, *midipitchbend*, *midipolyaftertouch*

Credits

Author: Michael Gogins

New in version 4.20

miditempo

miditempo — Returns the current tempo at k-rate, of either the MIDI file (if available) or the score

Description

Returns the current tempo at k-rate, of either the MIDI file (if available) or the score

Syntax

```
k sig miditempo
```

Credits

Author: Istvan Varga
March 2005
New in Csound5

midremot

midremot — An opcode which can be used to implement a remote midi orchestra. This opcode will send midi events from a source machine to one destination.

Description

With the *midremot* and *midglobal* opcodes you are able to perform instruments on remote machines and control them from a master machine. The remote opcodes are implemented using the master/client model. All the machines involved contain the same orchestra but only the master machine contains the information of the midi score. During the performance the master machine sends the midi events to the clients. The *midremot* opcode will send events from a source machine to one destination if you want to send events to many destinations (broadcast) use the *midglobal* opcode instead. These two opcodes can be used in combination.

Syntax

```
midremotdestination, isource, instrnum [,instrnum...]
```

Initialization

destination -- a string that is the intended host computer (e.g. 192.168.0.100). This is the destination host which receives the events from the given instrument.

isource -- a string that is the intended host computer (e.g. 192.168.0.100). This is the source host which generates the events of the given instrument and sends it to the address given by *destination*.

instrnum -- a list of instrument numbers which will be played on the destination machine

Example

Examples

Here is an example of the *midremot* opcode. It uses the files *insremot.csd* [examples/midremot.csd].

Example 302. Example of the *insremot* opcode.

The example shows a Bach fugue played on 4 remote computers. The master machine is named "192.168.1.100", client1 "192.168.1.101" and so on. Start the clients on each machine (they will be waiting to receive the events from the master machine) and then start the master. Here is the command on linux to start a client (`csound -dm0 -odac -+rtaudio=alsa midremot.csd -+rtmidi=None`), and the command on the master machine will look like this (`csound -dm0 -odac -+rtaudio=alsa midremot.csd -F midremot.mid`).

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o midremot.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```

```
sr = 44100
kr = 441
ksmps = 100
nchnls = 2

massign 1,1
massign 2,2
massign 3,3
massign 4,4
massign 5,5

gal init 0
ga2 init 0

gil sfload "19Trumpet.sf2"
gi2 sfload "01hpschd.sf2"
gi3 sfload "07AcousticGuitar.sf2"
gi4 sfload "22Bassoon.sf2"

gitab ftgen 1,0,1024,10,1

midremot "192.168.1.100", "192.168.1.101", 1
midremot "192.168.1.100", "192.168.1.102", 2
midremot "192.168.1.100", "192.168.1.103", 3

midglobal "192.168.1.100", 5

instr 1
  sfpassign 0, gil
ifreq cpsmidi
iamp ampmidi 10
inum notnum
ivel veloc
kamp linsegr 1,1,1,.1,0
kfreq init 1
a1,a2 sfplay ivel,inum,kamp*iamp,kfreq,0,0
  outs a1,a2
vincr gal, a1*.5
vincr ga2, a2*.5
endin

instr 2
  sfpassign 0, gi2
ifreq cpsmidi
iamp ampmidi 15
inum notnum
ivel veloc
kamp linsegr 1,1,1,.1,0
kfreq init 1
a1,a2 sfplay ivel,inum,kamp*iamp,kfreq,0,0
  outs a1,a2
vincr gal, a1*.4
vincr ga2, a2*.4
endin

instr 3
  sfpassign 0, gi3
ifreq cpsmidi
iamp ampmidi 10
inum notnum
ivel veloc
kamp linsegr 1,1,1,.1,0
kfreq init 1
a1,a2 sfplay ivel,inum,kamp*iamp,kfreq,0,0
  outs a1,a2
vincr gal, a1*.5
vincr ga2, a2*.5
endin

instr 4
  sfpassign 0, gi4
ifreq cpsmidi
iamp ampmidi 15
inum notnum
ivel veloc
kamp linsegr 1,1,1,.1,0
kfreq init 1
a1,a2 sfplay ivel,inum,kamp*iamp,kfreq,0,0
  outs a1,a2
vincr gal, a1*.5
vincr ga2, a2*.5
endin

instr 5
```

```
      kamp midic7 1,0,1
      denorm ga1
      denorm ga2
aL, aR reverbsc ga1, ga2, .9, 16000, sr, 0.5
      outs aL, aR
      ga1 = 0
      ga2 = 0
endin

</CsInstruments>
<CsScore>
; Score
f0 160
</CsScore>
</CsoundSynthesizer>
```

See also

insglobal, insremot, midglobal, remoteport

Credits

Author: Simon Schampijer
2006

New in version 5.03

midglobal

midglobal — An opcode which can be used to implement a remote midi orchestra. This opcode will broadcast the midi events to all the machines involved in the remote concert.

Description

With the *midremot* and *midglobal* opcodes you are able to perform instruments on remote machines and control them from a master machine. The remote opcodes are implemented using the master/client model. All the machines involved contain the same orchestra but only the master machine contains the information of the midi score. During the performance the master machine sends the midi events to the clients. The *midglobal* opcode sends the events to all the machines involved in the remote concert. These machines are determined by the *midremot* definitions made above the *midglobal* command. To send events to only one machine use *midremot*.

Syntax

```
midglobalsource, instrnum [,instrnum...]
```

Initialization

source -- a string that is the intended host computer (e.g. 192.168.0.100). This is the source host which generates the events of the given instrument(s) and sends it to all the machines involved in the remote concert.

instrnum -- a list of instrument numbers which will be played on the destination machines

Examples

See the entry for *midremot* for an example of usage.

See also

insglobal, *insremot*, *midremot*, *remoteport*

Credits

Author: Simon Schampijer
2006

New in version 5.03

min

min — Produces a signal that is the minimum of any number of input signals.

Description

The *min* opcode takes any number of a-rate or k-rate signals as input (all of the same rate), and outputs a signal at the same rate that is the minimum of all of the inputs. For a-rate signals, the inputs are compared one sample at a time (i.e. *min* does not scan an entire ksmps period of a signal for its local minimum as the *max_k* opcode does).

Syntax

```
amin min ain1 [, ain2] [, ain3] [, ain4] [...]
```

```
kmin min kin1 [, kin2] [, kin3] [, kin4] [...]
```

Performance

ain1, ain2, ... -- a-rate signals to be compared.

kin1, kin2, ... -- k-rate signals to be compared.

See Also

max, maxabs, minabs, maxaccum, minaccum, maxabsaccum, minabsaccum, max_k

Credits

Author: Anthony Kozar
March 2006

New in Csound version 5.01

minabs

minabs — Produces a signal that is the minimum of the absolute values of any number of input signals.

Description

The *minabs* opcode takes any number of a-rate or k-rate signals as input (all of the same rate), and outputs a signal at the same rate that is the minimum of all of the inputs. It is identical to the *min* opcode except that it takes the absolute value of each input before comparing them. Therefore, the output is always non-negative. For a-rate signals, the inputs are compared one sample at a time (i.e. *minabs* does not scan an entire ksmps period of a signal for its local minimum as the *max_k* opcode does).

Syntax

```
amin minabs ain1 [, ain2] [, ain3] [, ain4] [...]
```

```
kmin minabs kin1 [, kin2] [, kin3] [, kin4] [...]
```

Performance

ain1, ain2, ... -- a-rate signals to be compared.

kin1, kin2, ... -- k-rate signals to be compared.

See Also

maxabs, max, min, maxaccum, minaccum, maxabsaccum, minabsaccum, max_k

Credits

Author: Anthony Kozar
March 2006

New in Csound version 5.01

minabsaccum

minabsaccum — Accumulates the minimum of the absolute values of audio signals.

Description

minabsaccum compares two audio-rate variables and stores the minimum of their absolute values into the first.

Syntax

```
minabsaccum aAccumulator, aInput
```

Performance

aAccumulator -- audio variable to store the minimum value

aInput -- signal that *aAccumulator* is compared to

The *minabsaccum* opcode is designed to accumulate the minimum value from among many audio signals that may be in different note instances, different channels, or otherwise cannot all be compared at once using the *minabs* opcode. *minabsaccum* is identical to *minaccum* except that it takes the absolute value of *aInput* before the comparison. Its semantics are similar to *vincr* since *aAccumulator* is used as both an input and an output variable, except that *minabsaccum* keeps the minimum absolute value instead of adding the signals together. *minabsaccum* performs the following operation on each pair of samples:

$$\text{if } (\text{abs}(\text{aInput}) < \text{aAccumulator}) \text{ aAccumulator} = \text{abs}(\text{aInput})$$

aAccumulator will usually be a global audio variable. At the end of any given computation cycle (k-period), after its value is read and used in some way, the accumulator variable should usually be reset to some large enough positive value that will always be greater than the input signals to which it is compared.

See Also

maxabsaccum, *maxaccum*, *minaccum*, *max*, *min*, *maxabs*, *minabs*, *vincr*

Credits

Author: Anthony Kozar
March 2006

New in Csound version 5.01

minaccum

minaccum — Accumulates the minimum value of audio signals.

Description

minaccum compares two audio-rate variables and stores the minimum value between them into the first.

Syntax

```
minaccum aAccumulator, aInput
```

Performance

aAccumulator -- audio variable to store the minimum value

aInput -- signal that *aAccumulator* is compared to

The *minaccum* opcode is designed to accumulate the minimum value from among many audio signals that may be in different note instances, different channels, or otherwise cannot all be compared at once using the *min* opcode. Its semantics are similar to *vincr* since *aAccumulator* is used as both an input and an output variable, except that *minaccum* keeps the minimum value instead of adding the signals together. *minaccum* performs the following operation on each pair of samples:

```
if (aInput < aAccumulator) aAccumulator = aInput
```

aAccumulator will usually be a global audio variable. At the end of any given computation cycle (k-period), after its value is read and used in some way, the accumulator variable should usually be reset to some large enough positive value that will always be greater than the input signals to which it is compared.

See Also

maxaccum, *maxabsaccum*, *minabsaccum*, *max*, *min*, *maxabs*, *minabs*, *vincr*

Credits

Author: Anthony Kozar
March 2006

New in Csound version 5.01

mirror

mirror — Reflects the signal that exceeds the low and high thresholds.

Description

Reflects the signal that exceeds the low and high thresholds.

Syntax

```
ares mirror asig, klow, khigh
```

```
ires mirror isig, ilow, ihigh
```

```
kres mirror ksig, klow, khigh
```

Initialization

isig -- input signal

ilow -- low threshold

ihigh -- high threshold

Performance

xsig -- input signal

klow -- low threshold

khigh -- high threshold

mirror “reflects” the signal that exceeds the low and high thresholds.

This opcode is useful in several situations, such as table indexing or for clipping and modeling a-rate, i-rate or k-rate signals.

See Also

limit, *wrap*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.49

MixerSetLevel

MixerSetLevel — Sets the level of a send to a buss.

Syntax

```
MixerSetLevel isend, ibuss, kgain
```

Description

Sets the level at which signals from the send are added to the buss. The actual sending of the signal to the buss is performed by the *MixerSend* opcode.

Initialization

isend -- The number of the send, for example the number of the instrument sending the signal (but any integer can be used).

ibuss -- The number of the buss, for example the number of the instrument receiving the signal (but any integer can be used).

Setting the gain for a buss also creates the buss.

Performance

kgain -- The level (any real number) at which the signal from the send will be mixed onto the buss. The default is 0.

Use of the mixer requires that instruments setting gains have smaller numbers than instruments sending signals, and that instruments sending signals have smaller numbers than instruments receiving those signals. However, an instrument may have any number of sends or receives. After the final signal is received, *MixerClear* must be invoked to reset the busses before the next kperiod.

Examples

In the orchestra, define an instrument to control mixer levels:

```
instr 1
  MixerSetLevel      p4, p5, p6
endin
```

In the score, use that instrument to set mixer levels:

```
; SoundFonts
; to Chorus
i 1 0 0 100 200 0.9
; to Reverb
i 1 0 0 100 210 0.7
; to Output
i 1 0 0 100 220 0.3

; Kelley Harpsichord
; to Chorus
i 1 0 0 3 200 0.30
; to Reverb
i 1 0 0 3 210 0.9
; to Output
i 1 0 0 3 220 0.1

; Chorus to Reverb
i 1 0 0 200 210 0.5
```

```
; Chorus to Output  
i 1 0 0 200 220 0.5  
; Reverb to Output  
i 1 0 0 210 220 0.2
```

Credits

Michael Gogins (gogins at pipeline dot com).

MixerGetLevel

MixerGetLevel — Gets the level of a send to a buss.

Syntax

```
kgain MixerGetLevel isend, ibuss
```

Description

Gets the level at which signals from the send are being added to the buss. The actual sending of the signal to the buss is performed by the *MixerSend* opcode.

Initialization

isend -- The number of the send, for example the number of the instrument sending the signal.

ibuss -- The number of the buss, for example the number of the instrument receiving the signal.

Performance

kgain -- The level (any real number) at which the signal from the send will be mixed onto the buss.

This opcode reports the level set by *MixerSetLevel* for a send and buss pair.

Use of the mixer requires that instruments setting gains have smaller numbers than instruments sending signals, and that instruments sending signals have smaller numbers than instruments receiving those signals. However, an instrument may have any number of sends or receives. After the final signal is received, *MixerClear* must be invoked to reset the busses to 0 before the next kperiod.

Credits

Michael Gogins (gogins at pipeline dot com).

MixerSend

MixerSend — Mixes an arate signal into a channel of a buss.

Syntax

```
MixerSend asignal, isend, ibuss, ichannel
```

Description

Mixes an arate signal into a channel of a buss.

Initialization

isend -- The number of the send, for example the number of the instrument sending the signal. The gain of the send is controlled by the *MixerSetLevel* opcode. The reason that the sends are numbered is to enable different levels for different sends to be set independently of the actual level of the signals.

ibuss -- The number of the buss, for example the number of the instrument receiving the signal.

ichannel -- The number of the channel. Each buss has *nchnls* channels.

Performance

asignal -- The signal that will be mixed into the indicated channel of the buss.

Use of the mixer requires that instruments setting gains have smaller numbers than instruments sending signals, and that instruments sending signals have smaller numbers than instruments receiving those signals. However, an instrument may have any number of sends or receives. After the final signal is received, *MixerClear* must be invoked to reset the busses to 0 before the next kperiod.

Examples

```
instr 100 ; Fluidsynth output
; INITIALIZATION
; Normalize so iamplitude for p5 of 80 == ampdb(80).
iamplitude = ampdb(p5) * 2.0
; AUDIO
aleft, aright fluidAllOut giFluidsynth
asig1 = aleft * iamplitude
asig2 = aright * iamplitude
; To the chorus.
MixerSend asig1, 100, 200, 0
MixerSend asig2, 100, 200, 1
; To the reverb.
MixerSend asig1, 100, 210, 0
MixerSend asig2, 100, 210, 1
; To the output.
MixerSend asig1, 100, 220, 0
MixerSend asig2, 100, 220, 1
endin
```

Credits

Michael Gogins (gogins at pipeline dot com).

MixerReceive

MixerReceive — Receives an arate signal from a channel of a buss.

Syntax

```
asignal MixerReceive ibuss, ichannel
```

Description

Receives an arate signal that has been mixed onto a channel of a buss.

Initialization

ibuss -- The number of the buss, for example the number of the instrument receiving the signal.

ichannel -- The number of the channel. Each buss has `nchnls` channels.

Performance

asignal -- The signal that has been mixed onto the indicated channel of the buss.

Use of the mixer requires that instruments setting gains have smaller numbers than instruments sending signals, and that instruments sending signals have smaller numbers than instruments receiving those signals. However, an instrument may have any number of sends or receives. After the final signal is received, *MixerClear* must be invoked to reset the busses to 0 before the next kperiod.

Examples

```
instr 220 ; Master output
    ; It applies a bass enhancement, compression and fadeout
    ; to the whole piece, outputs signals, and clears the mixer.
    a1 MixerReceive 220, 0
    a2 MixerReceive 220, 1
    ; Bass enhancement
    a11 butterlp a1, 100
    a12 butterlp a2, 100
    a1 = a1*1.5 +a1
    a2 = a2*1.5 +a2

    ; Global amplitude shape
    kenv linseg 0., p5 / 2.0, p4, p3 - p5, p4, p5 / 2.0, 0.
    a1=a1*kenv
    a2=a2*kenv

    ; Compression
    a1 dam a1, 5000, 0.5, 1, 0.2, 0.1
    a2 dam a2, 5000, 0.5, 1, 0.2, 0.1

    ; Remove DC bias
    a1blocked dcblock dcblock a1
    a2blocked dcblock a2

    ; Output signals
    outs a1blocked, a2blocked
    MixerClear
endin
```

Credits

Michael Gogins (gogins at pipeline dot com).

MixerClear

MixerClear — Resets all channels of a buss to 0.

Syntax

MixerClear

Description

Resets all channels of a buss to 0.

Performance

Use of the mixer requires that instruments setting gains have smaller numbers than instruments sending signals, and that instruments sending signals have smaller numbers than instruments receiving those signals. However, an instrument may have any number of sends or receives. After the final signal is received, *MixerClear* must be invoked to reset the busses to 0 before the next kperiod.

Examples

```
instr 220 ; Master output
    ; It applies a bass enhancement, compression and fadeout
    ; to the whole piece, outputs signals, and clears the mixer.
    a1 MixerReceive 220, 0
    a2 MixerReceive 220, 1
    ; Bass enhancement
    a11 butterlp a1, 100
    a12 butterlp a2, 100
    a1 = a11*1.5 +a1
    a2 = a12*1.5 +a2

    ; Global amplitude shape
    kenv linseg 0., p5 / 2.0, p4, p3 - p5, p4, p5 / 2.0, 0.
    a1=a1*kenv
    a2=a2*kenv

    ; Compression
    a1 dam a1, 5000, 0.5, 1, 0.2, 0.1
    a2 dam a2, 5000, 0.5, 1, 0.2, 0.1

    ; Remove DC bias
    a1blocked dcblock a1
    a2blocked dcblock a2

    ; Output signals
    outs a1blocked, a2blocked
    MixerClear
endin
```

Credits

Michael Gogins (gogins at pipeline dot com).

mode

mode — A filter that simulates a mass-spring-damper system

Description

Filters the incoming signal with the specified resonance frequency and quality factor. It can also be seen as a signal generator for high quality factor, with an impulse for the excitation. You can combine several modes to built complex instruments such as bells or guitar tables.

Syntax

```
aout mode ain, kfreq, kQ [, iskip]
```

Initialization

iskip (optional, default=0) -- if non zero skip the initialisation of the filter.

Performance

aout -- filtered signal

ain -- signal to filter

kfreq -- resonant frequency of the filter



Warning

This filter becomes unstable if $sr/ikfreq < \pi$ (e.g $ikfreq > 14037$ Hz @44kHz)

kQ -- quality factor of the filter

The resonance time is roughly proportionnal to $kQ/kfreq$.

See *Modal Frequency Ratios* for frequency ratios of real intruments which can be used to determine the values of *kfreq*.

Examples

Here is an example of the mode opcode. It uses the file *mode.csd* [examples/mode.csd].

Example 303. Example of the mode opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in      No messages
-odac            -iadc         -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o moogvcf.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```

```
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

instr 1; 2 modes excitator

idur init p3
ifreql1 init p4
ifreql2 init p5
iQl1 init p6
iQl2 init p7
iamp init ampdb(p8)
ifreq21 init p9
ifreq22 init p10
iQ21 init p11
iQ22 init p12

; to simulate the shock between the excitator and the resonator
ashock mpulse 3,0

aexc1 mode ashock,ifreql1,iQl1
aexc1 = aexc1*iamp
aexc2 mode ashock,ifreql2,iQl2
aexc2 = aexc2*iamp

aexc = (aexc1+aexc2)/2

;"Contact" condition : when aexc reaches 0, the excitator loses
;contact with the resonator, and stops "pushing it"
aexc limit aexc,0,3*iamp

; 2modes resonator

ares1 mode aexc,ifreq21,iQ21
ares2 mode aexc,ifreq22,iQ22

ares = (ares1+ares2)/2

display aexc+ares,p3
outs aexc+ares,aexc+ares

endin

</CsInstruments>
<CsScore>

;wooden excitator against glass resonator
i1 0 8 1000 3000 12 8 70 440 888 500 420

;felt against glass
i1 4 8 80 188 8 3 70 440 888 500 420

;wood against wood
i1 8 8 1000 3000 12 8 70 440 630 60 53

;felt against wood
i1 12 8 80 180 8 3 70 440 630 60 53

i1 16 8 1000 3000 12 8 70 440 888 2000 1630
i1 23 8 80 180 8 3 70 440 888 2000 1630

;With a metallic excitator

i1 33 8 1000 1800 1000 720 70 440 882 500 500
i1 37 8 1000 1800 1000 850 70 440 630 60 53

i1 42 8 1000 1800 2000 1720 70 440 442 500 500

</CsScore>
</CsoundSynthesizer>
```

Credits

Original UDO and documentation/example by François Blanc

Opcode translation to C-code by Steven Yi

New in version 5.04

monitor

monitor — Returns the audio spout frame.

Description

Returns the audio spout frame (if active), otherwise it returns zero.

Syntax

```
aout1 [,aout2 ... aoutX] monitor
```

Performance

This opcode can be used for monitoring the output signal from csound. It should not be used for processing the signal further.

See the entry for the *fout* opcode for an example of usage of *monitor*.

See also

fout, the *Mixer opcodes* and the *Zak Patching System*.

Credits

Istvan Varga 2006

moog

moog — An emulation of a mini-Moog synthesizer.

Description

An emulation of a mini-Moog synthesizer.

Syntax

ares **moog** kamp, kfreq, kfiltq, kfiltrate, kvibf, kvamp, iafn, iwfn, ivfn

Initialization

iafn, *iwfn*, *ivfn* -- three table numbers containing the attack waveform (unlooped), the main looping wave form, and the vibrato waveform. The files *mandpluk.aiff* [examples/mandpluk.aiff] and *impuls20.aiff* [examples/impuls20.aiff] are suitable for the first two, and a sine wave for the last.



Note

The files “mandpluk.aiff” and “impuls20.aiff” are also available at <ftp://ftp.cs.bath.ac.uk/pub/dream/documentation/sounds/modelling/>.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kfiltq -- Q of the filter, in the range 0.8 to 0.9

kfiltrate -- rate control for the filter in the range 0 to 0.0002

kvibf -- frequency of vibrato in Hertz. Suggested range is 0 to 12

kvamp -- amplitude of the vibrato

Examples

Here is an example of the moog opcode. It uses the file *moog.csd* [examples/moog.csd], *mandpluk.aiff* [examples/mandpluk.aiff], and *impuls20.aiff* [examples/impuls20.aiff].

Example 304. Example of the moog opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o moog.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```



```
; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kfreq = 220
  kfiltq = 0.81
  kfiltrate = 0
  kvibf = 1.4
  kvamp = 2.22
  iafn = 1
  iwfn = 2
  ivfn = 3

  am moog kamp, kfreq, kfiltq, kfiltrate, kvibf, kvamp, iafn, iwfn, ivfn

; It tends to get loud, so clip moog's amplitude at 30,000.
  al clip am, 2, 30000
  out al
endin

</CsInstruments>
<CsScore>

; Table #1: the "mandpluk.aiff" audio file
f 1 0 8192 1 "mandpluk.aiff" 0 0 0
; Table #2: the "impuls20.aiff" audio file
f 2 0 256 1 "impuls20.aiff" 0 0 0
; Table #3: a sine wave
f 3 0 256 10 1

; Play Instrument #1 for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John fitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.47

moogladder

moogladder — Moog ladder lowpass filter.

Description

Moogladder is an new digital implementation of the Moog ladder filter based on the work of Antti Huovilainen, described in the paper "Non-Linear Digital Implementation of the Moog Ladder Filter" (Proceedings of DaFX04, Univ of Napoli). This implementation is probably a more accurate digital representation of the original analogue filter.

Syntax

```
asig moogladder ain, kcf, kres[, istor]
```

Initialization

istor --initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- input signal.

kcf -- filter cutoff frequency

kres -- resonance, generally < 1 , but not limited to it. Higher than 1 resonance values might cause aliasing, analogue synths generally allow resonances to be above 1.

Examples

Example 305. Example

```
kfe      expseg 500, p3*0.9, 1800, p3*0.1, 3000
kenv     linen 10000, 0.05, p3, 0.05
asig     buzz kenv, 100, sr/(200), 1
afil     moogladder asig, kfe, 1

out afil
```

Credits

Author: Victor Lazzarini;
January 2005

New plugin in version 5

January 2005.

moogvcf

moogvcf — A digital emulation of the Moog diode ladder filter configuration.

Description

A digital emulation of the Moog diode ladder filter configuration.

Syntax

```
ares moogvcf asig, xfco, xres [,iscale, iskip]
```

Initialization

iscale (optional, default=1) -- internal scaling factor. Use if *asig* is not in the range +/-1. Input is first divided by *iscale*, then output is multiplied *iscale*. Default value is 1. (New in Csound version 3.50)

iskip (optional, default=0) -- if non zero skip the initialisation of the filter. (New in Csound version 4.23f13 and 5.0)

Performance

asig -- input signal

xfco -- filter cut-off frequency in Hz. As of version 3.50, may i-,k-, or a-rate.

xres -- amount of resonance. Self-oscillation occurs when *xres* is approximately one. As of version 3.50, may a-rate, i-rate, or k-rate.

moogvcf is a digital emulation of the Moog diode ladder filter configuration. This emulation is based loosely on the paper “Analyzing the Moog VCF with Considerations for Digital Implementation” by Stilson and Smith (CCRMA). This version was originally coded in Csound by Josep Comajuncosas. Some modifications and conversion to C were done by Hans Mikelson



Warning

This filter requires that the input signal be normalized to one. This can be easily achieved using *Odbfs*, like this:

```
ares moogvcf asig, kfco, kres, Odbfs
```

You can also use *moogvcf2* which defaults scaling to *Odbfs*.

Examples

Here is an example of the moogvcf opcode. It uses the file *moogvcf.csd* [examples/moogvcf.csd].

Example 306. Example of the moogvcf opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

<CsoundSynthesizer>

```
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o moogvcf.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a nice sawtooth waveform.
asig vco 32000, 220, 1

; Vary the filter-cutoff frequency from .2 to 2 KHz.
kfco line 200, p3, 2000

; Set the resonance amount to one.
krez init 1

; Scale the amplitude to 32768.
iscale = 32768

al moogvcf asig, kfco, krez, iscale

out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave for the vco opcode.
f 1 0 16384 10 1

; Play Instrument #1 for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

moogvcf2, biquad, rezy

Credits

Author: Hans Mikelson
October 1998

Example written by Kevin Conder.

New in Csound version 3.49

moogvcf2

moogvcf2 — A digital emulation of the Moog diode ladder filter configuration.

Description

A digital emulation of the Moog diode ladder filter configuration.

Syntax

```
ares moogvcf2 asig, xfco, xres [,iscale, iskip]
```

Initialization

iscale (optional, default=0dBfs) -- internal scaling factor, as the operation of the code requires the signal to be in the range +/-1. Input is first divided by *iscale*, then output is multiplied by *iscale*.

iskip (optional, default=0) -- if non zero skip the initialisation of the filter.

Performance

asig -- input signal

xfco -- filter cut-off frequency in Hz. which may be i-,k-, or a-rate.

xres -- amount of resonance. Self-oscillation occurs when *xres* is approximately one. May be a-rate, i-rate, or k-rate.

moogvcf2 is a digital emulation of the Moog diode ladder filter configuration. This emulation is based loosely on the paper “Analyzing the Moog VCF with Considerations for Digital Implementation” by Stilson and Smith (CCRMA). This version was originally coded in Csound by Josep Comajuncosas. Some modifications and conversion to C were done by Hans Mikelson and then adjusted.

moogvcf2 is identical to *moogvcf*, except that the *iscale* parameter defaults to *0dbfs* instead of 0, guaranteeing that amplitude will usually be OK.

Examples

Here is an example of the *moogvcf2* opcode. It uses the file *moogvcf2.csd* [examples/moogvcf2.csd].

Example 307. Example of the moogvcf2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o moogvcf.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```

```
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a nice sawtooth waveform.
asig vco 32000, 220, 1

; Vary the filter-cutoff frequency from .2 to 2 KHz.
kfco line 200, p3, 2000

; Set the resonance amount to one.
krez init 1

al moogvcf2 asig, kfco, krez

out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave for the vco opcode.
f 1 0 16384 10 1

; Play Instrument #1 for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

moogvcf, biquad, rezzy

Credits

Author: Hans Mikelson and John ffitch
October 1998/ July 2006

Example written by Kevin Conder.

New in Csound version 5.03

moscil

moscil — Sends a stream of the MIDI notes.

Description

Sends a stream of the MIDI notes.

Syntax

```
moscil kchn, knum, kvel, kdur, kpause
```

Performance

kchn -- MIDI channel number (1-16)

knum -- note number (0-127)

kvel -- velocity (0-127)

kdur -- note duration in seconds

kpause -- pause duration after each noteoff and before new note in seconds

moscil and *midion* are the most powerful MIDI OUT opcodes. *moscil* (MIDI oscil) plays a stream of notes of *kdur* duration. Channel, pitch, velocity, duration and pause can be controlled at k-rate, allowing very complex algorithmically generated melodic lines. When current instrument is deactivated, the note played by current instance of *moscil* is forcedly truncated.

Any number of *moscil* opcodes can appear in the same Csound instrument, allowing a counterpoint-style polyphony within a single instrument.

Examples

Here is an example of the *moscil* opcode. It uses the file *moscil.csd* [examples/moscil.csd].

Example 308. Example of the moscil opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

This example generates a stream of notes for every note received on the MIDI input. It generates MIDI notes on csound's MIDI output, so be sure to connect something.

```
<CsoundSynthesizer>
<CsOptions>

; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d           -M0  -Q1;;RT audio I/O with MIDI in
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Example by Giorgio Zucco 2007
```

```
instr 1 ;Triggered by MIDI notes on channel 1

inote notnum
ivel      veloc

kpitch = 40
kfreq  = 2

kdur   = .04
kpause = .1

kl      lfo      kpitch, kfreq,5

;plays a stream of notes of kdur duration on MIDI channel 1
moscil 1, inote + kl, ivel,  kdur, kpause

endin

</CsInstruments>
<CsScore>
; Dummy ftable
f0 60
</CsScore>
</CsoundSynthesizer>
```

See Also

midion, midion2, noteon, noteoff, noteondur, noteondur2

Credits

Author: Gabriel Maldonado
Italy
May 1997

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

mpulse

mpulse — Generates a set of impulses.

Description

Generates a set of impulses of amplitude *kamp* separated by *kintvl* seconds (or samples if *kintvl* is negative). The first impulse is generated after a delay of *ioffset* seconds.

Syntax

```
ares mpulse kamp, kintvl [, ioffset]
```

Initialization

ioffset (optional, default=0) -- the delay before the first impulse. If it is negative, the value is taken as the number of samples, otherwise it is in seconds. Default is zero.

Performance

kamp -- amplitude of the impulses generated

kintvl -- Interval of time in seconds (or samples if *kintvl* is negative) to the next pulse.

After the initial delay, an impulse of *kamp* amplitude is generated as a single sample. Immediately after generating the impulse, the time of the next one is determined from the value of *kintvl* at that precise moment. This means that any changes in *kintvl* between impulses are discarded. If *kintvl* is zero, there is an infinite wait to the next impulse. If *kintvl* is negative, the interval is counted in number of samples rather than seconds.

Examples

Here is an example of the mpulse opcode. It uses the file *mpulse.csd* [examples/mpulse.csd].

Example 309. Example of the mpulse opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o mpulse.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

gkfreq init 0.1

instr 1
  kamp = 10000
```

```
al mpulse kamp, gkfreq
out al
endin

instr 2
; Assign the value of p4 to gkfreq
gkfreq init p4
endin
</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 10
i 2 2 1 0.05
i 2 4 1 0.01
i 2 6 1 0.005
i 2 8 1 0.001
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by John fitch.

New in version 4.08

Example written by Kevin Conder.

mrtmsg

mrtmsg — Send system real-time messages to the MIDI OUT port.

Description

Send system real-time messages to the MIDI OUT port.

Syntax

```
mrtmsg imgtype
```

Initialization

imgtype -- type of real-time message:

- 1 sends a START message (0xFA);
- 2 sends a CONTINUE message (0xFB);
- 0 sends a STOP message (0xFC);
- -1 sends a SYSTEM RESET message (0xFF);
- -2 sends an ACTIVE SENSING message (0xFE)

Performance

Sends a real-time message once, in init stage of current instrument. *imgtype* parameter is a flag to indicate the message type.

See Also

mclock

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

multitap

multitap — Multitap delay line implementation.

Description

Multitap delay line implementation.

Syntax

```
ares multitap asig [, itime1] [, igain1] [, itime2] [, igain2] [...]
```

Initialization

The arguments *itime* and *igain* set the position and gain of each tap.

The delay line is fed by *asig*.

Examples

```
a1      oscil      1000, 100, 1
a2      multitap   a1, 1.2, .5, 1.4, .2
out      out      a2
```

This results in two delays, one with length of 1.2 and gain of .5, and one with length of 1.4 and gain of .2.

Credits

Author: Paris Smaragdis
MIT, Cambridge
1996

mute

mute — Mutes/unmutes new instances of a given instrument.

Description

Mutes/unmutes new instances of a given instrument.

Syntax

```
mute insnum [, iswitch]
```

```
mute "insname" [, iswitch]
```

Initialization

insnum -- instrument number. Equivalent to *p1* in a score *i statement*.

"insname" -- A string (in double-quotes) representing a named instrument.

iswitch (optional, default=0) -- represents a switch to mute/unmute an instrument. A value of 0 will mute new instances of an instrument, other values will unmute them. The default value is 0.

Performance

All new instances of instrument *inst* will be muted (*iswitch* = 0) or unmuted (*iswitch* not equal to 0). There is no difficulty with muting muted instruments or unmuting unmuted instruments. The mechanism is the same as used by the score *q statement*. For example, it is possible to mute in the score and unmute in some instrument.

Muting/Unmuting is indicated by a message (depending on message level).

Examples

Here is an example of the mute opcode. It uses the file *mute.csd* [examples/mute.csd].

Example 310. Example of the mute opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o mute.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Mute Instrument #2.
mute 2
```

```
; Instrument #1.
instr 1
  al oscils 10000, 440, 0
  out al
endin

; Instrument #2.
instr 2
  al oscils 10000, 880, 0
  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Example written by Kevin Conder.

New in version 4.22

mxadsr

`mxadsr` — Calculates the classical ADSR envelope using the *expsegr* mechanism.

Description

Calculates the classical ADSR envelope using the *expsegr* mechanism.

Syntax

```
ares mxadsr iatt, idec, islev, irel [, idel] [, ireltim]
```

```
kres mxadsr iatt, idec, islev, irel [, idel] [, ireltim]
```

Initialization

iatt -- duration of attack phase

idec -- duration of decay

islev -- level for sustain phase

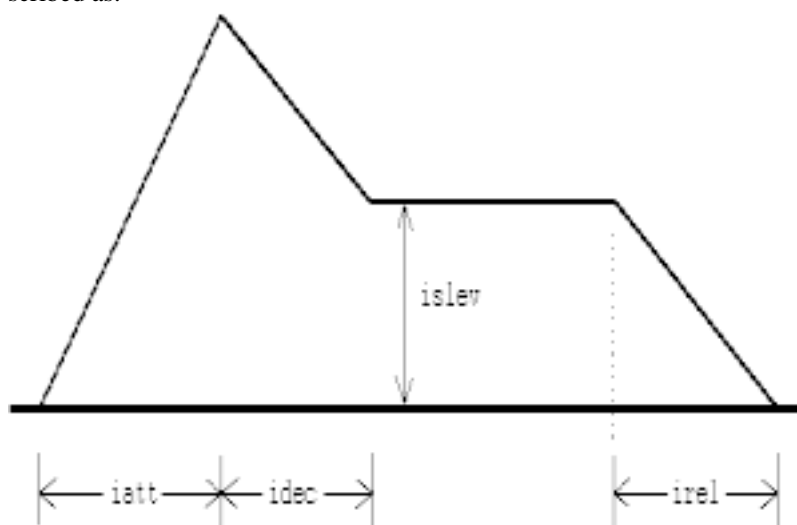
irel -- duration of release phase

idel (optional, default=0) -- period of zero before the envelope starts

ireltim (optional, default=-1) -- Control release time after receiving a MIDI noteoff event. If less than zero, the longest release time given in the current instrument is used. If zero or more, the given value will be used for release time. Its default value is -1. (New in Csound 3.59 - not yet properly tested)

Performance

The envelope is in the range 0 to 1 and may need to be scaled further. The envelope may be described as:



Picture of an ADSR envelope.

The length of the sustain is calculated from the length of the note. This means *adsr* is not suitable

for use with MIDI events. The opcode *madsr* uses the *linsegr* mechanism, and so can be used in MIDI applications. The opcode *mxadsr* is identical to *madsr* except it uses exponential, rather than linear, line segments.

You can use other pre-made envelopes which start a release segment upon receiving a note off message, like *linsegr* and *expsegr*, or you can construct more complex envelopes using *xtratim* and *release*. Note that you don't need to use *xtratim* if you are using *mxadsr*, since the time is extended automatically.

mxadsr is new in Csound version 3.51.

See Also

linsegr, *expsegr*, *envlpxr*, *mxadsr*, *madsr*, *adsr*, *expon*, *expseg*, *expsega* *line*, *linseg*, *xtratim*

Credits

Author: John ffitch

November 2002. Thanks to Rasmus Ekman, added documentation for the *ireltim* parameter.

November 2003. Thanks to Kanata Motohashi, fixed the link to the *linsegr* opcode.

nchnls

nchnls — Sets the number of channels of audio output.

Description

These statements are global value *assignments*, made at the beginning of an orchestra, before any instrument block is defined. Their function is to set certain *reserved symbol variables* that are required for performance. Once set, these reserved symbols can be used in expressions anywhere in the orchestra.

Syntax

```
nchnls = iarg
```

Initialization

nchnls = (optional) -- set number of channels of audio output to *iarg*. (1 = mono, 2 = stereo, 4 = quadraphonic.) The default value is 1 (mono).

In addition, any *global variable* [48] can be initialized by an *init-time assignment* anywhere before the first *instr statement*. All of the above assignments are run as instrument 0 (i-pass only) at the start of real performance.

See Also

kr, *ksmps*, *sr*

nestedap

nestedap — Three different nested all-pass filters.

Description

Three different nested all-pass filters, useful for implementing reverbs.

Syntax

```
ares nestedap asig, imode, imaxdel, idel1, igain1 [, idel2] [, igain2] \  
      [, idel3] [, igain3] [, istor]
```

Initialization

imode -- operating mode of the filter:

- 1 = simple all-pass filter
- 2 = single nested all-pass filter
- 3 = double nested all-pass filter

idel1, *idel2*, *idel3* -- delay times of the filter stages. Delay times are in seconds and must be greater than zero. *idel1* must be greater than the sum of *idel2* and *idel3*.

igain1, *igain2*, *igain3* -- gain of the filter stages.

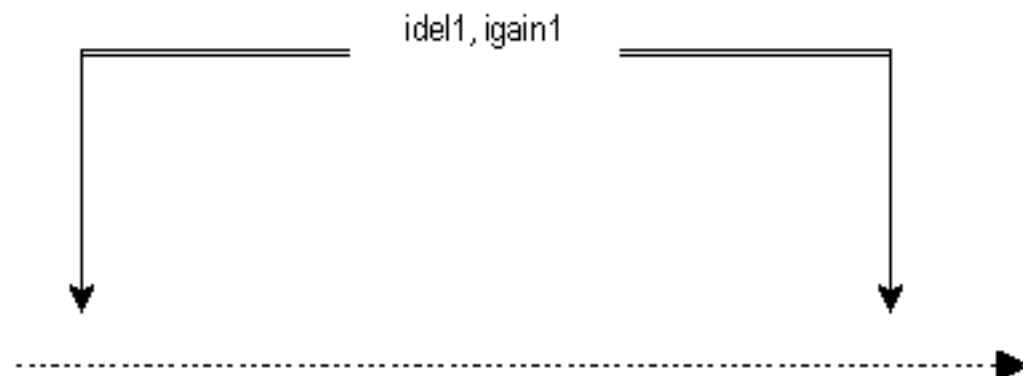
imaxdel -- will be necessary if k-rate delays are implemented. Not currently used.

istor -- Skip initialization if non-zero (default: 0).

Performance

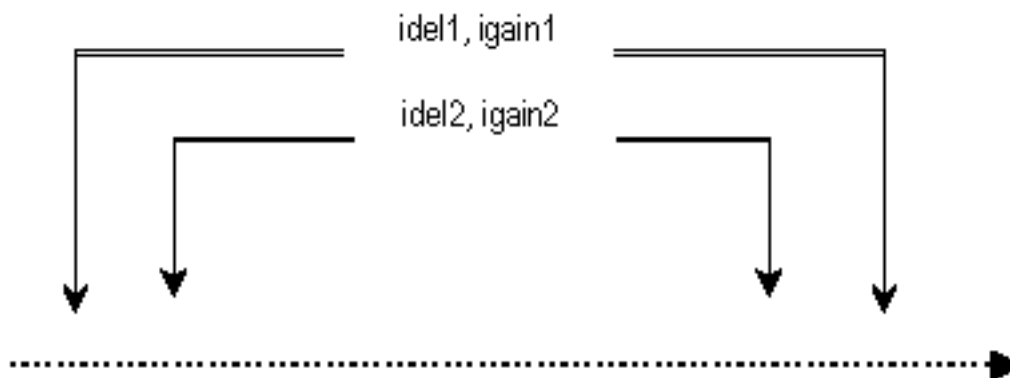
asig -- input signal

If *imode* = 1, the filter takes the form:



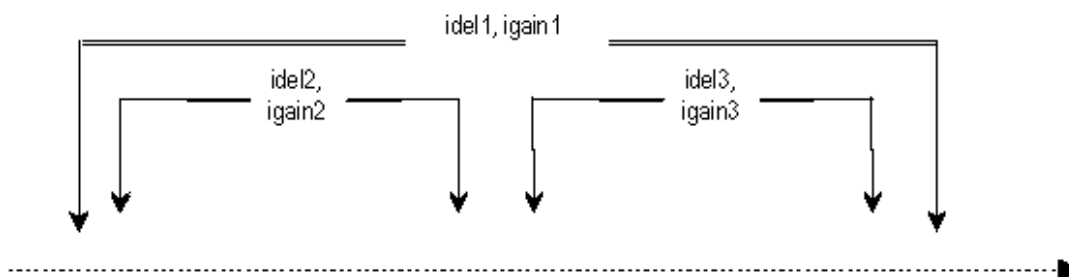
Picture of imode 1 filter.

If *imode* = 2, the filter takes the form:



Picture of imode 2 filter.

If *imode* = 3, the filter takes the form:



Picture of imode 3 filter.

Examples

Here is an example of the *nestedap* opcode. It uses the file *nestedap.csd* [examples/nestedap.csd], and *beats.wav* [examples/beats.wav].

Example 311. Example of the *nestedap* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o nestedap.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

instr 5
  insnd      =          p4
  gasig      diskin insnd, 1
endin

instr 10
  imax       =          1
  idel1      =          p4/1000
  igain1     =          p5
  idel2      =          p6/1000
  igain2     =          p7
  idel3      =          p8/1000
  igain3     =          p9
  idel4      =          p10/1000
```

```

    igain4    =      p11
    idel5     =      p12/1000
    igain5     =      p13
    idel6     =      p14/1000
    igain6     =      p15

    afdbk      init 0

    aout1      nestedap gasig+afdbk*.4, 3, imax, idel1, igain1, idel2, igain2, idel3, igain3

    aout2      nestedap aout1, 2, imax, idel4, igain4, idel5, igain5

    aout       nestedap aout2, 1, imax, idel6, igain6

    afdbk      butterlp aout, 1000

               outs gasig+(aout+aout1)/2, gasig-(aout+aout1)/2

gasig         =      0
endin

</CsInstruments>
<CsScore>

f1 0 8192 10 1

; Diskin
;   Sta Dur   Soundin
i5 0    3    "beats.wav"

; Reverb
;   St Dur Del1 Gn1 Del2 Gn2 Del3 Gn3 Del4 Gn4 Del5 Gn5 Del6 Gn6
i10 0    4    97    .11 23    .07 43    .09 72    .2  53    .2  119    .3
e

</CsScore>
</CsoundSynthesizer>

```

Credits

Author: Hans Mikelson
February 1999

New in Csound version 3.53

The example was updated May 2002, thanks to Hans Mikelson

nlfilt

nlfilt — A filter with a non-linear effect.

Description

Implements the filter:

$$Y\{n\} = a Y\{n-1\} + b Y\{n-2\} + d Y^2\{n-L\} + X\{n\} - C$$

described in Dobson and Fitch (ICMC'96)

Syntax

ares **nlfilt** ain, ka, kb, kd, kC, kL

Performance

1. Non-linear effect. The range of parameters are:

a = b = 0
d = 0.8, 0.9, 0.7
C = 0.4, 0.5, 0.6
L = 20

This affects the lower register most but there are audible effects over the whole range. We suggest that it may be useful for coloring drums, and for adding arbitrary highlights to notes.

2. Low Pass with non-linear. The range of parameters are:

a = 0.4
b = 0.2
d = 0.7
C = 0.11
L = 20, ... 200

There are instability problems with this variant but the effect is more pronounced of the lower register, but is otherwise much like the pure comb. Short values of L can add attack to a sound.

3. High Pass with non-linear. The range of parameters are:

a = 0.35
b = -0.3
d = 0.95
C = 0.2, ... 0.4
L = 200

4. High Pass with non-linear. The range of parameters are:

$a = 0.7$
 $b = -0.2, \dots 0.5$
 $d = 0.9$
 $C = 0.12, \dots 0.24$
 $L = 500, 10$

The high pass version is less likely to oscillate. It adds scintillation to medium-high registers. With a large delay L it is a little like a reverberation, while with small values there appear to be formant-like regions. There are arbitrary color changes and resonances as the pitch changes. Works well with individual notes.



Warning

The "useful" ranges of parameters are not yet mapped.

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
1997

New in version 3.44

noise

noise — A white noise generator with an IIR lowpass filter.

Description

A white noise generator with an IIR lowpass filter.

Syntax

```
ares noise xamp, kbeta
```

Performance

xamp -- amplitude of final output

kbeta -- beta of the lowpass filter. Should be in the range of -1 to 1.

The filter equation is:

$$y_n = \sqrt{(1 - \beta^2)} * x_n + \beta y_{(n-1)}$$

where x_n is the original white noise and y_n is lowpass filtered noise. The higher # is, the lower the filter's cut-off frequency. The cutoff frequency is roughly $sr * ((1-kbeta)/2)$.

Examples

Here is an example of the noise opcode. It uses the file *noise.csd* [examples/noise.csd].

Example 312. Example of the noise opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o noise.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000

  ; Change the beta value linearly from 0 to 1.
  kbeta line 0, p3, 1
```

```
    al noise kamp, kbeta
    out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the noise opcode controlling the kbeta parameter with a GUI interface. It uses the file *noise-2.csd* [examples/noise-2.csd].

Example 313. Example of the noise opcode controlled with a GUI.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in      No messages
-odac           -iadc          -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o noise.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

FLpanel "noise", 200, 50, -1, -1
    gkbeta, gislider1 FLslider "kbeta", -1, 1, 0, 5, -1, 180, 20, 10, 10
FLpanelEnd
FLrun

instr 1
    iamp = 0dbfs / 4 ; Peaks 12 dB below 0dbfs
    print iamp

    al noise iamp, gkbeta
    printk2 gkbeta
    outs al,al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one minute.
i 1 0 60
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch
University of Bath, Codemist. Ltd.
Bath, UK
December 2000

Example written by Kevin Conder.

New in Csound version 4.10

noteoff

noteoff — Send a noteoff message to the MIDI OUT port.

Description

Send a noteoff message to the MIDI OUT port.

Syntax

```
noteoff ichn, inum, ivel
```

Initialization

ichn -- MIDI channel number (1-16)

inum -- note number (0-127)

ivel -- velocity (0-127)

Performance

noteon (i-rate note on) and *noteoff* (i-rate note off) are the simplest MIDI OUT opcodes. *noteon* sends a MIDI noteon message to MIDI OUT port, and *noteoff* sends a noteoff message. A *noteon* opcode must always be followed by an *noteoff* with the same channel and number inside the same instrument, otherwise the note will play endlessly.

These *noteon* and *noteoff* opcodes are useful only when introducing a *timeout* statement to play a non-zero duration MIDI note. For most purposes, it is better to use *noteondur* and *noteondur2*.

See Also

noteon, *noteondur*, *noteondur2*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

noteon

noteon — Send a noteon message to the MIDI OUT port.

Description

Send a noteon message to the MIDI OUT port.

Syntax

```
noteon ichn, inum, ivel
```

Initialization

ichn -- MIDI channel number (1-16)

inum -- note number (0-127)

ivel -- velocity (0-127)

Performance

noteon (i-rate note on) and *noteoff* (i-rate note off) are the simplest MIDI OUT opcodes. *noteon* sends a MIDI noteon message to MIDI OUT port, and *noteoff* sends a noteoff message. A *noteon* opcode must always be followed by an *noteoff* with the same channel and number inside the same instrument, otherwise the note will play endlessly.

These *noteon* and *noteoff* opcodes are useful only when introducing a *timeout* statement to play a non-zero duration MIDI note. For most purposes, it is better to use *noteondur* and *noteondur2*.

See Also

noteoff, *noteondur*, *noteondur2*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

noteondur

noteondur — Sends a noteon and a noteoff MIDI message both with the same channel, number and velocity.

Description

Sends a noteon and a noteoff MIDI message both with the same channel, number and velocity.

Syntax

```
noteondur ichn, inum, ivel, idur
```

Initialization

ichn -- MIDI channel number (1-16)

inum -- note number (0-127)

ivel -- velocity (0-127)

idur -- how long, in seconds, this note should last.

Performance

noteondur (i-rate note on with duration) sends a noteon and a noteoff MIDI message both with the same channel, number and velocity. Noteoff message is sent after *idur* seconds are elapsed by the time *noteondur* was active.

noteondur differs from *noteondur2* in that *noteondur* truncates note duration when current instrument is deactivated by score or by real-time playing, while *noteondur2* will extend performance time of current instrument until *idur* seconds have elapsed. In real-time playing, it is suggested to use *noteondur* also for undefined durations, giving a large *idur* value.

Any number of *noteondur* opcodes can appear in the same Csound instrument, allowing chords to be played by a single instrument.

Examples

Here is an example of the *noteondur* opcode. It uses the file *noteondur.csd* [examples/noteondur.csd].

Example 314. Example of the *noteondur* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

This example generates notes for every note received on the MIDI input. It generates MIDI notes on csound's MIDI output, so be sure to connect something.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d           -M0  -Q1;;;RT audio I/O with MIDI in
</CsOptions>
```

```
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Example by Giorgio Zucco 2007

instr 1 ;Turned on by MIDI notes on channel 1

    ifund    notnum
    ivel      veloc
    idur = 1

    ;chord with single key
    noteondur      1, ifund,    ivel, idur
    noteondur      1, ifund+3, ivel, idur
    noteondur      1, ifund+7, ivel, idur
    noteondur      1, ifund+9, ivel, idur

endin

</CsInstruments>
<CsScore>
; Play Instrument #1 for 60 seconds.

i1 0 60

</CsScore>
</CsoundSynthesizer>
```

See Also

noteoff, noteon, noteondur2, midion, midion2

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

noteondur2

noteondur2 — Sends a noteon and a noteoff MIDI message both with the same channel, number and velocity.

Description

Sends a noteon and a noteoff MIDI message both with the same channel, number and velocity.

Syntax

```
noteondur2 ichn, inum, ivel, idur
```

Initialization

ichn -- MIDI channel number (1-16)

inum -- note number (0-127)

ivel -- velocity (0-127)

idur -- how long, in seconds, this note should last.

Performance

noteondur2 (i-rate note on with duration) sends a noteon and a noteoff MIDI message both with the same channel, number and velocity. Noteoff message is sent after *idur* seconds are elapsed by the time *noteondur2* was active.

noteondur differs from *noteondur2* in that *noteondur* truncates note duration when current instrument is deactivated by score or by real-time playing, while *noteondur2* will extend performance time of current instrument until *idur* seconds have elapsed. In real-time playing, it is suggested to use *noteondur* also for undefined durations, giving a large *idur* value.

Any number of *noteondur2* opcodes can appear in the same Csound instrument, allowing chords to be played by a single instrument.

Examples

Here is an example of the *noteondur2* opcode. It uses the file *noteondur2.csd* [examples/noteondur2.csd].

Example 315. Example of the noteondur2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

This example generates notes for every note received on the MIDI input. It generates MIDI notes on csound's MIDI output, so be sure to connect something.

```
<CsoundSynthesizer>
<CsOptions>

; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc     -d             -M0  -Q1;;;RT audio I/O with MIDI in
```

```
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Example by Giorgio Zucco 2007

instr 1

    ifund    notnum
    ivel          veloc
    idur = 1

    ;chord with single key
    noteondur2    1, ifund,    ivel, idur
    noteondur2    1, ifund+3, ivel, idur
    noteondur2    1, ifund+7, ivel, idur
    noteondur2    1, ifund+9, ivel, idur

endin

</CsInstruments>
<CsScore>
; Dummy ftable
f 0 60
</CsScore>
</CsoundSynthesizer>
```

See Also

noteoff, noteon, noteondur, midion, midion2

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

notnum

notnum — Get a note number from a MIDI event.

Description

Get a note number from a MIDI event.

Syntax

ival **notnum**

Performance

Get the MIDI byte value (0 - 127) denoting the note number of the current event.

Examples

Here is an example of the notnum opcode. It uses the file *notnum.csd* [examples/notnum.csd].

Example 316. Example of the notnum opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o notnum.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  il notnum

  print il
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 12 seconds.
i 1 0 12
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the notnum opcode used to produce audio output. It uses the file *notnum_complex.csd* [examples/notnum_complex.csd]

Example 317. Complex example of the notnum opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac      -iadc      -d          -M0    ;;RT audio I/O with MIDI in
</CsOptions>
<CsInstruments>

sr      =      44100
ksmps   =      10
nchnls  =      2

; Set MIDI channel 1 to play instr 1.
      massign 1, 1

      instr    1

; Returns MIDI note number - an integer in range (0-127)
iNum    notnum

; Convert MIDI note number to Hz
iHz      = (440.0*exp(log(2.0)*((iNum)-69.0)/12.0))

; Generate audio by indexing a table; fixed amplitude.
aosc     oscil    10000, iHz, 1

; Since there is no enveloping, there will be clicks.
outs     aosc, aosc

      endin

</CsInstruments>
<CsScore>

; Generate a Sine-wave to be indexed at audio rate
; by the oscil opcode.
f1       0      16384    10      1

; Keep the score "open" for 1 hour so that MIDI
; notes can allocate new note events, arbitrarily.
f0       3600

e
</CsScore>
</CsoundSynthesizer>
```

See Also

aftouch, ampmidi, cpsmidi, cpsmidib, midictrl, octmidi, octmidib, pchbend, pchmidi, pchmidib, velocity

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Examples written by Kevin Conder and David Akbari.

nreverb

nreverb — A reverberator consisting of 6 parallel comb-lowpass filters.

Description

This is a reverberator consisting of 6 parallel comb-lowpass filters being fed into a series of 5 all-pass filters. *nreverb* replaces *reverb2* (version 3.48) and so both opcodes are identical.

Syntax

```
ares nreverb asig, ktime, khdif [, iskip] [,inumCombs] [, ifnCombs] \  
      [, inumAlpas] [, ifnAlpas]
```

Initialization

iskip (optional, default=0) -- Skip initialization if present and non-zero.

inumCombs (optional) -- number of filter constants in comb filter. If omitted, the values default to the nreverb constants. New in Csound version 4.09.

ifnCombs - function table with *inumCombs* comb filter time values, followed the same number of gain values. The ftable should not be rescaled (use negative fgen number). Positive time values are in seconds. The time values are converted internally into number of samples, then set to the next greater prime number. If the time is negative, it is interpreted directly as time in sample frames, and no processing is done (except negation). New in Csound version 4.09.

inumAlpas, *ifnAlpas* (optional) -- same as *inumCombs/ifnCombs*, for allpass filter. New in Csound 4.09.

Performance

The input signal *asig* is reverberated for *ktime* seconds. The parameter *khdif* controls the high frequency diffusion amount. The values of *khdif* should be from 0 to 1. If *khdif* is set to 0 the all the frequencies decay with the same speed. If *khdif* is 1, high frequencies decay faster than lower ones. If *ktime* is inadvertently set to a non-positive number, *ktime* will be reset automatically to 0.01. (New in Csound version 4.07.)

As of Csound version 4.09, *nreverb* may read any number of comb and allpass filter from an ftable.

Examples

Here is a simple example of the nreverb opcode. It uses the file *nreverb.csd* [examples/nreverb.csd].

Example 318. Simple example of the nreverb opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  No messages  
-odac        -iadc     -d      ;;RT audio I/O  
; For Non-realtime output leave only the line below:  
; -o nreverb.wav -W ;; for file output any platform  
</CsOptions>
```

```
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 1
  a1 oscil 10000, 440, 1
  a2 nreverb a1, 2.5, .3
  out a1 + a2 * .2
endin

</CsInstruments>
<CsScore>

; Table 1: an ordinary sine wave.
f 1 0 32768 10 1

i 1 0.0 0.5
i 1 1.0 0.5
i 1 2.0 0.5
i 1 3.0 0.5
i 1 4.0 0.5
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the `nreverb` opcode using an `ftable` for filter constants. It uses the file `nreverb_ftable.csd` [examples/nreverb_ftable.csd], and `beats.wav` [examples/beats.wav].

Example 319. An example of the `nreverb` opcode using an `ftable` for filter constants.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o nreverb_ftable.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 1
  a1 soundin "beats.wav"
  a2 nreverb a1, 1.5, .75, 0, 8, 71, 4, 72
  out a1 + a2 * .4
endin

</CsInstruments>
<CsScore>

; freeverb time constants, as direct (negative) sample, with arbitrary gains
f71 0 16  -2  -1116 -1188 -1277 -1356 -1422 -1491 -1557 -1617  0.8  0.79  0.78  0.77  0.76  0.75

f72 0 16  -2  -556 -441 -341 -225  0.7  0.72  0.74  0.76

i1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Authors: Paris Smaragdis (*reverb2*)
MIT, Cambridge
1995

Author: Richard Karpen (*nreverb*)
Seattle, Wash
1998

nrpn

nrpn — Sends a Non-Registered Parameter Number to the MIDI OUT port.

Description

Sends a NPRN (Non-Registered Parameter Number) message to the MIDI OUT port each time one of the input arguments changes.

Syntax

```
nrpn kchan, kparmnum, kparmvalue
```

Performance

kchan -- MIDI channel (1-16)

kparmnum -- number of NRPN parameter

kparmvalue -- value of NRPN parameter

This opcode sends new message when the MIDI translated value of one of the input arguments changes. It operates at k-rate. Useful with the MIDI instruments that recognize NRPNs (for example with the newest sound-cards with internal MIDI synthesizer such as SB AWE32, AWE64, GUS etc. in which each patch parameter can be changed during the performance via NRPN)

Credits

Author: Gabriel Maldonado
Italy
1998

New in Csound version 3.492

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

nsamp

nsamp — Returns the number of samples loaded into a stored function table number.

Description

Returns the number of samples loaded into a stored function table number.

Syntax

`nsamp(x)` (init-rate args only)

Performance

Returns the number of samples loaded into stored function table number *x* by GEN01. This is useful when a sample is shorter than the power-of-two function table that holds it. New in Csound version 3.49.

As of Csound version 5.02, *flen* works with deferred-length function tables (see GEN01).

nsamp differs from *flen* in that *nsamp* gives the number of sample frames loaded, while *flen* gives the total number of samples. For example, with a stereo sound file of 10000 samples, *flen()* would return 19999 (i.e. a total of 20000 mono samples, not including a guard point), but *nsamp()* returns 10000.

Examples

Here is an example of the *nsamp* opcode. It uses the file *nsamp.csd* [examples/nsamp.csd], and *mary.wav* [examples/mary.wav].

Example 320. Example of the nsamp opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o nsamp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the size (in samples) of Table #1.
isz = nsamp(1)
print isz
endin

</CsInstruments>
<CsScore>

; Table #1: Use an audio file.
```

```
f 1 0 262144 1 "mary.wav" 0 0 0
; Play Instrument #1 for 1 second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Since the audio file “mary.wav” has 154390 samples, its output should include a line like this:

```
instr 1:  isz = 154390.000
```

See Also

ftchnls, filen, filptim, ftsr

Credits

Author: Gabriel Maldonado
Italy
October 1998

Example written by Kevin Conder.

nstrnum

nstrnum — Returns the number of a named instrument.

Description

Returns the number of a named instrument.

Syntax

```
insno nstrnum "name"
```

Initialization

insno -- the instrument number of the named instrument.

Performance

"name" -- the named instrument's name.

If an instrument with the specified name does not exist, an init error occurs, and -1 is returned.

Credits

Author: Istvan Varga
New in version 4.23
Written in the year 2002.

ntrpol

ntrpol — Calculates the weighted mean value of two input signals.

Description

Calculates the weighted mean value (i.e. linear interpolation) of two input signals

Syntax

```
ares ntrpol asig1, asig2, kpoint [, imin] [, imax]
```

```
ires ntrpol isig1, isig2, ipoint [, imin] [, imax]
```

```
kres ntrpol ksig1, ksig2, kpoint [, imin] [, imax]
```

Initialization

imin -- minimum xpoint value (optional, default 0)

imax -- maximum xpoint value (optional, default 1)

Performance

xsig1, *xsig2* -- input signals

xpoint -- interpolation point between the two values

ntrpol opcode outputs the linear interpolation between two input values. *xpoint* is the distance of evaluation point from the first value. With the default values of *imin* and *imax*, (0 and 1) a zero value indicates no distance from the first value and the maximum distance from the second one. With a 0.5 value, *ntrpol* will output the mean value of the two inputs, indicating the exact half point between *xsig1* and *xsig2*. A 1 value indicates the maximum distance from the first value and no distance from the second one. The range of *xpoint* can be also defined with *imin* and *imax* to make its management easier.

These opcodes are useful for crossfading two signals.

Credits

Author: Gabriel Maldonado
Italy
October 1998

New in Csound version 3.49

octave

octave — Calculates a factor to raise/lower a frequency by a given amount of octaves.

Description

Calculates a factor to raise/lower a frequency by a given amount of octaves.

Syntax

`octave(x)`

This function works at a-rate, i-rate, and k-rate.

Initialization

x -- a value expressed in octaves.

Performance

The value returned by the *octave* function is a factor. You can multiply a frequency by this factor to raise/lower it by the given amount of octaves.

Examples

Here is an example of the octave opcode. It uses the file *octave.csd* [examples/octave.csd].

Example 321. Example of the octave opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o octave.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; The root note is A above middle-C (440 Hz)
iroot = 440

; Raise the root note by two octaves.
ioctaves = 2

; Calculate the new note.
ifactor = octave(ioctaves)
inew = iroot * ifactor

; Print out of all of the values.
print iroot
print ifactor
```

```
    print inew
  endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  iroot = 440.000
instr 1:  ifactor = 4.000
instr 1:  inew = 1760.149
```

See Also

cent, db, semitone

Credits

Example written by Kevin Conder.

New in version 4.16

octcps

octcps — Converts a cycles-per-second value to octave-point-decimal.

Description

Converts a cycles-per-second value to octave-point-decimal.

Syntax

```
octcps (cps) (init- or control-rate args only)
```

where the argument within the parentheses may be a further expression.

Performance

octcps and its related opcodes are really *value converters* with a special function of manipulating pitch data.

Data concerning pitch and frequency can exist in any of the following forms:

Table 13. Pitch and Frequency Values

Name	Abbreviation
octave point pitch-class (8ve.pc)	pch
octave point decimal	oct
cycles per second	cps
Midi note number (0-127)	midinn

The first two forms consist of a whole number, representing octave registration, followed by a specially interpreted fractional part. For *pch*, the fraction is read as two decimal digits representing the 12 equal-tempered pitch classes from .00 for C to .11 for B. For *oct*, the fraction is interpreted as a true decimal fractional part of an octave. The two fractional forms are thus related by the factor 100/12. In both forms, the fraction is preceded by a whole number octave index such that 8.00 represents Middle C, 9.00 the C above, etc. Midi note number values range between 0 and 127 (inclusively) with 60 representing Middle C, and are usually whole numbers. Thus A440 can be represented alternatively by 440 (*cps*), 69 (*midinn*), 8.09 (*pch*), or 8.75 (*oct*). Microtonal divisions of the *pch* semitone can be encoded by using more than two decimal places.

The mnemonics of the pitch conversion units are derived from morphemes of the forms involved, the second morpheme describing the source and the first morpheme the object (result). Thus *cpspch*(8.09) will convert the pitch argument 8.09 to its *cps* (or Hertz) equivalent, giving the value of 440. Since the argument is constant over the duration of the note, this conversion will take place at i-time, before any samples for the current note are produced.

By contrast, the conversion *cpsoct*(8.75 + k1) which gives the value of A440 transposed by the octave interval *k1*. The calculation will be repeated every k-period since that is the rate at which *k1* varies.



Note

The conversion from *pch*, *oct*, or *midinn* into *cps* is not a linear operation but involves an exponential process that could be time-consuming when executed repeatedly. Csound now uses a built-in table lookup to do this efficiently, even at audio rates. Be-

cause the table index is truncated without interpolation, pitch resolution when using one of these opcodes is limited to 8192 discrete and equal divisions of the octave, and some pitches of the standard 12-tone equally-tempered scale are very slightly mistuned (by at most 0.15 cents).

Examples

Here is an example of the `octcps` opcode. It uses the file `octcps.csd` [examples/octcps.csd].

Example 322. Example of the `octcps` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o octcps.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Convert a cycles-per-second value into an
; octave value.
icps = 440
ioct = octcps(icps)

print ioct
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  ioct = 8.750
```

See Also

cpsoct, *cpspch*, *octpch*, *pchoct*, *cpsmidinn*, *octmidinn*, *pchmidinn*

Credits

Example written by Kevin Conder.

octmidi

octmidi — Get the note number, in octave-point-decimal units, of the current MIDI event.

Description

Get the note number, in octave-point-decimal units, of the current MIDI event.

Syntax

iout **octmidi**

Performance

Get the note number of the current MIDI event, expressed in octave-point-decimal units, for local processing.



octmidi vs. octmidinn

The *octmidi* opcode only produces meaningful results in a Midi-activated note (either real-time or from a Midi score with the -F flag). With *octmidi*, the Midi note number value is taken from the Midi event that is internally associated with the instrument instance. On the other hand, the *octmidinn* opcode may be used in any Csound instrument instance whether it is activated from a Midi event, score event, line event, or from another instrument. The input value for *octmidinn* might for example come from a p-field in a textual score or it may have been retrieved from the real-time Midi event that activated the current note using the *notnum* opcode.

Examples

Here is an example of the octmidi opcode. It uses the file *octmidi.csd* [examples/octmidi.csd].

Example 323. Example of the octmidi opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o octmidi.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; This example expects MIDI note inputs on channel 1
i1 octmidi

print i1
endin
```

```
</CsInstruments>
<CsScore>

;Dummy f-table to give time for real-time MIDI events
f 0 8000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

aftouch, ampmidi, cpsmidi, cpsmidib, midictrl, notnum, octmidib, pchbend, pchmidi, pchmidib, veloc, cpsmidinn, octmidinn, pchmidinn

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

octmidib

octmidib — Get the note number of the current MIDI event and modify it by the current pitch-bend value, express it in octave-point-decimal.

Description

Get the note number of the current MIDI event and modify it by the current pitch-bend value, express it in octave-point-decimal.

Syntax

```
ioct octmidib [irange]
```

```
koct octmidib [irange]
```

Initialization

irange (optional) -- the pitch bend range in semitones

Performance

Get the note number of the current MIDI event, modify it by the current pitch-bend value, and express the result in octave-point-decimal units. Available as an i-time value or as a continuous k-rate value.

Examples

Here is an example of the octmidib opcode. It uses the file *octmidib.csd* [examples/octmidib.csd].

Example 324. Example of the octmidib opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o octmidib.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; This example expects MIDI note inputs on channel 1
il octmidib

    print il
endin

</CsInstruments>
```

```
<CsScore>

;Dummy f-table to give time for real-time MIDI events
f 0 8000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

aftouch, ampmidi, cpsmidi, cpsmidib, midictrl, notnum, octmidi, pchbend, pchmidi, pchmidib, veloc

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

octmidinn

octmidinn — Converts a Midi note number value to octave-point-decimal.

Description

Converts a Midi note number value to octave-point-decimal.

Syntax

```
octmidinn (MidiNoteNumber) (init- or control-rate args only)
```

where the argument within the parentheses may be a further expression.

Performance

octmidinn is a function that takes an i-rate or k-rate value representing a Midi note number and returns the equivalent pitch value in Csound's octave-point-decimal format. This conversion assumes that Middle C (8.000 in *oct*) is Midi note number 60. Midi note number values are typically integers in the range from 0 to 127 but fractional values or values outside of this range will be interpreted consistently.



octmidinn vs. octmidi

The *octmidinn* opcode may be used in any Csound instrument instance whether it is activated from a Midi event, score event, line event, or from another instrument. The input value for *octmidinn* might for example come from a p-field in a textual score or it may have been retrieved from the real-time Midi event that activated the current note using the *notnum* opcode. You must specify an i-rate or k-rate expression for the Midi note number that is to be converted. On the other hand, the *octmidi* opcode only produces meaningful results in a Midi-activated note (either real-time or from a Midi score with the -F flag). With *octmidi*, the Midi note number value is taken from the Midi event associated with the instrument instance, and no location or expression for this value may be specified.

octmidinn and its related opcodes are really *value converters* with a special function of manipulating pitch data.

Data concerning pitch and frequency can exist in any of the following forms:

Table 14. Pitch and Frequency Values

Name	Abbreviation
octave point pitch-class (8ve.pc)	pch
octave point decimal	oct
cycles per second	cps
Midi note number (0-127)	midinn

The first two forms consist of a whole number, representing octave registration, followed by a specially interpreted fractional part. For *pch*, the fraction is read as two decimal digits representing the 12 equal-tempered pitch classes from .00 for C to .11 for B. For *oct*, the fraction is interpreted as a true decimal fractional part of an octave. The two fractional forms are thus related by the factor 100/12. In both forms, the fraction is preceded by a whole number octave index such that 8.00 rep-

resents Middle C, 9.00 the C above, etc. Midi note number values range between 0 and 127 (inclusively) with 60 representing Middle C, and are usually whole numbers. Thus A440 can be represented alternatively by 440 (*cps*), 69 (*midinn*), 8.09 (*pch*), or 8.75 (*oct*). Microtonal divisions of the *pch* semitone can be encoded by using more than two decimal places.

The mnemonics of the pitch conversion units are derived from morphemes of the forms involved, the second morpheme describing the source and the first morpheme the object (result). Thus *cpspch*(8.09) will convert the pitch argument 8.09 to its *cps* (or Hertz) equivalent, giving the value of 440. Since the argument is constant over the duration of the note, this conversion will take place at i-time, before any samples for the current note are produced.

By contrast, the conversion *cpsoct*(8.75 + k1) which gives the value of A440 transposed by the octave interval *k1*. The calculation will be repeated every k-period since that is the rate at which *k1* varies.



Note

The conversion from *pch*, *oct*, or *midinn* into *cps* is not a linear operation but involves an exponential process that could be time-consuming when executed repeatedly. Csound now uses a built-in table lookup to do this efficiently, even at audio rates. Because the table index is truncated without interpolation, pitch resolution when using one of these opcodes is limited to 8192 discrete and equal divisions of the octave, and some pitches of the standard 12-tone equally-tempered scale are very slightly mistuned (by at most 0.15 cents).

Examples

Here is an example of the *octmidinn* opcode. It uses the file *cpsmidinn.csd* [examples/cpsmidinn.csd].

Example 325. Example of the *octmidinn* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform.
; This example produces no audio, so we render in
; non-realtime and turn off sound to disk:
-n
</CsOptions>
<CsInstruments>

instr 1
; i-time loop to print conversion table
imidiNN = 0
loop1:
  icps = cpsmidinn(imidiNN)
  ioct = octmidinn(imidiNN)
  ipch = pchmidinn(imidiNN)

  print imidiNN, icps, ioct, ipch

  imidiNN = imidiNN + 1
  if (imidiNN < 128) igoto loop1
endin

instr 2
; test k-rate converters
kMiddleC = 60
kcps = cpsmidinn(kMiddleC)
koct = octmidinn(kMiddleC)
kpch = pchmidinn(kMiddleC)

printks "%d %f %f %f\n", 1.0, kMiddleC, kcps, koct, kpch
endin

</CsInstruments>
```

```
<CsScore>  
i1 0 0  
i2 0 0.1  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

cpsmidinn, *pchmidinn*, *octmidi*, *notnum*, *cpspch*, *cpsoct*, *octcps*, *octpch*, *pchoct*

Credits

Derived from original value converters by Barry Vercoe.

New in version 5.07

octpch

octpch — Converts a pitch-class value to octave-point-decimal.

Description

Converts a pitch-class value to octave-point-decimal.

Syntax

`octpch` (*pch*) (*init-* or *control-rate* args only)

where the argument within the parentheses may be a further expression.

Performance

octpch and its related opcodes are really *value converters* with a special function of manipulating pitch data.

Data concerning pitch and frequency can exist in any of the following forms:

Table 15. Pitch and Frequency Values

Name	Abbreviation
octave point pitch-class (8ve.pc)	pch
octave point decimal	oct
cycles per second	cps
Midi note number (0-127)	midinn

The first two forms consist of a whole number, representing octave registration, followed by a specially interpreted fractional part. For *pch*, the fraction is read as two decimal digits representing the 12 equal-tempered pitch classes from .00 for C to .11 for B. For *oct*, the fraction is interpreted as a true decimal fractional part of an octave. The two fractional forms are thus related by the factor 100/12. In both forms, the fraction is preceded by a whole number octave index such that 8.00 represents Middle C, 9.00 the C above, etc. Midi note number values range between 0 and 127 (inclusively) with 60 representing Middle C, and are usually whole numbers. Thus A440 can be represented alternatively by 440 (*cps*), 69 (*midinn*), 8.09 (*pch*), or 8.75 (*oct*). Microtonal divisions of the *pch* semitone can be encoded by using more than two decimal places.

The mnemonics of the pitch conversion units are derived from morphemes of the forms involved, the second morpheme describing the source and the first morpheme the object (result). Thus *cpspch*(8.09) will convert the pitch argument 8.09 to its *cps* (or Hertz) equivalent, giving the value of 440. Since the argument is constant over the duration of the note, this conversion will take place at i-time, before any samples for the current note are produced.

By contrast, the conversion *cpsoct*(8.75 + *k1*) which gives the value of A440 transposed by the octave interval *k1*. The calculation will be repeated every k-period since that is the rate at which *k1* varies.



Note

The conversion from *pch*, *oct*, or *midinn* into *cps* is not a linear operation but involves an exponential process that could be time-consuming when executed repeatedly. Csound now uses a built-in table lookup to do this efficiently, even at audio rates. Be-

cause the table index is truncated without interpolation, pitch resolution when using one of these opcodes is limited to 8192 discrete and equal divisions of the octave, and some pitches of the standard 12-tone equally-tempered scale are very slightly mistuned (by at most 0.15 cents).

Examples

Here is an example of the `octpch` opcode. It uses the file `octpch.csd` [examples/octpch.csd].

Example 326. Example of the `octpch` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o octpch.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Convert a pitch-class value into an
; octave-point-decimal value.
ipch = 8.09
ioct = octpch(ipch)

print ioct
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  ioct = 8.750
```

See Also

cpsoct, *cpspch*, *octcps*, *pchoct*, *cpsmidinn*, *octmidinn*, *pchmidinn*

Credits

Example written by Kevin Conder.

opcode

opcode — Defines the start of user-defined opcode block.

Defining opcodes

The *opcode* and *endop* statements allow defining a new opcode that can be used the same way as any of the built-in Csound opcodes. These opcode blocks are very similar to instruments (and are, in fact, implemented as special instruments), but cannot be called as a normal instrument e.g. with the *i* statements.

A user-defined opcode block must precede the instrument (or other opcode) from which it is used. But it is possible to call the opcode from itself. This allows recursion of any depth that is limited only by available memory. Additionally, there is an experimental feature that allows running the opcode definition at a higher control rate than the *kr* value specified in the orchestra header.

Similarly to instruments, the variables and labels of a user-defined opcode block are local and cannot be accessed from the caller instrument (and the opcode cannot access variables of the caller, either).

Some parameters are automatically copied at initialization, however:

- all p-fields (including *p1*)
- extra time (see also *xtratim*, *linsegr*, and related opcodes). This may affect the operation of *linsegr/expsegr/linenr/envlpxr* in the user-defined opcode block.
- MIDI parameters, if there are any.

Also, the release flag (see the *release* opcode) is copied at performance time.

Modifying the note duration in the opcode definition by assigning to *p3*, or using *ihold*, *turnoff*, *xtratim*, *linsegr*, or similar opcodes will also affect the caller instrument. Changes to MIDI controllers (for example with *ctrlinit*) will also apply to the instrument from which the opcode was called.

Use the *setksmps* opcode to set the local *ksmps* value.

The *xin* and *xout* opcodes copy variables to and from the opcode definition, allowing communication with the calling instrument.

The types of input and output variables are defined by the parameters *intypes* and *outtypes*.



Notes

- *xin* and *xout* should be called only once, and *xin* should precede *xout*, otherwise an init error and deactivation of the current instrument may occur.
- These opcodes actually run only at i-time. Performance time copying is done by the user opcode call. This means that skipping *xin* or *xout* with *kgoto* has no effect, while skipping with *igoto* affects both init and performance time operation.

Syntax

```
opcode name, outtypes, intypes
```

Initialization

name -- name of the opcode. It may consist of any combination of letters, digits, and underscore but should not begin with a digit. If an opcode with the specified name already exists, it is redefined (a warning is printed in such cases). Some reserved words (like *instr* and *endin*) cannot be redefined.

intypes -- list of input types, any combination of the characters: a, k, K, i, o, p, and j. A single 0 character can be used if there are no input arguments. Double quotes and delimiter characters (e.g. comma) are *not* needed.

The meaning of the various *intypes* is shown in the following table:

Type	Description	Variable Types Al- lowed	Updated At
a	a-rate variable	a-rate	a-rate
i	i-rate variable	i-rate	i-time
j	optional i-time, defaults to -1	i-rate, constant	i-time
k	k-rate variable	k- and i-rate, constant	k-rate
K	k-rate with initialization	k- and i-rate, constant	i-time and k-rate
o	optional i-time, defaults to 0	i-rate, constant	i-time
p	optional i-time, defaults to 1	i-rate, constant	i-time
S	string variable	i-rate string	i-time

The maximum allowed number of input arguments is 256.

outtypes -- list of output types. The format is the same as in the case of *intypes*.

Here are the available *outtypes*:

Type	Description	Variable Types Al- lowed	Updated At
a	a-rate variable	a-rate	a-rate
i	i-rate variable	i-rate	i-time
k	k-rate variable	k-rate	k-rate
K	k-rate with initialization	k-rate	i-time and k-rate

The maximum allowed number of output arguments is 256.

ksmps (optional, default=0) -- sets the local *ksmps* value. Must be a positive integer, and also the *ksmps* of the calling instrument or opcode must be an integer multiple of this value. For example, if *ksmps* is 10 in the instrument from which the opcode was called, the allowed values for *iksmips* are 1, 2, 5, and 10.

If *iksmips* is set to zero, the *ksmps* of the caller instrument or opcode is used (this is the default behavior).



Note

The local *ksmps* is implemented by splitting up a control period into smaller subperiods and temporarily modifying internal Csound global variables. This also requires converting the rate of k-rate input and output arguments (input variables receive

the same value in all sub-kperiods, while outputs are written only in the last one).



Warning about local *ksmps*

When the local *ksmps* is not the same as the orchestra level *ksmps* value (as specified in the orchestra header), global a-rate operations must not be used in the user-defined opcode block.

These include:

- any access to “ga” variables
- a-rate zak opcodes (*zar*, *zaw*, etc.)
- *tablera* and *tablewa* (these two opcodes may in fact work, but caution is needed)
- The *in* and *out* opcode family (these read from, and write to global a-rate buffers)

In general, the local *ksmps* should be used with care as it is an experimental feature, although it works correctly in most cases.

The *setksmps* statement can be used to set the local *ksmps* value of the user-defined opcode block. It has one i-time parameter specifying the new *ksmps* value (which is left unchanged if zero is used, see also the notes about *iksmpts* above). *setksmps* should be used before any other opcodes (but allowed after *xin*), otherwise unpredictable results may occur.

The input parameters can be read with *xin*, and the output is written by *xout* opcode. Only one instance of these units should be used, as *xout* overwrites and does not accumulate the output. The number and type of arguments for *xin* and *xout* must be the same as in the declaration of the user-defined opcode block (see tables above).

The input and output arguments must agree with the definition both in number (except if the optional i-time input is used) and type. An optional i-time input parameter (*iksmpts*) is automatically added to the *intypes* list, and (similarly to *setksmps*) sets the local *ksmps* value.

Performance

The syntax of a user-defined opcode block is as follows:

```
opcode name, outtypes, intypes
xinarg1 [, xinarg2] [, xinarg3] ... [xinargN] xin
[setksmps iksmpts]
... the rest of the instrument's code.
xout xoutarg1 [, xoutarg2] [, xoutarg3] ... [xoutargN]
endop
```

The new opcode can then be used with the usual syntax:

```
[xoutarg1] [, xoutarg2] ... [xoutargN] name [xinarg1] [, xinarg2] ... [xinargN] [, iksmpts]
```



Note

The opcode call is always executed both at initialization and performance time, even if there are no a- or k-rate arguments. If there are many user opcode calls that are known to have no effect at performance time in an instrument, then it may save some CPU time to jump over groups of such opcodes with *kgoto*.

Examples

Here is an example of a user-defined opcode. It uses the file *opcode.csd* [examples/opcode_example.csd].

Example 327. Example of a user-defined opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o opcode_example.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr      = 44100
ksmps   = 50
nchnls  = 1

/* example opcode 1: simple oscillator */

opcode Oscillator, a, kk

kamp, kcps      xin          ; read input parameters
a1      vco2 kamp, kcps      ; sawtooth oscillator
xout a1          ; write output

endop

/* example opcode 2: lowpass filter with local ksmps */

opcode Lowpass, a, akk

setksmps 1          ; need sr=kr
ain, kal, ka2      xin          ; read input parameters
aout      init 0          ; initialize output
aout      = ain*kal + aout*ka2 ; simple tone-like filter
xout aout          ; write output

endop

/* example opcode 3: recursive call */

opcode RecursiveLowpass, a, akkpp

ain, kal, ka2, idep, icnt      xin          ; read input parameters
if (icnt >= idep) goto skip1    ; check if max depth reached
ain      RecursiveLowpass ain, kal, ka2, idep, icnt + 1
skip1:
aout      Lowpass ain, kal, ka2          ; call filter
xout aout          ; write output

endop

/* example opcode 4: de-click envelope */

opcode DeClick, a, a

ain      xin
aenv      linseg 0, 0.02, 1, p3 - 0.05, 1, 0.02, 0, 0.01, 0
xout ain * aenv          ; apply envelope and write output

endop

/* instr 1 uses the example opcodes */

instr 1

kamp      = 20000          ; amplitude
kcps      expon 50, p3, 500 ; pitch
a1      Oscillator kamp, kcps          ; call oscillator
kflt      linseg 0.4, 1.5, 0.4, 1, 0.8, 1.5, 0.8 ; filter envelope
a1      RecursiveLowpass a1, kflt, 1 - kflt, 10 ; 10th order lowpass
a1      DeClick a1
out a1
```

```
        endin

</CsInstruments>
<CsScore>

i 1 0 4
e

</CsScore>
</CsoundSynthesizer>
```

See Also

endop, *setksmps*, *xin*, *xout*

Credits

Author: Istvan Varga, 2002; based on code by Matt J. Ingalls

New in version 4.22

OSCsend

OSCsend — Sends data to other processes using the OSC protocol

Description

Uses the OSC protocol to send message to other OSC listening processes.

Syntax

```
OSCsend kwhen, ihost, iport, idestination, itype [, kdata1, kdata2, ...]
```

Initialization

ihost -- a string that is the intended host computer domain name. An empty string is interpreted as the current computer.

iport -- the number of the port that is used for the communication.

idest -- a string that is the destination address. This takes the form of a file name with directories. Csound just passes this string to the raw sending code and makes no interpretation.

itype -- a string that indicates the types of the optional arguments that are read at k-rate. The string can contain the characters "bcdfilmst" which stand for Boolean, character, double, float, 32-bit integer, 64-bit integer, MIDI, string and timestamp.

Performance

kwhen -- a message is sent whenever this value changes. A message will always be sent on the first call.

The data is taken from the k-values that follow the format string. In a similar way to a printf format, the characters in order determine how the argument is interpreted. Note that a time stamp takes two arguments.

Example

The example shows a simple instrument, which when called, sends a group of 3 messages to a computer called "xenakis", on port 7770, to be read by a process that recognises /foo/bar as its address.

```
instr      1
  OSCsend  1, "xenakis.cs.bath.ac.uk", 7770, "/foo/bar", "sis", "FOO", 42, "bar"
endin
```

See the entry for *OSClisten*, for an example of send/recieve usage using OSC.

See Also

OSClisten, *OSCinit*

Credits

Author: John ffitch
2005

OSCinit

OSCinit — Start a listening process for OSC messages to a particular port.

Description

Starts a listening process, which can be used by OSClisten.

Syntax

```
ihandle OSCinit iport
```

Initialization

ihandle -- handle returned that can be passed to any number of OSClisten opcodes to receive messages on this port.

iport -- the port on which to listen.

Performance

The listener runs in the background. See OSClisten for details.

Example

The example shows a pair of floating point numbers being received on port 7770.

```
sr = 44100
ksmps = 100
nchnls = 2

gihandle OSCinit 7770

instr 1
  kf1 init 0
  kf2 init 0
nextmsg:
  kk OSClisten gihandle, "/foo/bar", "ff", kf1, kf2
if (kk == 0) goto ex
  printk 0,kf1
  printk 0,kf2
  kgoto nextmsg
ex:
endin
```

Credits

Author: John fitch
2005

OSClisten

OSClisten — Listen for OSC messages to a particular path.

Description

On each k-cycle looks to see if an OSC message has been send to a given path of a given type.

Syntax

```
kans OSClisten ihandle, idest, itype [, xdata1, xdata2, ...]
```

Initialization

ihandle -- a handle returned by an earlier call to OSCinit, to associate OSClisten with a particular port number.

idest -- a string that is the destination address. This takes the form of a file name with directories. Csound uses this address to decide if messages are meant for csound.

itype -- a string that indicates the types of the optional arguments that are to be read. The string can contain the characters "cdfhis" which stand for character, double, float, 64-bit integer, 32-bit integer, and string. All types other than 's' require a k-rate variable, while 's' requires a string variable.

A handler is inserted into the listener (see OSCinit) to intercept messages of this pattern.

Performance

kans -- set to 1 if a new message was received, or zero if not. If multiple messages are received in a single control period, the messages are buffered, and OSClisten can be called again until zero is returned.

If there was a message the *xdata* variables are set to the incoming values, as interpreted by the *itype* parameter. Note that although the *xdata* variables are on the right of an operation they are actually outputs, and so must be variables of type k, gk, S, or gS, and may need to be declared with init, or = in the case of string variables, before calling OSClisten.

Example

The example shows a pair of floating point numbers being received on port 7770.

```
sr = 44100
ksmps = 100
nchnls = 2

gihandle OSCinit 7770

instr 1
  kf1 init 0
  kf2 init 0
nxtmsg:
  kk OSClisten gihandle, "/foo/bar", "ff", kf1, kf2
if (kk == 0) goto ex
  printk 0,kf1
  printk 0,kf2
  kgoto nxtmsg
ex:
endin
```

Below are two .csd files which demonstrate the usage of the OSC opcodes. They use the files *OSCmidisend.csd* [examples/OSCmidisend.csd] and *OSCmidircv.csd* [examples/OSCmidircv.csd].

Example 328. Example of the OSC opcodes.

The following two .csd files demonstrate the usage of the OSC opcodes in csound. The first file, *OSCmidisend.csd* [examples/OSCmidisend.csd], transforms received real-time MIDI messages into OSC data. The second file, *OSCmidircv.csd* [examples/OSCmidircv.csd], can take these OSC messages, and interpret them to generate sound from note messages, and store controller values. It will use controller number 7 to control volume. Note that these files are designed to be on the same machine, but if a different host address (in the IPADDRESS macro) is used, they can be separate machines on a network, or connected through the internet.

CSD file to send OSC messages:

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
</CsOptions>
<CsInstruments>

    sr      = 44100
    ksmpps  = 128
    nchnls  = 1

; Example by David Akbari 2007
; Modified by Jonathan Murphy
; Use this file to generate OSC events for OSCmidircv.csd

#define IPADDRESS # "localhost" #
#define PORT      # 47120 #

turnon 1000

    instr 1000

    kst, kch, kdl, kd2  midiin

    OSCsend    kst+kch+kdl+kd2, $IPADDRESS, $PORT, "/midi", "iiii", kst, kch, kdl, kd2

    endin

</CsInstruments>
<CsScore>
f 0 3600 ;Dummy f-table
e
</CsScore>
</CsoundSynthesizer>
```

CSD file to receive OSC messages:

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
</CsOptions>
<CsInstruments>

    sr      = 44100
    ksmpps  = 128
    nchnls  = 1

; Example by Jonathan Murphy and Andres Cabrera 2007
; Use file OSCmidisend.csd to generate OSC events for this file

    odbfs    = 1

    gilisten  OSCinit    47120

    gisin     ftgen      1, 0, 16384, 10, 1
```

```
givel      ftgen      2, 0, 128, -2, 0
gicc      ftgen      3, 0, 128, -7, 100, 128, 100 ;Default all controllers to 100

;Define scale tuning
giji_12    ftgen      202, 0, 32, -2, 12, 2, 256, 60, 1, 16/15, 9/8, 6/5, 5/4, 4/3, 7/5, \
              3/2, 8/5, 5/3, 9/5, 15/8, 2

#define DEST #"/midi"#
; Use controller number 7 for volume
#define VOL #7#

turnon 1000

instr 1000

kst      init      0
kch      init      0
kd1      init      0
kd2      init      0

next:

kk      OSClisten gilisten, $DEST, "iiii", kst, kch, kd1, kd2

if (kk == 0) goto done

printks "kst = %i, kch = %i, kd1 = %i, kd2 = %i\\n", \
0, kst, kch, kd1, kd2

if (kst == 176) then
;Store controller information in a table
tablew    kd2, kd1, gicc
endif

if (kst == 144) then
;Process noteon and noteoff messages.
kkey      = kd1
kvel      = kd2
kcps      cpstun    kvel, kkey, giji_12
kamp      = kvel/127

if (kvel == 0) then
turnoff2  1001, 4, 1
elseif (kvel > 0) then
event     "i", 1001, 0, -1, kcps, kamp
endif
endif

kgoto next ;Process all events in queue

done:
endin

instr 1001 ;Simple instrument

icps      init      p4
kvol      table     $VOL, gicc ;Read MIDI volume from controller table
kvol      = kvol/127

aenv      linsegr    0, .003, p5, 0.03, p5 * 0.5, 0.3, 0
aosc      oscil      aenv, icps, gisin

out       aosc * kvol

endin

</CsInstruments>
<CsScore>
f 0 3600 ;Dummy f-table
e
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch

2005

Examples by: David Akbari, Andrés Cabrera and Jonathan Murphy 2007

oscbnk

oscbnk — Mixes the output of any number of oscillators.

Description

This unit generator mixes the output of any number of oscillators. The frequency, phase, and amplitude of each oscillator can be modulated by two LFOs (all oscillators have a separate set of LFOs, with different phase and frequency); additionally, the output of each oscillator can be filtered through an optional parametric equalizer (also controlled by the LFOs). This opcode is most useful for rendering ensemble (strings, choir, etc.) instruments.

Although the LFOs run at k-rate, amplitude, phase and filter modulation are interpolated internally, so it is possible (and recommended in most cases) to use this unit at low (~1000 Hz) control rates without audible quality degradation.

The start phase and frequency of all oscillators and LFOs can be set by a built-in seedable 31-bit random number generator, or specified manually in a function table (GEN2).

Syntax

```
ares oscbnk kcps, kamd, kfmd, kpmd, iovrlap, iseed, kl1minf, kl1maxf, \
      kl2minf, kl2maxf, ilfomode, keqminf, keqmaxf, keqminl, keqmaxl, \
      keqming, keqmaxg, ieqmode, kfn [, il1fn] [, il2fn] [, ieqffn] \
      [, ieqlfn] [, ieqqfn] [, itabl] [, ioutfn]
```

Initialization

iovrlap -- Number of oscillator units.

iseed -- Seed value for random number generator (positive integer in the range 1 to 2147483646 ($2^{31} - 2$)). *iseed* <= 0 seeds from the current time.

ieqmode -- Parametric equalizer mode

- -1: disable EQ (faster)
- 0: peak
- 1: low shelf
- 2: high shelf
- 3: peak (filter interpolation disabled)
- 4: low shelf (interpolation disabled)
- 5: high shelf (interpolation disabled)

The non-interpolated modes are faster, and in some cases (e.g. high shelf filter at low cutoff frequencies) also more stable; however, interpolation is useful for avoiding “zipper noise” at low control rates.

ilfomode -- LFO modulation mode, sum of:

- 128: LFO1 to frequency
- 64: LFO1 to amplitude

- 32: LFO1 to phase
- 16: LFO1 to EQ
- 8: LFO2 to frequency
- 4: LFO2 to amplitude
- 2: LFO2 to phase
- 1: LFO2 to EQ

If an LFO does not modulate anything, it is not calculated, and the *ftable* number (*il1fn* or *il2fn*) can be omitted.

il1fn (optional: default=0) -- LFO1 function table number. The waveform in this table has to be normalized (absolute value ≤ 1), and is read with linear interpolation.

il2fn (optional: default=0) -- LFO2 function table number. The waveform in this table has to be normalized, and is read with linear interpolation.

ieqffn, *ieqlfn*, *ieqqfn* (optional: default=0) -- Lookup tables for EQ frequency, level, and Q (optional if EQ is disabled). Table read position is 0 if the modulator signal is less than, or equal to -1, (table length / 2) if the modulator signal is zero, and the guard point if the modulator signal is greater than, or equal to 1. These tables have to be normalized to the range 0 - 1, and have an extended guard point (table length = power of two + 1). All tables are read with linear interpolation.

itabl (optional: default=0) -- Function table storing phase and frequency values for all oscillators (optional). The values in this table are in the following order (5 for each oscillator unit):

oscillator phase, lfo1 phase, lfo1 frequency, lfo2 phase, lfo2 frequency, ...

All values are in the range 0 to 1; if the specified number is greater than 1, it is wrapped (phase) or limited (frequency) to the allowed range. A negative value (or end of table) will use the output of the random number generator. The random seed is always updated (even if no random number was used), so switching one value between random and fixed will not change others.

ioutfn (optional: default=0) -- Function table to write phase and frequency values (optional). The format is the same as in the case of *itabl*. This table is useful when experimenting with random numbers to record the best values.

The two optional tables (*itabl* and *ioutfn*) are accessed only at i-time. This is useful to know, as the tables can be safely overwritten after opcode initialization, which allows precalculating parameters at i-time and storing in a temporary table before *oscbnk* initialization.

Performance

ares -- Output signal.

kcps -- Oscillator frequency in Hz.

kamd -- AM depth (0 - 1).

(AM output) = (AM input) * ((1 - (AM depth)) + (AM depth) * (modulator))

If *ilfomode* isn't set to modulate the amplitude, then (AM output) = (AM input) regardless of the value of *kamd*. That means that *kamd* will have no effect.

Note: Amplitude modulation is applied before the parametric equalizer.

kfmd -- FM depth (in Hz).

kpmd -- Phase modulation depth.

kl1minf, *kl1maxf* -- LFO1 minimum and maximum frequency in Hz.

kl2minf, *kl2maxf* -- LFO2 minimum and maximum frequency in Hz. (Note: oscillator and LFO frequencies are allowed to be zero or negative.)

keqminf, *keqmaxf* -- Parametric equalizer minimum and maximum frequency in Hz.

keqminl, *keqmaxl* -- Parametric equalizer minimum and maximum level.

keqminq, *keqmaxq* -- Parametric equalizer minimum and maximum Q.

kfn -- Oscillator waveform table. Table number can be changed at k-rate (this is useful to select from a set of band-limited tables generated by GEN30, to avoid aliasing). The table is read with linear interpolation.



Note

oscblk uses the same random number generator as *rnd31*. So reading *its documentation* is also recommended.

Examples

Here is an example of *oscblk* opcode. It uses the file *oscblk.csd* [examples/oscblk.csd].

Example 329. Example of the *oscblk* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o oscblk.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Istvan Varga */
sr = 48000
kr = 750
ksmps = 64
nchnls = 2

ga01 init 0
ga02 init 0

/* sawtooth wave */
i_ ftgen 1, 0, 16384, 7, 1, 16384, -1
/* FM waveform */
i_ ftgen 3, 0, 4096, 7, 0, 512, 0.25, 512, 1, 512, 0.25, 512, \
    0, 512, -0.25, 512, -1, 512, -0.25, 512, 0

/* AM waveform */
i_ ftgen 4, 0, 4096, 5, 1, 4096, 0.01
/* FM to EQ */
i_ ftgen 5, 0, 1024, 5, 1, 512, 32, 512, 1
/* sine wave */
i_ ftgen 6, 0, 1024, 10, 1
/* room parameters */
i_ ftgen 7, 0, 64, -2, 4, 50, -1, -1, -1, 11, \
    1, 26.833, 0.05, 0.85, 10000, 0.8, 0.5, 2, \
    1, 1.753, 0.05, 0.85, 5000, 0.8, 0.5, 2, \
    1, 39.451, 0.05, 0.85, 7000, 0.8, 0.5, 2, \
    1, 33.503, 0.05, 0.85, 7000, 0.8, 0.5, 2, \
```

```
1, 36.151, 0.05, 0.85, 7000, 0.8, 0.5, 2, \
1, 29.633, 0.05, 0.85, 7000, 0.8, 0.5, 2

/* generate bandlimited sawtooth waves */

i0 = 0
loop1:
imaxh = sr / (2 * 440.0 * exp(log(2.0) * (i0 - 69) / 12))
i_ftgen i0 + 256, 0, 4096, -30, 1, 1, imaxh
i0 = i0 + 1
    if (i0 < 127.5) igoto loop1

instr 1

p3 = p3 + 0.4

; note frequency
kcps = 440.0 * exp(log(2.0) * (p4 - 69) / 12)
; lowpass max. frequency
klpmaxf limit 64 * kcps, 1000.0, 12000.0
; FM depth in Hz
kfmdl = 0.02 * kcps
; AM frequency
kamfr = kcps * 0.02
kamfr2 = kcps * 0.1
; table number
kfnum = (256 + 69 + 0.5 + 12 * log(kcps / 440.0) / log(2.0))
; amp. envelope
aenv linseg 0, 0.1, 1.0, p3 - 0.5, 1.0, 0.1, 0.5, 0.2, 0, 1.0, 0

/* oscillator / left */

a1 oscbnk kcps, 0.0, kfmdl, 0.0, 40, 200, 0.1, 0.2, 0, 0, 144, \
    0.0, klpmaxf, 0.0, 0.0, 1.5, 1.5, 2, \
    kfnum, 3, 0, 5, 5, 5
a2 oscbnk kcps, 1.0, kfmdl, 0.0, 40, 201, 0.1, 0.2, kamfr, kamfr2, 148, \
    0, 0, 0, 0, 0, 0, -1, \
    kfnum, 3, 4
a2 pareq a2, kcps * 8, 0.0, 0.7071, 2
a0 = a1 + a2 * 0.12
/* delay */
adel = 0.001
a01 vdelayx a0, adel, 0.01, 16
a_ oscili 1.0, 0.25, 6, 0.0
adel = adel + 1.0 / (exp(log(2.0) * a_) * 8000)
a02 vdelayx a0, adel, 0.01, 16
a0 = a01 + a02

ga01 = ga01 + a0 * aenv * 2500

/* oscillator / right */

; lowpass max. frequency

a1 oscbnk kcps, 0.0, kfmdl, 0.0, 40, 202, 0.1, 0.2, 0, 0, 144, \
    0.0, klpmaxf, 0.0, 0.0, 1.0, 1.0, 2, \
    kfnum, 3, 0, 5, 5, 5
a2 oscbnk kcps, 1.0, kfmdl, 0.0, 40, 203, 0.1, 0.2, kamfr, kamfr2, 148, \
    0, 0, 0, 0, 0, 0, -1, \
    kfnum, 3, 4
a2 pareq a2, kcps * 8, 0.0, 0.7071, 2
a0 = a1 + a2 * 0.12
/* delay */
adel = 0.001
a01 vdelayx a0, adel, 0.01, 16
a_ oscili 1.0, 0.25, 6, 0.25
adel = adel + 1.0 / (exp(log(2.0) * a_) * 8000)
a02 vdelayx a0, adel, 0.01, 16
a0 = a01 + a02

ga02 = ga02 + a0 * aenv * 2500

endin

/* output / left */

instr 81

i1 = 0.000001
aL1, aLh, aR1, aRh spat3di ga01 + i1*i1*i1*i1, -8.0, 4.0, 0.0, 0.3, 7, 4
ga01 = 0
aL1 butterlp aL1, 800.0
aR1 butterlp aR1, 800.0

outs aL1 + aLh, aR1 + aRh

endin
```

```
/* output / right */

    instr 82

i1 = 0.000001
aLl, aLh, aRl, aRh spat3di ga02 + i1*i1*i1*i1, 8.0, 4.0, 0.0, 0.3, 7, 4
ga02 = 0
aLl butterlp aLl, 800.0
aRl butterlp aRl, 800.0

    outs aLl + aLh, aRl + aRh

    endin

</CsInstruments>
<CsScore>

/* Written by Istvan Varga */
t 0 60

i 1 0 4 41
i 1 0 4 60
i 1 0 4 65
i 1 0 4 69

i 81 0 5.5
i 82 0 5.5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Istvan Varga
2001

New in version 4.15

Updated April 2002 by Istvan Varga

oscil

oscil — A simple oscillator.

Description

Table *ifn* is incrementally sampled modulo the table length and the value obtained is multiplied by *amp*.

Syntax

```
ares oscil xamp, xcps, ifn [, iphs]
```

```
kres oscil kamp, kcps, ifn [, iphs]
```

Initialization

ifn -- function table number. Requires a wrap-around guard point.

iphs (optional, default=0) -- initial phase of sampling, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is 0.

Performance

kamp, *xamp* -- amplitude

kcps, *xcps* -- frequency in cycles per second.

The *oscil* opcode generates periodic control (or audio) signals consisting of the value of *kamp* (*xamp*) times the value returned from control rate (audio rate) sampling of a stored function table. The internal phase is simultaneously advanced in accordance with the *kcps* or *xcps* input value.

Examples

Here is an example of the *oscil* opcode. It uses the file *oscil.csd* [examples/oscil.csd].

Example 330. Example of the *oscil* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o oscil.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
ksmps = 128
nchnls = 1

; Instrument #1 - a basic oscillator.
instr 1
  kamp = 10000
```

```
kcps = 440
ifn = 1

al oscil kamp, kcps, ifn
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

oscili, oscil3

Credits

Example written by Kevin Conder.

oscil1

oscil1 — Accesses table values by incremental sampling.

Description

Accesses table values by incremental sampling.

Syntax

```
kres oscil1 idel, kamp, idur, ifn
```

Initialization

idel -- delay in seconds before *oscil1* incremental sampling begins.

idur -- duration in seconds to sample through the *oscil1* table just once. A zero or negative value will cause all initialization to be skipped.

ifn -- function table number. *tablei*, *oscil1i* require the extended guard point.

Performance

kamp -- amplitude factor.

oscil1 accesses values by sampling once through the function table at a rate determined by *idur*. For the first *idel* seconds, the point of scan will reside at the first location of the table; it will then begin moving through the table at a constant rate, reaching the end in another *idur* seconds; from that time on (i.e. after *idel* + *idur* seconds) it will remain pointing at the last location. Each value obtained from sampling is then multiplied by an amplitude factor *kamp* before being written into the result.

See Also

table, *tablei*, *table3*, *oscil1i*, *osciln*

oscil1i

oscil1i — Accesses table values by incremental sampling with linear interpolation.

Description

Accesses table values by incremental sampling with linear interpolation.

Syntax

```
kres oscil1i idel, kamp, idur, ifn
```

Initialization

idel -- delay in seconds before *oscil1* incremental sampling begins.

idur -- duration in seconds to sample through the *oscil1* table just once. A zero or negative value will cause all initialization to be skipped.

ifn -- function table number. *oscil1i* requires the extended guard point.

Performance

kamp -- amplitude factor

oscil1i is an interpolating unit in which the fractional part of index is used to interpolate between adjacent table entries. The smoothness gained by interpolation is at some small cost in execution time (see also *oscili*, etc.), but the interpolating and non-interpolating units are otherwise interchangeable.

See Also

table, *tablei*, *table3*, *oscil1*, *osciln*

oscil3

oscil3 — A simple oscillator with cubic interpolation.

Description

Table *ifn* is incrementally sampled modulo the table length and the value obtained is multiplied by *amp*.

Syntax

```
ares oscil3 xamp, xcps, ifn [, iphs]
```

```
kres oscil3 kamp, kcps, ifn [, iphs]
```

Initialization

ifn -- function table number. Requires a wrap-around guard point.

iphs (optional) -- initial phase of sampling, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is 0.

Performance

kamp, *xamp* -- amplitude

kcps, *xcps* -- frequency in cycles per second.

oscil3 is experimental, and is identical to *oscili*, except that it uses cubic interpolation. (New in Csound version 3.50.)

Examples

Here is an example of the oscil3 opcode. It uses the file *oscil3.csd* [examples/oscil3.csd].

Example 331. Example of the oscil3 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o oscil3.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a basic oscillator.
instr 1
  kamp = 10000
  kcps = 220
```

```
    ifn = 1

    al oscil kamp, kcps, ifn
    out al
endin

; Instrument #2 - the basic oscillator with cubic interpolation.
instr 2
    kamp = 10000
    kcps = 220
    ifn = 1

    al oscil3 kamp, kcps, ifn
    out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave table with a small amount of data.
f 1 0 32 10 0 1

; Play Instrument #1, the basic oscillator, for
; two seconds. This should sound relatively rough.
i 1 0 2

; Play Instrument #2, the cubic interpolated oscillator, for
; two seconds. This should sound relatively smooth.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

oscil, oscili

Credits

Author: John ffitch

Example written by Kevin Conder.

oscili

oscili — A simple oscillator with linear interpolation.

Description

Table *ifn* is incrementally sampled modulo the table length and the value obtained is multiplied by *amp*.

Syntax

```
ares oscili xamp, xcps, ifn [, iphs]
```

```
kres oscili kamp, kcps, ifn [, iphs]
```

Initialization

ifn -- function table number. Requires a wrap-around guard point.

iphs (optional) -- initial phase of sampling, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is 0.

Performance

kamp, *xamp* -- amplitude

kcps, *xcps* -- frequency in cycles per second.

oscili differs from *oscil* in that the standard procedure of using a truncated phase as a sampling index is here replaced by a process that interpolates between two successive lookups. Interpolating generators will produce a noticeably cleaner output signal, but they may take as much as twice as long to run. Adequate accuracy can also be gained without the time cost of interpolation by using large stored function tables of 2K, 4K or 8K points if the space is available.

Examples

Here is an example of the *oscili* opcode. It uses the file *oscili.csd* [examples/oscili.csd].

Example 332. Example of the *oscili* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o oscili.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

```
; Instrument #1 - a basic oscillator.
instr 1
  kamp = 10000
  kcps = 220
  ifn = 1

  al oscil kamp, kcps, ifn
  out al
endin

; Instrument #2 - the basic oscillator with extra interpolation.
instr 2
  kamp = 10000
  kcps = 220
  ifn = 1

  al oscili kamp, kcps, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave table with a small amount of data.
f 1 0 32 10 0 1

; Play Instrument #1, the basic oscillator, for
; two seconds. This should sound relatively rough.
i 1 0 2

; Play Instrument #2, the interpolated oscillator, for
; two seconds. This should sound relatively smooth.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

oscil, oscil3

Credits

Example written by Kevin Conder.

oscilikt

oscilikt — A linearly interpolated oscillator that allows changing the table number at k-rate.

Description

oscilikt is very similar to *oscili*, but allows changing the table number at k-rate. It is slightly slower than *oscili* (especially with high control rate), although also more accurate as it uses a 31-bit phase accumulator, as opposed to the 24-bit one used by *oscili*.

Syntax

```
ares oscilikt xamp, xcps, kfn [, iphs] [, istor]
```

```
kres oscilikt kamp, kcps, kfn [, iphs] [, istor]
```

Initialization

iphs (optional, defaults to 0) -- initial phase in the range 0 to 1. Other values are wrapped to the allowed range.

istor (optional, defaults to 0) -- skip initialization.

Performance

kamp, *xamp* -- amplitude.

kcps, *xcps* -- frequency in Hz. Zero and negative values are allowed. However, the absolute value must be less than *sr* (and recommended to be less than *sr*/2).

kfn -- function table number. Can be varied at control rate (useful to “morph” waveforms, or select from a set of band-limited tables generated by *GEN30*).

Examples

Here is an example of the *oscilikt* opcode. It uses the file *oscilikt.csd* [examples/oscilikt.csd].

Example 333. Example of the *oscilikt* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o oscilikt.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
```

```
instr 1
; Generate a uni-polar (0-1) square wave.
kamp1 init 1
kcps1 init 2
itype = 3
ksquare lfo kamp1, kcps1, itype

; Use the square wave to switch between Tables #1 and #2.
kamp2 init 20000
kcps2 init 220
kfn = ksquare + 1

a1 oscilikt kamp2, kcps2, kfn
out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine waveform.
f 1 0 4096 10 0 1
; Table #2: a sawtooth wave
f 2 0 3 -2 1 0 -1

; Play Instrument #1 for two seconds.
i 1 0 2

</CsScore>
</CsoundSynthesizer>
```

See Also

osciliktp and *oscilikts*.

Credits

Author: Istvan Varga

Example written by Kevin Conder.

New in version 4.22

oscilikt

oscilikt — A linearly interpolated oscillator that allows allows phase modulation.

Description

oscilikt allows phase modulation (which is actually implemented as k-rate frequency modulation, by differentiating phase input). The disadvantage is that there is no amplitude control, and frequency can be varied only at the control-rate. This opcode can be faster or slower than *oscilikt*, depending on the control-rate.

Syntax

```
ares oscilikt kcps, kfn, kphs [, istor]
```

Initialization

istor (optional, defaults to 0) -- Skips initialization.

Performance

ares -- audio-rate output signal.

kcps -- frequency in Hz. Zero and negative values are allowed. However, the absolute value must be less than *sr* (and recommended to be less than *sr*/2).

kfn -- function table number. Can be varied at control rate (useful to “morph” waveforms, or select from a set of band-limited tables generated by *GEN30*).

kphs -- phase (k-rate), the expected range is 0 to 1. The absolute value of the difference of the current and previous value of *kphs* must be less than *ksmps*.

Examples

Here is an example of the oscilikt opcode. It uses the file *oscilikt.csd* [examples/oscilikt.csd].

Example 334. Example of the oscilikt opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o oscilikt.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1: oscilikt example
instr 1
  kphs line 0, p3, 4
```

```
alx oscilikt 220.5, 1, 0
aly oscilikt 220.5, 1, -kphs
al = alx - aly

out al * 14000
endin

</CsInstruments>
<CsScore>

; Table #1: Sawtooth wave
f 1 0 3 -2 1 0 -1

; Play Instrument #1 for four seconds.
i 1 0 4
e

</CsScore>
</CsoundSynthesizer>
```

See Also

oscilikt and *oscilikts*.

Credits

Author: Istvan Varga

New in version 4.22

oscilikts

oscilikts — A linearly interpolated oscillator with sync status that allows changing the table number at k-rate.

Description

oscilikts is the same as *oscilikt*. Except it has a sync input that can be used to re-initialize the oscillator to a k-rate phase value. It is slower than *oscilikt* and *osciliktp*.

Syntax

```
ares oscilikts xamp, xcps, kfn, async, kphs [, istor]
```

Initialization

istor (optional, defaults to 0) -- skip initialization.

Performance

xamp -- amplitude.

xcps -- frequency in Hz. Zero and negative values are allowed. However, the absolute value must be less than *sr* (and recommended to be less than $sr/2$).

kfn -- function table number. Can be varied at control rate (useful to “morph” waveforms, or select from a set of band-limited tables generated by *GEN30*).

async -- any positive value resets the phase of *oscilikts* to *kphs*. Zero or negative values have no effect.

kphs -- sets the phase, initially and when it is re-initialized with *async*.

Examples

Here is an example of the *oscilikts* opcode. It uses the file *oscilikts.csd* [examples/oscilikts.csd].

Example 335. Example of the *oscilikts* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o oscilikts.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1: oscilikts example.
```

```
instr 1
; Frequency envelope.
kfrq expon 400, p3, 1200
; Phase.
kphs line 0.1, p3, 0.9

; Sync 1
atmp1 phasor 100
; Sync 2
atmp2 phasor 150
async diff 1 - (atmp1 + atmp2)

a1 oscilikts 14000, kfrq, 1, async, 0
a2 oscilikts 14000, kfrq, 1, async, -kphs

out a1 - a2
endin

</CsInstruments>
<CsScore>

; Table #1: Sawtooth wave
f 1 0 3 -2 1 0 -1

; Play Instrument #1 for four seconds.
i 1 0 4
e

</CsScore>
</CsoundSynthesizer>
```

See Also

oscilikt and *osciliktp*.

Credits

Author: Istvan Varga

New in version 4.22

osciln

osciln — Accesses table values at a user-defined frequency.

Description

Accesses table values at a user-defined frequency. This opcode can also be written as *oscilx*.

Syntax

```
ares osciln kamp, ifrq, ifn, itimes
```

Initialization

ifrq, itimes -- rate and number of times through the stored table.

ifn -- function table number.

Performance

kamp -- amplitude factor

osciln will sample several times through the stored table at a rate of *ifrq* times per second, after which it will output zeros. Generates audio signals only, with output values scaled by *kamp*.

See Also

table, tablei, table3, oscil1, oscil1i

oscils

oscils — A simple, fast sine oscillator

Description

Simple, fast sine oscillator, that uses only one multiply, and two add operations to generate one sample of output, and does not require a function table.

Syntax

```
ares oscils iamp, icps, iphs [, iflg]
```

Initialization

iamp -- output amplitude.

icps -- frequency in Hz (may be zero or negative, however the absolute value must be less than *sr/2*).

iphs -- start phase between 0 and 1.

iflg -- sum of the following values:

- 2: use double precision even if Csound was compiled to use floats. This improves quality (especially in the case of long performance time), but may be up to twice as slow.
- 1: skip initialization.

Performance

ares -- audio output

Examples

Here is an example of the oscils opcode. It uses the file *oscils.csd* [examples/oscils.csd].

Example 336. Example of the oscils opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o oscils.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

```
; Instrument #1 - a fast sine oscillator.
instr 1
  iamp = 10000
  icps = 440
  iphs = 0

  al oscils iamp, icps, iphs
  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Istvan Varga
January 2002

Example written by Kevin Conder.

New in version 4.18

oscilx

oscilx — Same as the osciln opcode.

Description

Same as the *osciln* opcode.

out

out — Writes mono audio data to an external device or stream.

Description

Writes mono audio data to an external device or stream.

Syntax

```
out asig
```

Performance

Sends mono audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with *nchnls* statement.

See Also

outh, uto, outq, outq1, outq2, outq3, outq4, outs, outs1, outs2, soundout

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

Original in Csound v1

out32

out32 — Writes 32-channel audio data to an external device or stream.

Description

Writes 32-channel audio data to an external device or stream.

Syntax

```
out32 asig1, asig2, asig3, asig4, asig5, asig6, asig7, asig8, asig10, \  
      asig11, asig12, asig13, asig14, asig15, asig16, asig17, asig18, \  
      asig19, asig20, asig21, asig22, asig23, asig24, asig25, asig26, \  
      asig27, asig28, asig29, asig30, asig31, asig32
```

Performance

out32 outputs 32 channels of audio.

Credits

outc, outch, outx, outz

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

outc

outc — Writes audio data with an arbitrary number of channels to an external device or stream.

Description

Writes audio data with an arbitrary number of channels to an external device or stream.

Syntax

```
outc asig1 [, asig2] [...]
```

Performance

outc outputs as many channels as provided. Any channels greater than *nchnls* are ignored. Zeros are added as necessary

Credits

out32, outch, outx, outz

Credits

Author: John fitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

outch

outch — Writes multi-channel audio data, with user-controllable channels, to an external device or stream.

Description

Writes multi-channel audio data, with user-controllable channels, to an external device or stream.

Syntax

```
outch ksig1, asig1 [, ksig2] [, asig2] [...]
```

Performance

outch outputs *asig1* on the channel determined by *ksig1*, *asig2* on the channel determined by *ksig2*, etc.

Credits

out32, *outc*, *outx*, *outz*

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

outh

outh — Writes 6-channel audio data to an external device or stream.

Description

Writes 6-channel audio data to an external device or stream.

Syntax

```
outh asig1, asig2, asig3, asig4, asig5, asig6
```

Performance

Sends 6-channel audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with *nchnls* statement.

See Also

out, outo, outq, outq1, outq2, outq3, outq4, outs, outs1, outs2, soundout

Credits

Author: John ffitch

Introduced before Version 3

outiat

outiat — Sends MIDI aftertouch messages at i-rate.

Description

Sends MIDI aftertouch messages at i-rate.

Syntax

```
outiat ichn, ivalue, imin, imax
```

Initialization

ichn -- MIDI channel number (1-16)

ivalue -- floating point value

imin -- minimum floating point value (converted in MIDI integer value 0)

imax -- maximum floating point value (converted in MIDI integer value 127 (7 bit))

Performance

outiat (i-rate aftertouch output) sends aftertouch messages. It works only with MIDI instruments which recognize them. It can drive a different value of a parameter for each note currently active.

It can scale an i-value floating-point argument according to the *imin* and *imax* values. For example, set *imin* = 1.0 and *imax* = 2.0. When the *ivalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *ivalue* argument receives a 1.0 value, it will send a 0 value. i-rate opcodes send their message once during instrument initialization.

See Also

outic14, *outic*, *outipat*, *outipb*, *outipc*, *outkat*, *outkc14*, *outkc*, *outkpat*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outic

outic — Sends MIDI controller output at i-rate.

Description

Sends MIDI controller output at i-rate.

Syntax

```
outic ichn, inum, ivalue, imin, imax
```

Initialization

ichn -- MIDI channel number (1-16)

inum -- controller number (0-127 for example 1 = ModWheel; 2 = BreathControl etc.)

ivalue -- floating point value

imin -- minimum floating point value (converted in MIDI integer value 0)

imax -- maximum floating point value (converted in MIDI integer value 127 (7 bit))

Performance

outic (i-rate MIDI controller output) sends controller messages to the MIDI OUT device. It works only with MIDI instruments which recognize them. It can drive a different value of a parameter for each note currently active.

It can scale an i-value floating-point argument according to the *imin* and *imax* values. For example, set *imin* = 1.0 and *imax* = 2.0. When the *ivalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *ivalue* argument receives a 1.0 value, it will send a 0 value. i-rate opcodes send their message once during instrument initialization.

See Also

outiat, *outic14*, *outipat*, *outipb*, *outipc*, *outkat*, *outkc14*, *outkc*, *outkpat*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outic14

outic14 — Sends 14-bit MIDI controller output at i-rate.

Description

Sends 14-bit MIDI controller output at i-rate.

Syntax

```
outic14 ichn, imsb, ilsb, ivalue, imin, imax
```

Initialization

ichn -- MIDI channel number (1-16)

imsb -- most significant byte controller number when using 14-bit parameters (0-127)

ilsb -- least significant byte controller number when using 14-bit parameters (0-127)

ivalue -- floating point value

imin -- minimum floating point value (converted in MIDI integer value 0)

imax -- maximum floating point value (converted in MIDI integer value 16383 (14-bit))

Performance

outic14 (i-rate MIDI 14-bit controller output) sends a pair of controller messages. This opcode can drive 14-bit parameters on MIDI instruments that recognize them. The first control message contains the most significant byte of *ivalue* argument while the second message contains the less significant byte. *imsb* and *ilsb* are the number of the most and less significant controller.

This opcode can drive a different value of a parameter for each note currently active.

It can scale an i-value floating-point argument according to the *imin* and *imax* values. For example, set *imin* = 1.0 and *imax* = 2.0. When the *ivalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *ivalue* argument receives a 1.0 value, it will send a 0 value. i-rate opcodes send their message once during instrument initialization.

See Also

outiat, *outic*, *outipat*, *outipb*, *outipc*, *outkat*, *outkc14*, *outkc*, *outkpat*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outipat

outipat — Sends polyphonic MIDI aftertouch messages at i-rate.

Description

Sends polyphonic MIDI aftertouch messages at i-rate.

Syntax

```
outipat ichn, inotenum, ivalue, imin, imax
```

Initialization

ichn -- MIDI channel number (1-16)

inotenum -- MIDI note number (used in polyphonic aftertouch messages)

ivalue -- floating point value

imin -- minimum floating point value (converted in MIDI integer value 0)

imax -- maximum floating point value (converted in MIDI integer value 127 (7 bit))

Performance

outipat (i-rate polyphonic aftertouch output) sends polyphonic aftertouch messages. It works only with MIDI instruments which recognize them. It can drive a different value of a parameter for each note currently active.

It can scale an i-value floating-point argument according to the *imin* and *imax* values. For example, set *imin* = 1.0 and *imax* = 2.0. When the *ivalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *ivalue* argument receives a 1.0 value, it will send a 0 value. i-rate opcodes send their message once during instrument initialization.

See Also

outiat, *outic14*, *outic*, *outipb*, *outipc*, *outkat*, *outkc14*, *outkc*, *outkpat*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outipb

outipb — Sends MIDI pitch-bend messages at i-rate.

Description

Sends MIDI pitch-bend messages at i-rate.

Syntax

```
outipb ichn, ivalue, imin, imax
```

Initialization

ichn -- MIDI channel number (1-16)

ivalue -- floating point value

imin -- minimum floating point value (converted in MIDI integer value 0)

imax -- maximum floating point value (converted in MIDI integer value 127 (7 bit))

Performance

outipb (i-rate pitch bend output) sends pitch bend messages. It works only with MIDI instruments which recognize them. It can drive a different value of a parameter for each note currently active.

It can scale an i-value floating-point argument according to the *imin* and *imax* values. For example, set *imin* = 1.0 and *imax* = 2.0. When the *ivalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *ivalue* argument receives a 1.0 value, it will send a 0 value. i-rate opcodes send their message once during instrument initialization.

See Also

outiat, *outic14*, *outic*, *outipat*, *outipc*, *outkat*, *outkc14*, *outkc*, *outkpat*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outipc

outipc — Sends MIDI program change messages at i-rate

Description

Sends MIDI program change messages at i-rate

Syntax

```
outipc ichn, iprog, imin, imax
```

Initialization

ichn -- MIDI channel number (1-16)

iprog -- program change number in floating point

imin -- minimum floating point value (converted in MIDI integer value 0)

imax -- maximum floating point value (converted in MIDI integer value 127 (7 bit))

Performance

outipc (i-rate program change output) sends program change messages. It works only with MIDI instruments which recognize them. It can drive a different value of a parameter for each note currently active.

It can scale an i-value floating-point argument according to the *imin* and *imax* values. For example, set *imin* = 1.0 and *imax* = 2.0. When the *ivalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *ivalue* argument receives a 1.0 value, it will send a 0 value. i-rate opcodes send their message once during instrument initialization.

See Also

outiat, *outic14*, *outic*, *outipat*, *outipb*, *outkat*, *outkc14*, *outkc*, *outkpat*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outkat

outkat — Sends MIDI aftertouch messages at k-rate.

Description

Sends MIDI aftertouch messages at k-rate.

Syntax

outkat *kchn*, *kvalue*, *kmin*, *kmax*

Performance

kchn -- MIDI channel number (1-16)

kvalue -- floating point value

kmin -- minimum floating point value (converted in MIDI integer value 0)

kmax -- maximum floating point value (converted in MIDI integer value 127)

outkat (k-rate aftertouch output) sends aftertouch messages. It works only with MIDI instruments which recognize them. It can drive a different value of a parameter for each note currently active.

It can scale the k-value floating-point argument according to the *kmin* and *kmax* values. For example: set *kmin* = 1.0 and *kmax* = 2.0. When the *kvalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *kvalue* argument receives a 1.0 value, it will send a 0 value. k-rate opcodes send a message each time the MIDI converted value of argument *kvalue* changes.

See Also

outiat, *outic14*, *outic*, *outipat*, *outipb*, *outipc*, *outkc14*, *outkc*, *outkpat*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outkc

outkc — Sends MIDI controller messages at k-rate.

Description

Sends MIDI controller messages at k-rate.

Syntax

outkc *kchn*, *knum*, *kvalue*, *kmin*, *kmax*

Performance

kchn -- MIDI channel number (1-16)

knum -- controller number (0-127 for example 1 = ModWheel; 2 = BreathControl etc.)

kvalue -- floating point value

kmin -- minimum floating point value (converted in MIDI integer value 0)

kmax -- maximum floating point value (converted in MIDI integer value 127 (7 bit))

outkc (k-rate MIDI controller output) sends controller messages to MIDI OUT device. It works only with MIDI instruments which recognize them. It can drive a different value of a parameter for each note currently active.

It can scale the k-value floating-point argument according to the *kmin* and *kmax* values. For example: set *kmin* = 1.0 and *kmax* = 2.0. When the *kvalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *kvalue* argument receives a 1.0 value, it will send a 0 value. k-rate opcodes send a message each time the MIDI converted value of argument *kvalue* changes.

See Also

outiat, *outic14*, *outic*, *outipat*, *outipb*, *outipc*, *outkat*, *outkc14*, *outkpat*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outkc14

outkc14 — Sends 14-bit MIDI controller output at k-rate.

Description

Sends 14-bit MIDI controller output at k-rate.

Syntax

```
outkc14 kchn, kmsb, klsb, kvalue, kmin, kmax
```

Performance

kchn -- MIDI channel number (1-16)

kmsb -- most significant byte controller number when using 14-bit parameters (0-127)

klsb -- least significant byte controller number when using 14-bit parameters (0-127)

kvalue -- floating point value

kmin -- minimum floating point value (converted in MIDI integer value 0)

kmax -- maximum floating point value (converted in MIDI integer value 16383 (14-bit))

outkc14 (k-rate MIDI 14-bit controller output) sends a pair of controller messages. It works only with MIDI instruments which recognize them. These opcodes can drive 14-bit parameters on MIDI instruments that recognize them. The first control message contains the most significant byte of *kvalue* argument while the second message contains the less significant byte. *kmsb* and *klsb* are the number of the most and less significant controller.

It can drive a different value of a parameter for each note currently active.

It can scale the k-value floating-point argument according to the *kmin* and *kmax* values. For example: set *kmin* = 1.0 and *kmax* = 2.0. When the *kvalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *kvalue* argument receives a 1.0 value, it will send a 0 value. k-rate opcodes send a message each time the MIDI converted value of argument *kvalue* changes.

See Also

outiat, *outic14*, *outic*, *outipat*, *outipb*, *outipc*, *outkat*, *outkc*, *outkpat*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outkpat

outkpat — Sends polyphonic MIDI aftertouch messages at k-rate.

Description

Sends polyphonic MIDI aftertouch messages at k-rate.

Syntax

outkpat *kchn*, *knotenum*, *kvalue*, *kmin*, *kmax*

Performance

kchn -- MIDI channel number (1-16)

knotenum -- MIDI note number (used in polyphonic aftertouch messages)

kvalue -- floating point value

kmin -- minimum floating point value (converted in MIDI integer value 0)

kmax -- maximum floating point value (converted in MIDI integer value 127 (7 bit))

outkpat (k-rate polyphonic aftertouch output) sends polyphonic aftertouch messages. It works only with MIDI instruments which recognize them. It can drive a different value of a parameter for each note currently active.

It can scale the k-value floating-point argument according to the *kmin* and *kmax* values. For example: set *kmin* = 1.0 and *kmax* = 2.0. When the *kvalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *kvalue* argument receives a 1.0 value, it will send a 0 value. k-rate opcodes send a message each time the MIDI converted value of argument *kvalue* changes.

See Also

outiat, *outic14*, *outic*, *outipat*, *outipb*, *outipc*, *outkat*, *outkc14*, *outkc*, *outkpb*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outkpb

outkpb — Sends MIDI pitch-bend messages at k-rate.

Description

Sends MIDI pitch-bend messages at k-rate.

Syntax

outkpb *kchn*, *kvalue*, *kmin*, *kmax*

Performance

kchn -- MIDI channel number (1-16)

kvalue -- floating point value

kmin -- minimum floating point value (converted in MIDI integer value 0)

kmax -- maximum floating point value (converted in MIDI integer value 127 (7 bit))

outkpb (k-rate pitch-bend output) sends pitch-bend messages. It works only with MIDI instruments which recognize them. It can drive a different value of a parameter for each note currently active.

It can scale the k-value floating-point argument according to the *kmin* and *kmax* values. For example: set *kmin* = 1.0 and *kmax* = 2.0. When the *kvalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *kvalue* argument receives a 1.0 value, it will send a 0 value. k-rate opcodes send a message each time the MIDI converted value of argument *kvalue* changes.

See Also

outiat, *outic14*, *outic*, *outipat*, *outipb*, *outipc*, *outkat*, *outkc14*, *outkc*, *outkpat*, *outkpc*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outkpc

outkpc — Sends MIDI program change messages at k-rate.

Description

Sends MIDI program change messages at k-rate.

Syntax

outkpc kchn, kprog, kmin, kmax

Performance

kchn -- MIDI channel number (1-16)

kprog -- program change number in floating point

kmin -- minimum floating point value (converted in MIDI integer value 0)

kmax -- maximum floating point value (converted in MIDI integer value 127 (7 bit))

outkpc (k-rate program change output) sends program change messages. It works only with MIDI instruments which recognize them. These opcodes can drive a different value of a parameter for each note currently active.

It can scale the k-value floating-point argument according to the *kmin* and *kmax* values. For example: set *kmin* = 1.0 and *kmax* = 2.0. When the *kvalue* argument receives a 2.0 value, the opcode will send a 127 value to the MIDI OUT device. When the *kvalue* argument receives a 1.0 value, it will send a 0 value. k-rate opcodes send a message each time the MIDI converted value of argument *kvalue* changes.

Examples

Here is an example of the outkpc opcode. It uses the file *outkpc.csd* [examples/outkpc.csd].

Example 337. Example of the outkpc opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

This example generates a program change and a note on Csound's MIDI output port whenever a note is received on channel 1. Be sure to have something connected to Csound's MIDI out port to hear the result.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadac    -d          -M0   -Q1;;RT audio I/O with MIDI in
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Example by Giorgio Zucco 2007
```

```
kprogram init 0

instr 1 ;Triggered by MIDI notes on channel 1

    ifund    notnum
    ivel     veloc
    idur = 1

; Sends a MIDI program change message according to
; the triggering note's velocity
outkpc      1 ,ivel ,0 ,127

noteondur   1 ,ifund ,ivel ,idur

endin

</CsInstruments>
<CsScore>
; Dummy ftable
f 0 60
</CsScore>
</CsoundSynthesizer>
```

Here is another example of the outkpc opcode. It uses the file *outkpc_flkt.csd* [examples/outkpc_flkt.csd].

Example 338. Example of the outkpc opcode using FLTK.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d           -M0  -Q1;;;RT audio I/O with MIDI in
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Example by Giorgio Zucco 2007

FLpanel "outkpc",200,100,90,90;start of container
gkpg, gihandle FLcount "Midi-Program change",0,127,1,5,1,152,40,16,23,-1
FLpanelEnd

FLrun

instr 1

ktrig changed gkpg
outkpc      ktrig,gkpg,0,127

endin

</CsInstruments>
<CsScore>
; Run instrument 1 for 60 seconds
i 1 0 60
</CsScore>
</CsoundSynthesizer>
```

See Also

outiat, outic14, outic, outipat, outipb, outipc, outkat, outkc14, outkc, outkpat, outkpb

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

outo

outo — Writes 8-channel audio data to an external device or stream.

Description

Writes 8-channel audio data to an external device or stream.

Syntax

```
outo asig1, asig2, asig3, asig4, asig5, asig6, asig7, asig8
```

Performance

Sends 8-channel audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with *nchnls* statement.

See Also

out, outh, outq, outq1, outq2, outq3, outq4, outs, outs1, outs2, soundout

Credits

Author: John ffitch

New after 3.30

outq

outq — Writes 4-channel audio data to an external device or stream.

Description

Writes 4-channel audio data to an external device or stream.

Syntax

```
outq asig1, asig2, asig3, asig4
```

Performance

Sends 4-channel audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with the *nchnls* statement. Opcodes can be chosen to direct sound to any particular channel: *outs1* sends to stereo channel 1, *outq3* to quad channel 3, etc.

See Also

out, *outh*, *outo*, *outq1*, *outq2*, *outq3*, *outq4*, *outs*, *outs1*, *outs2*, *soundout*

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

outq1

outq1 — Writes samples to quad channel 1 of an external device or stream.

Description

Writes samples to quad channel 1 of an external device or stream.

Syntax

```
outq1 asig
```

Performance

Sends audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with the *nchnls* statement. Opcodes can be chosen to direct sound to any particular channel: *outs1* sends to stereo channel 1, *outq3* to quad channel 3, etc.

See Also

out, *outh*, *outo*, *outq*, *outq2*, *outq3*, *outq4*, *outs*, *outs1*, *outs2*, *soundout*

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

outq2

outq2 — Writes samples to quad channel 2 of an external device or stream.

Description

Writes samples to quad channel 2 of an external device or stream.

Syntax

```
outq2 asig
```

Performance

Sends audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with the *nchnls* statement. Opcodes can be chosen to direct sound to any particular channel: *outs1* sends to stereo channel 1, *outq3* to quad channel 3, etc.

See Also

out, *outh*, *outo*, *outq*, *outq1*, *outq3*, *outq4*, *outs*, *outs1*, *outs2*, *soundout*

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

outq3

outq3 — Writes samples to quad channel 3 of an external device or stream.

Description

Writes samples to quad channel 3 of an external device or stream.

Syntax

```
outq3 asig
```

Performance

Sends audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with the *nchnls* statement. Opcodes can be chosen to direct sound to any particular channel: *outs1* sends to stereo channel 1, *outq3* to quad channel 3, etc.

See Also

out, *outh*, *outo*, *outq*, *outq1*, *outq2*, *outq4*, *outs*, *outs1*, *outs2*, *soundout*

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

outq4

outq4 — Writes samples to quad channel 4 of an external device or stream.

Description

Writes samples to quad channel 4 of an external device or stream.

Syntax

```
outq4 asig
```

Performance

Sends audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with the *nchnls* statement. Opcodes can be chosen to direct sound to any particular channel: *outs1* sends to stereo channel 1, *outq3* to quad channel 3, etc.

See Also

out, *outh*, *outo*, *outq*, *outq1*, *outq2*, *outq3*, *outs*, *outs1*, *outs2*, *soundout*

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

outrg

outrg — Allow output to a range of adjacent audio channels on the audio input device

Description

outrg outputs audio to a range of adjacent audio channels on the audio output device.

Syntax

```
outrg kstart, aout1 [,aout2, aout3, ..., aoutN]
```

Performance

kstart - the number of the first channel of the output device to be accessed (channel numbers starts with 1, which is the first channel)

aout1, *aout2*, ... *aoutN* - the arguments containing the audio to be output to the corresponding output channels.

outrg allows to output a range of adjacent channels to the output device. *kstart* indicates the first channel to be accessed (channel 1 is the first channel). The user must be sure that the number obtained by summing *kstart* plus the number of accessed channels -1 is $\leq nchnls$.

Credits

Author: Gabriel Maldonado

New in version 5.06

outs

outs — Writes stereo audio data to an external device or stream.

Description

Writes stereo audio data to an external device or stream.

Syntax

```
outs asig1, asig2
```

Performance

Sends stereo audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with the *nchnls* statement. Opcodes can be chosen to direct sound to any particular channel: *outs1* sends to stereo channel 1, *outq3* to quad channel 3, etc.

See Also

out, outh, outo, outq, outq1, outq2, outq3, outq4, outs1, outs2, soundout

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

outs1

outs1 — Writes samples to stereo channel 1 of an external device or stream.

Description

Writes samples to stereo channel 1 of an external device or stream.

Syntax

```
outs1 asig
```

Performance

Sends audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with the *nchnls* statement. Opcodes can be chosen to direct sound to any particular channel: *outs1* sends to stereo channel 1, *outq3* to quad channel 3, etc.

See Also

out, *outh*, *outo*, *outq*, *outq1*, *outq2*, *outq3*, *outq4*, *outs*, *outs2*, *soundout*

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

outs2

outs2 — Writes samples to stereo channel 2 of an external device or stream.

Description

Writes samples to stereo channel 2 of an external device or stream.

Syntax

```
outs2 asig
```

Performance

Sends audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument.

The type (mono, stereo, quad, hex, or oct) should agree with *nchnls*. But as of version 3.50, Csound will attempt to change an incorrect opcode to agree with the *nchnls* statement. Opcodes can be chosen to direct sound to any particular channel: *outs1* sends to stereo channel 1, *outq3* to quad channel 3, etc.

See Also

out, outh, outo, outq, outq1, outq2, outq3, outq4, outs, outs1, soundout

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

outvalue

outvalue — Sends a k-rate signal or string to a user-defined channel.

Description

Sends a k-rate signal or string to a user-defined channel.

Syntax

```
outvalue "channel name", kvalue
```

```
outvalue "channel name", "string"
```

Performance

"channel name" -- An integer or string (in double-quotes) representing channel.

kvalue -- The k-rate value that is sent to the channel.

string -- The string or string variable that is sent to the channel.

See Also

invalue

Credits

Author: Matt Ingalls

outx

outx — Writes 16-channel audio data to an external device or stream.

Description

Writes 16-channel audio data to an external device or stream.

Syntax

```
outx asig1, asig2, asig3, asig4, asig5, asig6, asig7, asig8, \  
    asig9, asig10, asig11, asig12, asig13, asig14, asig15, asig16
```

Performance

outx outputs 32 channels of audio.

Credits

out32, outc, outch, outz

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

outz

outz — Writes multi-channel audio data from a ZAK array to an external device or stream.

Description

Writes multi-channel audio data from a ZAK array to an external device or stream.

Syntax

```
outz ksig1
```

Performance

outz outputs from a ZAK array for *nchnls* of audio.

Credits

out32, outc, outch, outx

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.06

p

p — Show the value in a given p-field.

Description

Show the value in a given p-field.

Syntax

p(x)

This function works at i-rate and k-rate.

Initialization

x -- the number of the p-field.

Performance

The value returned by the *p* function is the value in a p-field.

Examples

Here is an example of the p opcode. It uses the file *p.csd* [examples/p.csd].

Example 339. Example of the p opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o p.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Get the value in the fourth p-field, p4.
i1 = p(4)

print i1
endin

</CsInstruments>
<CsScore>

; p4 = value to be printed.
; Play Instrument #1 for one second, p4 = 50.375.
i 1 0 1 50.375
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  i1 = 50.375
```

Credits

Example written by Kevin Conder.

pan

pan — Distribute an audio signal amongst four channels.

Description

Distribute an audio signal amongst four channels with localization control.

Syntax

```
a1, a2, a3, a4 pan asig, kx, ky, ifn [, imode] [, ioffset]
```

Initialization

ifn -- function table number of a stored pattern describing the amplitude growth in a speaker channel as sound moves towards it from an adjacent speaker. Requires extended guard-point.

imode (optional) -- mode of the *kx*, *ky* position values. 0 signifies raw index mode, 1 means the inputs are normalized (0 - 1). The default value is 0.

ioffset (optional) -- offset indicator for *kx*, *ky*. 0 infers the origin to be at channel 3 (left rear); 1 requests an axis shift to the quadrasonic center. The default value is 0.

Performance

pan takes an input signal *asig* and distributes it amongst four outputs (essentially quad speakers) according to the controls *kx* and *ky*. For normalized input (mode=1) and no offset, the four output locations are in order: left-front at (0,1), right-front at (1,1), left-rear at the origin (0,0), and right-rear at (1,0). In the notation (*kx*, *ky*), the coordinates *kx* and *ky*, each ranging 0 - 1, thus control the 'rightness' and 'forwardness' of a sound location.

Movement between speakers is by amplitude variation, controlled by the stored function table *ifn*. As *kx* goes from 0 to 1, the strength of the right-hand signals will grow from the left-most table value to the right-most, while that of the left-hand signals will progress from the right-most table value to the left-most. For a simple linear pan, the table might contain the linear function 0 - 1. A more correct pan that maintains constant power would be obtained by storing the first quadrant of a sinusoid. Since *pan* will scale and truncate *kx* and *ky* in simple table lookup, a medium-large table (say 8193) should be used.

kx, *ky* values are not restricted to 0 - 1. A circular motion passing through all four speakers (inscribed) would have a diameter of root 2, and might be defined by a circle of radius R = root 1/2 with center at (.5,.5). *kx*, *ky* would then come from Rcos(angle), Rsin(angle), with an implicit origin at (.5,.5) (i.e. *ioffset* = 1). Unscaled raw values operate similarly. Sounds can thus be located anywhere in the polar or Cartesian plane; points lying outside the speaker square are projected correctly onto the square's perimeter as for a listener at the center.

Examples

```
instr      1
  k1      phasor      1/p3      ; fraction of circle
  k2      tablei      k1, 1, 1    ; sin of angle (sinusoid in f1)
  k3      tablei      k1, 1, 1, .25, 1 ; cos of angle (sin offset 1/4 circle)
  a1      oscili      10000,440, 1 ; audio signal..
  a1,a2,a3,a4 pan      a1, k2/2, k3/2, 2, 1, 1 ; sent in a circle (f2=1st quad sin)

          outq a1, a2, a3, a4
endin
```


pan2

pan2 — Distribute an audio signal across two channels.

Description

Distribute an audio signal across two channels with a choice of methods.

Syntax

```
a1, a2 pan2 asig, xp [, imode]
```

Initialization

imode (optional) -- mode of the stereo positioning algorithm. 0 signifies equal power (harmonic) panning, 1 means the square root method, and 2 means simple linear. The default value is 0.

Performance

pan2 takes an input signal *asig* and distributes it across two outputs (essentially stereo speakers) according to the control *xp* which can be k- or a-rate. A zero value for *xp* indicates hard left, and a 1 is hard right.

Examples

```
instr      1  
  kline line    0, p3, 1      ; straight line  
  ain   oscili 10000,440, 1   ; audio signal..  
  a1,a2 pan2   ain, kline    ; sent across image  
  
  outs      a1, a2  
endin
```

Credits

Author: John fitch
University of Bath, Codemist Ltd.
Bath, UK
September 2007

New in version 5.07

pareq

pareq — Implementation of Zoelzer's parametric equalizer filters.

Description

Implementation of Zoelzer's parametric equalizer filters, with some modifications by the author.

The formula for the low shelf filter is:

```
omega = 2*pi*f/sr
K      = tan(omega/2)

b0  = 1 + sqrt(2*V)*K + V*K^2
b1  = 2*(V*K^2 - 1)
b2  = 1 - sqrt(2*V)*K + V*K^2

a0  = 1 + K/Q + K^2
a1  = 2*(K^2 - 1)
a2  = 1 - K/Q + K^2
```

The formula for the high shelf filter is:

```
omega = 2*pi*f/sr
K      = tan((pi-omega)/2)

b0  = 1 + sqrt(2*V)*K + V*K^2
b1  = -2*(V*K^2 - 1)
b1  = 1 - sqrt(2*V)*K + V*K^2

a0  = 1 + K/Q + K^2
a1  = -2*(K^2 - 1)
a2  = 1 - K/Q + K^2
```

The formula for the peaking filter is:

```
omega = 2*pi*f/sr
K      = tan(omega/2)

b0 = 1 + V*K/2 + K^2
b1 = 2*(K^2 - 1)
b2 = 1 - V*K/2 + K^2

a0 = 1 + K/Q + K^2
a1 = 2*(K^2 - 1)
a2 = 1 - K/Q + K^2
```

Syntax

```
ares pareq asig, kc, kv, kq [, imode] [, iskip]
```

Initialization

imode (optional, default: 0) -- operating mode

- 0 = Peaking
- 1 = Low Shelving
- 2 = High Shelving

iskip (optional, default=0) -- if non zero skip the initialisation of the filter. (New in Csound version 4.23f13 and 5.0)

Performance

kc -- center frequency in peaking mode, corner frequency in shelving mode.

kv -- amount of boost or cut. A value less than 1 is a cut. A value greater than 1 is a boost. A value of 1 is a flat response.

kq -- Q of the filter (sqrt(.5) is no resonance)

asig -- the incoming signal

Examples

Here is an example of the *pareq* opcode. It uses the file *pareq.csd* [examples/pareq.csd].

Example 340. Example of the *pareq* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o pareq.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

instr 15
  ifc      =      p4          ; Center / Shelf
  kq       =      p5          ; Quality factor sqrt(.5) is no resonance
  kv       =      ampdb(p6)    ; Volume Boost/Cut
  imode    =      p7          ; Mode 0=Peaking EQ, 1=Low Shelf, 2=High Shelf
  kfc      linseg ifc*2, p3, ifc/2
  asig     rand 5000          ; Random number source for testing
  aout     pareq asig, kfc, kv, kq, imode ; Parametric equalization
  outs     aout, aout         ; Output the results
endin

</CsInstruments>
<CsScore>

; SCORE:
;   Sta  Dur  Fcenter  Q          Boost/Cut(dB)  Mode
i15 0    1    10000    .2          12            1
i15 +    .    5000    .2          12            1
```

```
i15 . . 1000 .707 -12 2
i15 . . 5000 .1 -12 0
e
```

```
</CsScore>
</CsoundSynthesizer>
```

Credits

Hans Mikelson
December 1998

New in Csound version 3.50

partials

partials — Partial track spectral analysis.

Description

The `partials` opcode takes two input PV streaming signals containing `AMP_FREQ` and `AMP_PHASE` signals (as generated for instance by `pvsifd` or in the first case, by `pvsanal`) and performs partial track analysis, as described in Lazzarini et al, "Time-stretching using the Instantaneous Frequency Distribution and Partial Tracking", Proc.of ICMC05, Barcelona. It generates a TRACKS PV streaming signal, containing amplitude, frequency, phase and track ID for each output track. This type of signal will contain a variable number of output tracks, up to the total number of analysis bins contained in the inputs ($\text{fftsize}/2 + 1$ bins). The second input (`AMP_PHASE`) is optional, as it can take the same signal as the first input. In this case, however, all phase information will be NULL and resynthesis using phase information cannot be performed.

Syntax

```
ftrks partials ffr, fphs, kthresh, kminpts, kmaxgap, imaxtracks
```

Performance

ftrks -- output pv stream in TRACKS format

ffr -- input pv stream in `AMP_FREQ` format

fphs -- input pv stream in `AMP_PHASE` format

kthresh -- analysis threshold. Tracks below $\text{kthresh} * \text{max_magnitude}$ will be discarded ($1 > \text{kthresh} \geq 0$).

kminpoints -- minimum number of time points for a detected peak to make a track (1 is the minimum). Since this opcode works with streaming signals, larger numbers will increase the delay between input and output, as we have to wait for the required minimum number of points.

kmaxgap -- maximum gap between time-points for track continuation (> 0). Tracks that have no continuation after *kmaxgap* will be discarded.

imaxtracks -- maximum number of analysis tracks (number of bins \geq *imaxtracks*)

Examples

Example 341. Example

```
ain inch 1 ; input signal
fs1,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fs1,fsi2,.003,1,3,500 ; partial tracking
aout resyn fst, 1, 1.5, 500, 1 ; resynthesis (up a 5th)
out aout
```

The example above shows partial tracking of an ifd-analysis signal and cubic-phase additive resynthesis with pitch shifting.

Credits

Author: Victor Lazzarini;
June 2005

New plugin in version 5

November 2004.

partikkel

partikkel — Granular synthesizer with "per grain" control over many of its parameters. Has a sync input to synchronize its internal grain scheduler clock to an external clock source.

Description

partikkel was conceived after reading Curtis Roads' book "Microsound", and the goal was to create an opcode that was capable of all time-domain varieties of granular synthesis described in this book. The idea being that most of the techniques only differ in parameter values, and by having a single opcode that can do all varieties of granular synthesis makes it possible to interpolate between techniques. Granular synthesis is sometimes dubbed particle synthesis, and it was thought apt to name the opcode *partikkel* to distinguish it from other granular opcodes.

Some of the input parameters to *partikkel* is table numbers, pointing to tables where values for the "per grain" parameter changes are stored. *partikkel* can use single-cycle or complex (e.g. sampled sound) waveforms as source waveforms for grains. Each grain consists of a mix of 4 source waveforms. Individual tuning of the base frequency can be done for each of the 4 source waveforms. Frequency modulation inside each grain is enabled via an auxillary audio input (*awavfm*). Trainlet synthesis is available, and trainlets can be mixed with wavetable based grains. Up to 8 separate audio outputs can be used.

Syntax

```
a1 [, a2, a3, a4, a5, a6, a7, a8] partikkel agrainfreq, \  
    kdistribution, idisttab, async, kenv2amt, ienv2tab, ienv_attack, \  
    ienv_decay, ksustain_amount, ka_d_ratio, kduration, kamp, igainmasks, \  
    kwavfreq, ksweepshape, iwavfreqstarttab, iwavfreqendtab, awavfm, \  
    ifmampstab, kfmenv, icosine, ktraincps, knumpartials, kchroma, \  
    ichannelmasks, krandommask, kwaveform1, kwaveform2, kwaveform3, \  
    kwaveform4, iwaveamptab, asamplepos1, asamplepos2, asamplepos3, \  
    asamplepos4, kwavekey1, kwavekey2, kwavekey3, kwavekey4, imax_grains \  
    [, iopcode_id]
```

Initialization

idisttab -- function table number, distribution for random grain displacements over time. The table values are interpreted as "displacement amount" scaled by 1/grainrate. This means that a value of 0.5 in the table will displace a grain by half the grainrate period. The table values are read randomly, and scaled by *kdistribution*. For realistic stochastic results, it is advisable not to use a too small table size, as this limits the amount of possible displacement values. This can also be utilized for other purposes, e.g. using quantized displacement values to work with controlled time displacement from the periodic grain rate. If *kdistribution* is negative, the table values will be read sequentially. A default table might be selected by using -1 as the ftable number, for *idisttab* the default uses a zero distribution (no displacement).

ienv_attack -- function table number, attack shape of grain. Needs extended guard point. A default table might be selected by using -1 as the ftable number, for *ienv_attack* the default uses a square window (no enveloping).

ienv_decay -- function table number, decay shape of grain. Needs extended guard point. A default table might be selected by using -1 as the ftable number, for *ienv_decay* the default uses a square window (no enveloping).

ienv2tab -- function table number, additional envelope applied to grain, done after attack and decay envelopes. Can be used e.g. for fof formant synthesis. Needs extended guard point. A default table might be selected by using -1 as the ftable number, for *ienv2tab* the default uses a square window (no enveloping).

icosine -- function table number, must contain a cosine, used for trainlets. Table size should be at

least 2048 for good quality trainlets.

igainmasks -- function table number, gain per grain. The sequence of values in the table is as follows: index 0 is used as a loop start point in reading the values, index 1 is used as a loop end point. Remaining indices contain gain values (normally in range 0 - 1, but other values are allowed, negative values will invert phase of waveform inside grain) for a sequence of grains, these are read at grain rate enabling exact patterns of "gain per grain". The loop start and end points are zero based with an origin at index 2, e.g. a loop start value of 0 and loop end value of 3 will read indices 2,3,4,5 in a loop at grain rate. A default table might be selected by using -1 as the ftable number, for *igainmasks* the default disables gain masking (all grains are given a gain masking value of 1).

ichannelmasks -- function table number, see *igainmasks* for a description of how the values in the table are read. Range is 0 to N, where N is the number of output channels minus 1. A value of zero will send the grain to audio output 1 from the opcode. Fractional values are allowed, e.g. a value of 3.5 will mix the grain equally to outputs 4 and 5. The user is responsible for keeping the values in range, no range checking is done. The opcode will crash with out of range values. A default table might be selected by using -1 as the ftable number, for *ichannelmasks* the default disables channel masking (all grains are given a channel masking value of 0 and are sent to *partikkel* audio out 1).

iwavfreqstarttab -- function table number, see *igainmasks* for a description of how the values in the table are read. Start frequency multiplier for each grain. Pitch will glide from start frequency to end frequency following a line or curve as set by *ksweepshape*. A default table might be selected by using -1 as the ftable number, for *iwavfreqstarttab* the default uses a multiplier of 1, disabling any start frequency modification.

iwavfreqendtab -- function table number, see *iwavfreqstarttab*. End frequency multiplier for each grain. A default table might be selected by using -1 as the ftable number, for *iwavfreqendtab* the default uses a multiplier of 1, disabling any end frequency modification.

ifmamptab -- function table number, see *igainmasks* for a description of how the values in the table are read. Frequency modulation index per grain. The signal *awavfm* will be multiplied by values read from this table. A default table might be selected by using -1 as the ftable number, for *ifmamptab* the default uses 1 as the index multiplier, enabling fm for all grains.

iwaveamptab -- function table number, the indices are read in a similar way to what is used for *igainmasks*. Index 0 is used as a loop start point, and index 1 is used as a loop end point. The rest of the indices are read in groups of 5, where each value represent a gain value for each of the 4 source waveforms, and the 5th value represent trainlet amplitude. A default table might be selected by using -1 as the ftable number, for *iwaveamptab* the default uses an equal mix of all 4 source waveforms (each with an amplitude of 0.5) and setting trainlet amp to zero.

Computation of trainlets can be CPU intensive, and setting *ktrainamp* to zero will skip most of the trainlet computations. Trainlets will be normalized to peak (*ktrainamp*), compensating for amplitude variations caused by variations in *kpartials* and *kchroma*.

imax_grains -- maximum number of grains per k-period. Estimating a large value should not affect performance, exceeding this value will lead to "oldest grains" being deleted.

iopcode_id -- the opcode id, linking an instance of *partikkel* to an instance of *partikkelsync*, the linked *partikkelsync* will output trigger pulses synchronized to *partikkel*'s grain maker scheduler. The default value is zero, which means no connection to any *partikkelsync* instances.

Performance

xgrainfreq -- number of grains per second. A value of zero is allowed, and this will defer all grain scheduling to the sync input.

async -- sync input. Input values are added to the phase value of the internal grain maker clock, enabling tempo synchronization with an external clock source. As this is an a-rate signal, inputs are usually pulses of length 1/sr. Using such pulses, the internal phase value can be "nudged" up or down, enabling soft or hard synchronization. Negative input values decrements the internal phase, while positive values in the range 0 to 1 increments the internal phase. An input value of 1 will always make *partikkel* generate a grain. If the value remains at 1, the internal grain scheduler clock

will pause but any currently playing grains will still play to end.

kdistribution -- periodic or stochastic distribution of grains, 0 = periodic. Normal range 0 to 1, but higher values can be used for the classic stochastic grain distribution effect. If *kdistribution* is negative, the result is deterministic time displacement as described by *idisttab*.

kenv2amt -- amount of enveloping for the secondary envelope for each grain. Range 0 to 1, where 0 is no secondary enveloping (square window), a value of 0.5 will use an interpolation between a square window and the shape set by *ienv2tab*.

ksustain_amount -- sustain time as fraction of grain duration. I.e. balance between enveloped time(attack+decay) and sustain level time. The sustain level is taken from the last value of the *ienv_attack* ftable.

ka_d_ratio -- balance between attack time and decay time. For example, with *ksustain_amount* set to 0.5 and *ka_d_ratio* set to 0.5, the attack envelope of each grain will take 25% of the grain duration, full amplitude (sustain) will be held for 50% of the grain duration, and the decay envelope will take the remaining 25% of the grain duration.

kduration -- grain duration in milliseconds.

kamp -- amplitude scaling of the opcode's output. Multiplied by per grain amplitude read from *igainmasks*.

kwavfreq -- transposition scaling. Multiplied with start and end transposition values read from *iwavfreqstarttab* and *iwavfreqendtab*.

ksweepshape -- transposition sweep shape, controls the curvature of the transposition sweep. Range 0 to 1. Low values will hold the transposition at the start value longer and then drop to the end value quickly, high values will drop to the end value quickly. A value of 0.5 will give a linear sweep. A value of exactly 0 will bypass sweep and only use the start frequency, while a value of exactly 1 will bypass sweep and only use the end frequency. The sweep generator might be slightly inaccurate in hitting the end frequency when using a steep curve and very long grains.

awavfm -- audio input for frequency modulation inside grain.

kfmenv -- function table number, envelope for FM modulator signal enabling the modulation index to change over the duration of a grain.

ktraincps -- trainlet fundamental frequency.

knumpartials -- number of partials in trainlets.

kchroma -- chroma of trainlets. A value of 1 give equal amplitude to each partial, higher values will reduce the amplitude of lower partials while strengthening the amplitude of the higher partials.

krandommask -- random masking (muting) of individual grains. Range 0 to 1, where a value of 0 will give no masking (all grains are played), and a value of 1 will mute all grains.

kwaveform1 -- table number for source waveform 1.

kwaveform2 -- table number for source waveform 2.

kwaveform3 -- table number for source waveform 3.

kwaveform4 -- table number for source waveform 4.

asamplepos1 -- start position for reading source waveform 1.

asamplepos2 -- start position for reading source waveform 2.

asamplepos3 -- start position for reading source waveform 3.

asamplepos4 -- start position for reading source waveform 4.

kwavekey1 -- original key of source waveform 1. Can be used to transpose each source waveform independently.

kwavekey2 -- as *kwavekey1*, but for source waveform 2.

kwavekey3 -- as *kwavekey1*, but for source waveform 3.

kwavekey4 -- as *kwavekey1*, but for source waveform 4.

Examples

Here is an example of the *partikkel* opcode. It uses the file *PartikkelExample1.csd* [examples/PartikkelExample1.csd].

Example 342. Example of the *partikkel* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out
-odac          ;;RT audio
; For Non-realtime output leave only the line below:
; -o partikkel.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 20
nchnls = 2

; Example by Joachim Heintz and Oeyvind Brandtsegg 2008

giCosine ftgen 0, 0, 8193, 9, 1, 1, 90 ; cosine
giDisttab ftgen 0, 0, 32768, 7, 0, 32768, 1 ; for kdistribution
giFile    ftgen 0, 0, 0, 1, "fox.wav", 0, 0, 0 ; soundfile for source waveform
giWin     ftgen 0, 0, 4096, 20, 9, 1 ; grain envelope
giPan     ftgen 0, 0, 32768, -21, 1 ; for panning (random values between 0 and 1)

; *****
; partikkel example, processing of soundfile
; uses the file "fox.wav"
; *****
instr 1

/*score parameters*/
ispeed          = p4 ; 1 = original speed
igrainrate      = p5 ; grain rate
igrainsize      = p6 ; grain size in ms
icent           = p7 ; transposition in cent
iposrand        = p8 ; time position randomness (offset) of the pointer in ms
icentrand       = p9 ; transposition randomness in cents
ipan            = p10 ; panning narrow (0) to wide (1)
idist           = p11 ; grain distribution (0=periodic, 1=scattered)

/*get length of source wave file, needed for both transposition and time pointer*/
ifilen          tablen giFile
ifildur         = ifilen / sr

/*sync input (disabled)*/
async          = 0

/*grain envelope*/
kenv2amt        = 1 ; use only secondary envelope
ienv2tab        = giWin ; grain (secondary) envelope
ienv_attack     = -1 ; default attack envelope (flat)
ienv_decay      = -1 ; default decay envelope (flat)
ksustain_amount = 0.5 ; no meaning in this case (use only secondary envelope, ie
ka_d_ratio      = 0.5 ; no meaning in this case (use only secondary envelope, ie

/*amplitude*/
kamp            = 0.4*0dbfs ; grain amplitude
igainmasks     = -1 ; (default) no gain masking
```

```
/*transposition*/
kcentrand      rand icentrand ; random transposition
iorig          = 1 / ifildur ; original pitch
kwavfreq       = iorig * cent(icent + kcentrand)

/*other pitch related (disabled)*/
ksweepshape    = 0 ; no frequency sweep
iwavfreqstarttab = -1 ; default frequency sweep start
iwavfreqendtab  = -1 ; default frequency sweep end
awavfm         = 0 ; no FM input
ifmampptab     = -1 ; default FM scaling (=1)
kfmenv         = -1 ; default FM envelope (flat)

/*trainlet related (disabled)*/
icosine        = giCosine ; cosine ftable
kTrainCps      = igrainrate ; set trainlet cps equal to grain rate for single-cycle trainlet in
knumpartials   = 1 ; number of partials in trainlet
kchroma        = 1 ; balance of partials in trainlet

/*panning, using channel masks*/
imid           = .5; center
ileftmost      = imid - ipan/2
irightmost     = imid + ipan/2
giPanthis      ftgen 0, 0, 32768, -24, giPan, ileftmost, irightmost ; rescales giPan according to
tableiw        0, 0, giPanthis ; change index 0 ..
tableiw        32766, 1, giPanthis ; ... and 1 for ichannel
ichannelmasks  = giPanthis ; ftable for panning

/*random gain masking (disabled)*/
krandommask    = 0

/*source waveforms*/
kwaveform1     = giFile ; source waveform
kwaveform2     = giFile ; all 4 sources are the same
kwaveform3     = giFile
kwaveform4     = giFile
iwaveamptab    = -1 ; (default) equal mix of source waveforms and no amplitude for tra

/*time pointer*/
afilposphas    phasor ispeed / ifildur
/*generate random deviation of the time pointer*/
iposrandsec    = iposrand / 1000 ; ms -> sec
iposrand       = iposrandsec / ifildur ; phase values (0-1)
krndpos        linrand iposrand ; random offset in phase values
/*add random deviation to the time pointer*/
asamplepos1    = afileposphas + krndpos; resulting phase values (0-1)
asamplepos2    = asamplepos1
asamplepos3    = asamplepos1
asamplepos4    = asamplepos1

/*original key for each source waveform*/
kwavekey1      = 1
kwavekey2      = kwavekey1
kwavekey3      = kwavekey1
kwavekey4      = kwavekey1

/* maximum number of grains per k-period*/
imax_grains    = 100

aL, aR          partikkel igrainrate, idist, giDisttab, async, kenv2amt, ienv2tab, \
                ienv_attack, ienv_decay, ksustain_amount, ka_d_ratio, igrainsize, kamp, igainmasks, \
                kwavfreq, ksweepshape, iwavfreqstarttab, iwavfreqendtab, awavfm, \
                ifmampptab, kfmenv, icosine, kTrainCps, knumpartial, \
                kchroma, ichannelmasks, krandommask, kwaveform1, kwaveform2, kwaveform3, kwaveform4, \
                iwaveamptab, asamplepos1, asamplepos2, asamplepos3, asamplepos4, \
                kwavekey1, kwavekey2, kwavekey3, kwavekey4, imax_grains

                outs          aL, aR

endin

</CsInstruments>
<CsScore>
;il st dur speed grate gsize cent posrnd cntrnd pan dist
il 0 2.757 1 200 15 0 0 0 0 0
s
il 0 2.757 1 200 15 400 0 0 0 0
s
il 0 2.757 1 15 450 400 0 0 0 0
s
il 0 2.757 1 15 450 400 0 0 0 0.4
s
il 0 2.757 1 200 15 0 400 0 0 1
s
il 0 5.514 .5 200 20 0 0 600 .5 1
s
```

```
i1 0 11.028 .25 200 15 0 1000 400 1 1
```

```
</CsScore>  
</CsoundSynthesizer>
```

Here is another example of the partikkel opcode. It uses the file *partikkel_softsync.csd* [examples/partikkel_softsync.csd].

Example 343. Example with soft sync of two partikkel generators.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  
-odac ;;;RT audio  
; For Non-realtime ouput leave only the line below:  
; -o partikkel_softsync.wav -W ;;; for file output any platform  
</CsOptions>  
<CsInstruments>  
  
sr = 44100  
ksmps = 20  
nchnls = 2  
  
; Example by Oeyvind Brandtsegg 2007, revised 2008  
  
giSine          ftgen 0, 0, 65537, 10, 1  
giCosine ftgen 0, 0, 8193, 9, 1, 1, 90  
giSigmoidRise ftgen          0, 0, 8193, 19, 0.5, 1, 270, 1 ; rising sigmoid  
giSigmoidFall ftgen          0, 0, 8193, 19, 0.5, 1, 90, 1 ; falling sigmoid  
  
; *****  
; example of soft synchronization of two partikkel instances  
; *****  
instr 1  
  
/*score parameters*/  
igrainrate = p4 ; grain rate  
igrainsize = p5 ; grain size in ms  
igrainFreq = p6 ; fundamental frequency of source waveform  
iosc2Dev = p7 ; partikkel instance 2 grain rate deviation factor  
iMaxSync = p8 ; max soft sync amount (increasing to this value during length of note)  
  
/*overall envelope*/  
iattack = 0.001  
idecay = 0.2  
isustain = 0.7  
irelease = 0.2  
amp linsegr 0, iattack, 1, idecay, isustain, 1, isustain, irelease, 0  
  
kgrainfreq = igrainrate ; grains per second  
kdistribution = 0 ; periodic grain distribution  
kdisttab = -1 ; (default) flat distribution used  
; for grain distribution  
  
async = 0 ; no sync input  
kenv2amt = 0 ; no secondary enveloping  
ienv2tab = -1 ; default secondary envelope (flat)  
ienv_attack = giSigmoidRise ; default attack envelope (flat)  
ienv_decay = giSigmoidFall ; default decay envelope (flat)  
ksustain_amount = 0.3 ; time (in fraction of grain dur) at  
; sustain level for each grain  
; balance between attack and decay time  
ka_d_ratio = 0.2  
kduration = igrainsize ; set grain duration in ms  
kamp = 0.2*0dbfs ; amp  
igainmasks = -1 ; (default) no gain masking  
kwavfreq = igrainFreq ; fundamental frequency of source waveform  
ksweepshape = 0 ; shape of frequency sweep (0=no sweep)  
iwavfreqstarttab = -1 ; default frequency sweep start  
; (value in table = 1, which give  
; no frequency modification)  
iwavfreqendtab = -1 ; default frequency sweep end  
; (value in table = 1, which give  
; no frequency modification)  
  
awavfm = 0 ; no FM input  
ifmampmtab = -1 ; default FM scaling (=1)  
kfmenv = -1 ; default FM envelope (flat)  
icosine = giCosine ; cosine ftable  
kTrainCps = kgrainfreq ; set trainlet cps equal to grain
```



```

; rate for single-cycle trainlet in
; each grain
; number of partials in trainlet
; balance of partials in trainlet
; (default) no channel masking,
; all grains to output 1
; no random grain masking
; source waveforms
; mix of 4 source waveforms and
; trainlets (set to default)
; phase offset for reading source waveform
;
;
; original key for source waveform
;
;
; max grains per k period
; id of opcode, linking partikkel
; to partikkelsync

a1 partikkel kgrainfreq, kdistribution, idisttab, async, kenv2amt, \
    ienv2tab, ienv_attack, ienv_decay, ksustain_amount, ka_d_ratio, \
    kduration, kamp, igainmasks, kwavfreq, ksweepshape, \
    iwavfreqstarttab, iwavfreqendtab, awavfm, ifmamptab, kfmenv, \
    icosine, kTrainCps, knumpartials, kchroma, ichannelmasks, \
    krandommask, kwaveform1, kwaveform2, kwaveform3, kwaveform4, \
    iwaveamptab, asamplepos1, asamplepos2, asamplepos3, asamplepos4, \
    kwavekey1, kwavekey2, kwavekey3, kwavekey4, imax_grains, iopcode_id

async1 partikkelsync iopcode_id ; clock pulse output of the
; partikkel instance above
ksyncGravity line 0, p3, iMaxSync ; strength of synchronization
aphase2 init 0
asyncPolarity limit (int(aphase2*2)*2)-1, -1, 1
; use the phase of partikkelsync instance 2 to find sync
; polarity for partikkel instance 2.
; If the phase of instance 2 is less than 0.5, we want to
; nudge it down when synchronizing,
; and if the phase is > 0.5 we want to nudge it upwards.
async1 = async1*ksyncGravity*asyncPolarity ; prepare sync signal
; with polarity and strength

kgrainfreq2 = igrainrate * iosc2Dev ; grains per second for second partikkel instance
iopcode_id2 = 2
a2 partikkel kgrainfreq2, kdistribution, idisttab, async1, kenv2amt, \
    ienv2tab, ienv_attack, ienv_decay, ksustain_amount, ka_d_ratio, \
    kduration, kamp, igainmasks, kwavfreq, ksweepshape, \
    iwavfreqstarttab, iwavfreqendtab, awavfm, ifmamptab, kfmenv, \
    icosine, kTrainCps, knumpartials, kchroma, ichannelmasks, \
    krandommask, kwaveform1, kwaveform2, kwaveform3, kwaveform4, \
    iwaveamptab, asamplepos1, asamplepos2, asamplepos3, \
    asamplepos4, kwavekey1, kwavekey2, kwavekey3, kwavekey4, \
    imax_grains, iopcode_id2

async2, aphase2 partikkelsync iopcode_id2
; clock pulse and phase
; output of the partikkel instance above,
; we will only use the phase

outs a1*amp, a2*amp

endin

</CsInstruments>
<CsScore>

/*score parameters
igrainrate = p4 ; grain rate
igrainsize = p5 ; grain size in ms
igrainFreq = p6 ; frequency of source wave within grain
iosc2Dev = p7 ; partikkel instance 2 grain rate deviation factor
iMaxSync = p8 ; max soft sync amount (increasing to this value during length of note)
*/
; GrRate GrSize GrFund Osc2Dev MaxSync

i1 0 10 2 20 880 1.3 0.3
s
i1 0 10 5 20 440 0.8 0.3
s
i1 0 6 55 15 660 1.8 0.45
s
i1 0 6 110 10 440 0.6 0.6
s

```

```
i1 0      6 220 3 660 2.6 0.45
s
i1 0      6 220 3 660 2.1 0.45
s
i1 0      6 440 3 660 0.8 0.22
s
e
e
</CsScore>
</CsoundSynthesizer>
```

See Also

fof, fof2, fog, grain, grain2, grain3, granule, sndwarp, sndwarpst, syncgrain, syncloop, partikkel-sync

Credits

Author: Thom Johansen
Author: Torgeir Strand Henriksen
Author: Oeyvind Brandtsegg
April 2007

Examples written by Joachim Heintz and Oeyvind Brandtsegg.

New in version 5.06

partikkelsync

partikkelsync — Outputs *partikkel*'s grain scheduler clock pulse and phase to synchronize several instances of the *partikkel* opcode to the same clock source.

Description

partikkelsync is an opcode for outputting *partikkel*'s grain scheduler clock pulse and phase. *partikkelsync*'s output can be used to synchronize other instances of the *partikkel* opcode to the same clock.

Syntax

```
async [,aphase] partikkelsync iopcode_id
```

Initialization

iopcode_id -- the opcode id, linking an instance of *partikkel* to an instance of *partikkelsync*.

Performance

async -- trigger pulse signal. Outputs trigger pulses synchronized to a *partikkel* opcode's grain scheduler clock. One trigger pulse is generated for each grain started in the *partikkel* opcode with the same *opcode_id*. The normal usage would be to send this signal to another *partikkel* opcode's *async* input to synchronize several instances of *partikkel*.

aphase -- clock phase. Outputs a linear ramping phase signal. Can be used e.g. for softsynchronization, or just as a phase generator ala *phasor*.

See Also

partikkel

Credits

Author: Thom Johansen
Author: Torgeir Strand Henriksen
Author: Oeyvind Brandtsegg
April 2007

New in version 5.06

pcauchy

pcauchy — Cauchy distribution random number generator (positive values only).

Description

Cauchy distribution random number generator (positive values only). This is an x-class noise generator.

Syntax

ares **pcauchy** kalpha

ires **pcauchy** kalpha

kres **pcauchy** kalpha

Performance

pcauchy kalpha -- controls the spread from zero (big kalpha = big spread). Outputs positive numbers only.

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the pcauchy opcode. It uses the file *pcauchy.csd* [examples/pcauchy.csd].

Example 344. Example of the pcauchy opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o pcauchy.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number between 0 and 1.
; kalpha = 1
```

```
i1 pcauchy 1
print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1: i1 = 0.012
```

See Also

seed, betarand, bexpnd, cauchy, exprand, gauss, linrand, poisson, trirand, unirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder.

pchbend

pchbend — Get the current pitch-bend value for this channel.

Description

Get the current pitch-bend value for this channel.

Syntax

```
ibend pchbend [imin] [, imax]
```

```
kbend pchbend [imin] [, imax]
```

Initialization

imin, *imax* (optional) -- set minimum and maximum limits on values obtained

Performance

Get the current pitch-bend value for this channel. Note that this access to pitch-bend data is independent of the MIDI pitch, enabling the value here to be used for any arbitrary purpose.

Examples

Here is an example of the pchbend opcode. It uses the file *pchbend.csd* [examples/pchbend.csd].

Example 345. Example of the pchbend opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac          -iadc     -d         -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o pchbend.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 pchbend

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 12 seconds.
i 1 0 12
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

aftouch, ampmidi, cpsmidi, cpsmidib, midictrl, notnum, octmidi, octmidib, pchmidi, pchmidib, veloc

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

pchmidi

pchmidi — Get the note number of the current MIDI event, expressed in pitch-class units.

Description

Get the note number of the current MIDI event, expressed in pitch-class units.

Syntax

ipch **pchmidi**

Performance

Get the note number of the current MIDI event, expressed in pitch-class units for local processing.



pchmidi vs. pchmidinn

The *pchmidi* opcode only produces meaningful results in a Midi-activated note (either real-time or from a Midi score with the -F flag). With *pchmidi*, the Midi note number value is taken from the Midi event that is internally associated with the instrument instance. On the other hand, the *pchmidinn* opcode may be used in any Csound instrument instance whether it is activated from a Midi event, score event, line event, or from another instrument. The input value for *pchmidinn* might for example come from a p-field in a textual score or it may have been retrieved from the real-time Midi event that activated the current note using the *notnum* opcode.

Examples

Here is an example of the pchmidi opcode. It uses the file *pchmidi.csd* [examples/pchmidi.csd].

Example 346. Example of the pchmidi opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc     -d         -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o pchmidi.wav -W   ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; This example expects MIDI note inputs on channel 1
i1 pchmidi

print i1
endin
```



```
</CsInstruments>
<CsScore>

;Dummy f-table to give time for real-time MIDI events
f 0 8000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

aftouch, ampmidi, cpsmidi, cpsmidib, midictrl, notnum, octmidi, octmidib, pchbend, pchmidib, veloc, cpsmidinn, octmidinn, pchmidinn

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

pchmidib

pchmidib — Get the note number of the current MIDI event and modify it by the current pitch-bend value, express it in pitch-class units.

Description

Get the note number of the current MIDI event and modify it by the current pitch-bend value, express it in pitch-class units.

Syntax

```
ipch pchmidib [irange]
```

```
kpch pchmidib [irange]
```

Initialization

irange (optional) -- the pitch bend range in semitones

Performance

Get the note number of the current MIDI event, modify it by the current pitch-bend value, and express the result in pitch-class units. Available as an i-time value or as a continuous k-rate value.

Examples

Here is an example of the pchmidib pchmidib. It uses the file *pchmidib.csd* [examples/pchmidib.csd].

Example 347. Example of the pchmidib pchmidib.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d           -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o pchmidib.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; This example expects MIDI note inputs on channel 1
il pchmidib

    print il
endin

</CsInstruments>
```

```
<CsScore>  
  
;Dummy f-table to give time for real-time MIDI events  
f 0 8000  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

aftouch, ampmidi, cpsmidi, cpsmidib, midictrl, notnum, octmidi, octmidib, pchbend, pchmidi, veloc

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

pchmidinn

pchmidinn — Converts a Midi note number value to octave point pitch-class units.

Description

Converts a Midi note number value to octave point pitch-class units.

Syntax

```
pchmidinn (MidiNoteNumber) (init- or control-rate args only)
```

where the argument within the parentheses may be a further expression.

Performance

pchmidinn is a function that takes an i-rate or k-rate value representing a Midi note number and returns the equivalent pitch value in Csound's octave point pitch-class format. This conversion assumes that Middle C (8.00 in *pch*) is Midi note number 60. Midi note number values are typically integers in the range from 0 to 127 but fractional values or values outside of this range will be interpreted consistently.



pchmidinn vs. pchmidi

The *pchmidinn* opcode may be used in any Csound instrument instance whether it is activated from a Midi event, score event, line event, or from another instrument. The input value for *pchmidinn* might for example come from a p-field in a textual score or it may have been retrieved from the real-time Midi event that activated the current note using the *notnum* opcode. You must specify an i-rate or k-rate expression for the Midi note number that is to be converted. On the other hand, the *pchmidi* opcode only produces meaningful results in a Midi-activated note (either real-time or from a Midi score with the -F flag). With *pchmidi*, the Midi note number value is taken from the Midi event associated with the instrument instance, and no location or expression for this value may be specified.

pchmidinn and its related opcodes are really *value converters* with a special function of manipulating pitch data.

Data concerning pitch and frequency can exist in any of the following forms:

Table 16. Pitch and Frequency Values

Name	Abbreviation
octave point pitch-class (8ve.pc)	pch
octave point decimal	oct
cycles per second	cps
Midi note number (0-127)	midinn

The first two forms consist of a whole number, representing octave registration, followed by a specially interpreted fractional part. For *pch*, the fraction is read as two decimal digits representing the 12 equal-tempered pitch classes from .00 for C to .11 for B. For *oct*, the fraction is interpreted as a true decimal fractional part of an octave. The two fractional forms are thus related by the factor 100/12. In both forms, the fraction is preceded by a whole number octave index such that 8.00 rep-

resents Middle C, 9.00 the C above, etc. Midi note number values range between 0 and 127 (inclusively) with 60 representing Middle C, and are usually whole numbers. Thus A440 can be represented alternatively by 440 (*cps*), 69 (*midinn*), 8.09 (*pch*), or 8.75 (*oct*). Microtonal divisions of the *pch* semitone can be encoded by using more than two decimal places.

The mnemonics of the pitch conversion units are derived from morphemes of the forms involved, the second morpheme describing the source and the first morpheme the object (result). Thus *cpspch*(8.09) will convert the pitch argument 8.09 to its *cps* (or Hertz) equivalent, giving the value of 440. Since the argument is constant over the duration of the note, this conversion will take place at i-time, before any samples for the current note are produced.

By contrast, the conversion *cpsoct*(8.75 + *k1*) which gives the value of A440 transposed by the octave interval *k1*. The calculation will be repeated every *k*-period since that is the rate at which *k1* varies.



Note

The conversion from *pch*, *oct*, or *midinn* into *cps* is not a linear operation but involves an exponential process that could be time-consuming when executed repeatedly. Csound now uses a built-in table lookup to do this efficiently, even at audio rates. Because the table index is truncated without interpolation, pitch resolution when using one of these opcodes is limited to 8192 discrete and equal divisions of the octave, and some pitches of the standard 12-tone equally-tempered scale are very slightly mistuned (by at most 0.15 cents).

Examples

Here is an example of the *pchmidinn* opcode. It uses the file *cpsmidinn.csd* [examples/cpsmidinn.csd].

Example 348. Example of the *pchmidinn* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform.
; This example produces no audio, so we render in
; non-realtime and turn off sound to disk:
-n
</CsOptions>
<CsInstruments>

instr 1
; i-time loop to print conversion table
imidiNN = 0
loop1:
  icps = cpsmidinn(imidiNN)
  ioct = octmidinn(imidiNN)
  ipch = pchmidinn(imidiNN)

  print imidiNN, icps, ioct, ipch

  imidiNN = imidiNN + 1
  if (imidiNN < 128) igoto loop1
endin

instr 2
; test k-rate converters
kMiddleC = 60
kcps = cpsmidinn(kMiddleC)
koct = octmidinn(kMiddleC)
kpch = pchmidinn(kMiddleC)

printks "%d %f %f %f\n", 1.0, kMiddleC, kcps, koct, kpch
endin

</CsInstruments>
```

```
<CsScore>  
i1 0 0  
i2 0 0.1  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

See Also

cpsmidinn, octmidinn, pchmidi, notnum, cpspch, cpsoct, octcps, octpch, pchoct

Credits

Derived from original value converters by Barry Vercoe.

New in version 5.07

pchoct

pchoct — Converts an octave-point-decimal value to pitch-class.

Description

Converts an octave-point-decimal value to pitch-class.

Syntax

```
pchoct (oct) (init- or control-rate args only)
```

where the argument within the parentheses may be a further expression.

Performance

pchoct and its related opcodes are really *value converters* with a special function of manipulating pitch data.

Data concerning pitch and frequency can exist in any of the following forms:

Table 17. Pitch and Frequency Values

Name	Abbreviation
octave point pitch-class (8ve.pc)	pch
octave point decimal	oct
cycles per second	cps
Midi note number (0-127)	midinn

The first two forms consist of a whole number, representing octave registration, followed by a specially interpreted fractional part. For *pch*, the fraction is read as two decimal digits representing the 12 equal-tempered pitch classes from .00 for C to .11 for B. For *oct*, the fraction is interpreted as a true decimal fractional part of an octave. The two fractional forms are thus related by the factor 100/12. In both forms, the fraction is preceded by a whole number octave index such that 8.00 represents Middle C, 9.00 the C above, etc. Midi note number values range between 0 and 127 (inclusively) with 60 representing Middle C, and are usually whole numbers. Thus A440 can be represented alternatively by 440 (*cps*), 69 (*midinn*), 8.09 (*pch*), or 8.75 (*oct*). Microtonal divisions of the *pch* semitone can be encoded by using more than two decimal places.

The mnemonics of the pitch conversion units are derived from morphemes of the forms involved, the second morpheme describing the source and the first morpheme the object (result). Thus *cpspch*(8.09) will convert the pitch argument 8.09 to its *cps* (or Hertz) equivalent, giving the value of 440. Since the argument is constant over the duration of the note, this conversion will take place at i-time, before any samples for the current note are produced.

By contrast, the conversion *cpsoct*(8.75 + k1) which gives the value of A440 transposed by the octave interval *k1*. The calculation will be repeated every k-period since that is the rate at which *k1* varies.



Note

The conversion from *pch*, *oct*, or *midinn* into *cps* is not a linear operation but involves an exponential process that could be time-consuming when executed repeatedly. Csound now uses a built-in table lookup to do this efficiently, even at audio rates. Be-

cause the table index is truncated without interpolation, pitch resolution when using one of these opcodes is limited to 8192 discrete and equal divisions of the octave, and some pitches of the standard 12-tone equally-tempered scale are very slightly mistuned (by at most 0.15 cents).

Examples

Here is an example of the `pchoct` opcode. It uses the file `pchoct.csd` [examples/pchoct.csd].

Example 349. Example of the `pchoct` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o pchoct.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Convert an octave-point-decimal value into a
; pitch-class value.
ioct = 8.75
ipch = pchoct(ioct)

print ipch
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1: ipch = 8.090
```

See Also

cpsoct, *cpspch*, *octcps*, *octpch*, *cpsmidinn*, *octmidinn*, *pchmidinn*

Credits

Example written by Kevin Conder.

pconvolve

convolve — Convolution based on a uniformly partitioned overlap-save algorithm

Description

Convolution based on a uniformly partitioned overlap-save algorithm. Compared to the *convolve* opcode, 'pconvolve' has these benefits:

- small delay
- possible to run in real-time for shorter impulse files
- no pre-process analysis pass
- can often render faster than convolve

Syntax

```
ar1 [, ar2] [, ar3] [, ar4] pconvolve ain, ifilcod [, ipartitionsizes, ichannel]
```

Initialization

ifilcod -- integer or character-string denoting an impulse response soundfile. multichannel files are supported, the file must have the same sample-rate as the orc. [Note: cval files cannot be used!] Keep in mind that longer files require more calculation time [and probably larger partition sizes and more latency]. At current processor speeds, files longer than a few seconds may not render in real-time.

ipartitionsizes (optional, defaults to the output buffersize [-b]) -- the size in samples of each partition of the impulse file. This is the parameter that needs tweaking for best performance depending on the impulse file size. Generally, a small size means smaller latency but more computation time. If you specify a value that is not a power-of-2 the opcode will find the next power-of-2 greater and use that as the actual partition size.

ichannel (optional) -- which channel to use from the impulse response data file.

Performance

ain -- input audio signal.

The overall latency of the opcode can be calculated as such [assuming ipartitionsizes is a power of 2]

```
latency = (ksmps < ipartitionsizes ? ipartitionsizes + ksmps : ipartitionsizes)/sr
```

Examples

Instrument 1 shows an example of real-time convolution.

Instrument 2 shows how to do file-based convolution with a 'look ahead' method to remove all delay.



NOTE

You will need to download the impulse response files from noisevault.com or replace the filenames with your own impulse files

```
sr = 44100
ksmps = 100
nchnls = 2

instr 1
kmix = .5 ; Wet/dry mix. Vary as desired.
kvol = .5*kmix ; Overall volume level of reverb. May need to adjust
               ; when wet/dry mix is changed, to avoid clipping.

; do some safety checking to make sure we the parameters a good
kmix = (kmix < 0 || kmix > 1 ? .5 : kmix)
kvol = (kvol < 0 ? 0 : .5*kvol*kmix)

; size of each convolution partion -- for best performance, this parameter needs to be tweaked
ipartitionsize = p4

; calculate latency of pconvolve opcode
idel = (ksmps < ipartitionsize ? ipartitionsize + ksmps : ipartitionsize)/sr
prints "Convoluting with a latency of %f seconds%n", idel

; actual processing
al, ar ins

awetl, awetr pconvolve kvol*(al+ar), "Mercedes-van.wav", ipartitionsize

; Delay dry signal, to align it with the convoled sig
adryl delay (1-kmix)*al, idel
adryr delay (1-kmix)*ar, idel

outs adryl+awetl, adryr+awetr

endin

instr 2
imix = 0.5 ; Wet/dry mix. Vary as desired.
ivol = .5*imix ; Overall volume level of reverb. May need to adjust
               ; when wet/dry mix is changed, to avoid clipping.

ipartitionsize = 32768 ; size of each convolution partion
idel = (ksmps < ipartitionsize ? ipartitionsize + ksmps : ipartitionsize)/sr ; latency of pco

kcount init idel*kr

; since we are using a soundin [instead of ins] we can
; do a kind of "look ahead" by looping during one k-pass
; without output, creating zero-latency

loop:
al, ar soundin "John_Cage_1.aif", 0

awetl, awetr pconvolve ivol*(al+ar), "FactoryHall.aif", ipartitionsize

adryl delay (1-imix)*al, idel ; Delay dry signal, to align it with
adryr delay (1-imix)*ar, idel ;

kcount = kcount - 1
if kcount > 0 kgoto loop

outs awetl+adryl, awetr+adryr

endin
```

See also

convolve, *dconv*.

Credits

Author: Matt Ingalls
2004

pcount

pcount — Returns the number of pfields belonging to a note event.

Description

pcount returns the number of pfields belonging to a note event.

Syntax

icount **pcount**

Initialization

icount - stores the number of pfields for the current note event.



Note

Note that the reported number of pfields is not necessarily what's explicitly written in the score, but the pfields available to the instrument through mechanisms like *pfield carry*.

Examples

Here is an example of the pcount opcode. It uses the file *pcount.csd* [examples/pcount.csd].

Example 350. Example of the pcount opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac      -iadc      ; -d          -MO      ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
;-o pcount.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
;Example by Anthony Kozar Dec 2006
instr 1
  inum pcount
  print inum
endin

</CsInstruments>
<CsScore>
i1  0 3 4 5      ; has 5 pfields
i1  1 3          ; has 5 due to carry
i1  2 3 4 5 6 7  ; has 7
e
</CsScore>
</CsoundSynthesizer>
```

The example will produce the following output:

SECTION 1:

```
new alloc for instr 1:
WARNING: instr 1 uses 3 p-fields but is given 5
instr 1:  inum = 5.000
B 0.000 .. 1.000 T 1.000 TT 1.000 M:      0.0
new alloc for instr 1:
WARNING: instr 1 uses 3 p-fields but is given 5
instr 1:  inum = 5.000
B 1.000 .. 2.000 T 2.000 TT 2.000 M:      0.0
new alloc for instr 1:
WARNING: instr 1 uses 3 p-fields but is given 7
instr 1:  inum = 7.000
```

The warnings occur because pfields are not used explicitly by the instrument.

See Also

pindex

Credits

Example by: Anthony Kozar

Dec. 2006

pdclip

pdclip — Performs linear clipping on an audio signal or a phasor.

Description

The *pdclip* opcode allows a percentage of the input range of a signal to be clipped to fullscale. It is similar to simply multiplying the signal and limiting the range of the result, but *pdclip* allows you to think about how much of the signal range is being distorted instead of the scalar factor and has a offset parameter for assymetric clipping of the signal range. *pdclip* is also useful for remapping phasors for phase distortion synthesis.

Syntax

```
aout pdclip ain, kWidth, kCenter [, ibipolar [, ifullscale]]
```

Initialization

ibipolar -- an optional parameter specifying either unipolar (0) or bipolar (1) mode. Defaults to unipolar mode.

ifullscale -- an optional parameter specifying the range of input and output values. The maximum will be *ifullscale*. The minimum depends on the mode of operation: zero for unipolar or *-ifullscale* for bipolar. Defaults to 1.0 -- you should set this parameter to the maximum expected input value.

Performance

ain -- the input signal to be clipped.

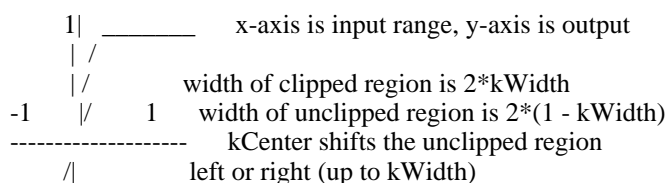
aout -- the output signal.

kWidth -- the percentage of the signal range that is clipped (must be between 0 and 1).

kCenter -- an offset for shifting the unclipped window of the signal higher or lower in the range (essentially a DC offset). Values should be in the range [-1, 1] with a value of zero representing no shift (regardless of whether bipolar or unipolar mode is used).

The *pdclip* opcode performs linear clipping on the input signal *ain*. *kWidth* specifies the percentage of the signal range that is clipped. The rest of the input range is mapped linearly from zero to ifullscale in unipolar mode and from -ifullscale to ifullscale in bipolar mode. When *kCenter* is zero, equal amounts of the top and bottom of the signal range are clipped. A negative value shifts the unclipped range more towards the bottom of the input range and a positive value shifts it more towards the top. *ibipolar* should be 1 for bipolar operation and 0 for unipolar mode. The default is unipolar mode (*ibipolar* = 0). *ifullscale* sets the maximum amplitude of the input and output signals (defaults to 1.0).

This amounts to waveshaping the input with the following transfer function (normalized to ifullscale=1.0 in bipolar mode):



```
    /|  
    /|  
-----|-1
```

Bipolar mode can be used for direct, linear distortion of an audio signal. Alternatively, unipolar mode is useful for modifying the output of a phasor before it is used to index a function table, effectively making this a phase distortion technique.

See Also

pdhalf, *pdhalfy*, *limit*, *clip*, *distort1*

Examples

Here is an example of the *pdclip* opcode. It uses the file *pdclip.csd* [examples/pdclip.csd].

Example 351. Example of the *pdclip* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out   Audio in  
-odac         -iadc      ;;RT audio I/O  
; For Non-realtime ouput leave only the line below:  
; -o abs.wav -W ;; for file output any platform  
</CsOptions>  
<CsInstruments>  
  
; test instrument for pdclip opcode  
instr 3  
  
    idur          = p3  
    iamp          = p4  
    ifreq         = p5  
    ifn           = p6  
  
    kenv          linseg      0, .05, 1.0, idur - .1, 1.0, .05, 0  
    aosc          oscil      1.0, ifreq, ifn  
  
    kmod          expseg      0.00001, idur, 1.0  
    aout          pdclip     aosc, kmod, 0.0, 1.0  
  
                                out      kenv*aout*iamp  
  
endin  
  
</CsInstruments>  
<CsScore>  
f1 0 16385 10 1  
f2 0 16385 10 1 .5 .3333 .25 .5  
  
; pdclipped sine wave  
i3 0 3 15000 440 1  
i3 + 3 15000 330 1  
i3 + 3 15000 220 1  
s  
  
; pdclipped composite wave  
i3 0 3 15000 440 2  
i3 + 3 15000 330 2  
i3 + 3 15000 220 2  
e  
</CsScore>  
</CsoundSynthesizer>
```

Credits

Author: Anthony Kozar
January 2008

New in Csound version 5.08

pdhalf

pdhalf — Distorts a phasor for reading the two halves of a table at different rates.

Description

The *pdhalf* opcode is designed to emulate the "classic" phase distortion synthesis method of the Casio CZ-series of synthesizers from the mid-1980's. This technique reads the first and second halves of a function table at different rates in order to warp the waveform. For example, *pdhalf* can smoothly transform a sine wave into something approximating the shape of a saw wave.

Syntax

```
aout pdhalf ain, kShapeAmount [, ibipolar [, ifullscale]]
```

Initialization

ibipolar -- an optional parameter specifying either unipolar (0) or bipolar (1) mode. Defaults to unipolar mode.

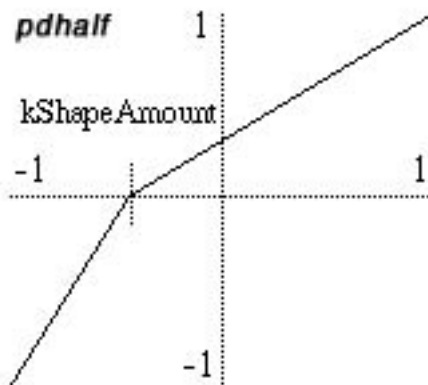
ifullscale -- an optional parameter specifying the range of input and output values. The maximum will be *ifullscale*. The minimum depends on the mode of operation: zero for unipolar or *-ifullscale* for bipolar. Defaults to 1.0 -- you should set this parameter to the maximum expected input value.

Performance

ain -- the input signal to be distorted.

aout -- the output signal.

kShapeAmount -- the amount of distortion applied to the input. Must be between negative one and one (-1 to 1). An amount of zero means no distortion.



Transfer function created by *pdhalf* and a negative *kShapeAmount*.

The *pdhalf* opcode calculates a transfer function that is composed of two linear segments (see the graph). These segments meet at a "pivot point" which always lies on the same horizontal axis. (In unipolar mode, the axis is $y = 0.5$, and for bipolar mode it is the x axis). The *kShapeAmount* parameter specifies where on the horizontal axis this point falls. When *kShapeAmount* is zero, the pivot point is in the middle of the input range, forming a straight line for the transfer function and thus causing no change in the input signal. As *kShapeAmount* changes from zero (0) to negative one (-1), the pivot point moves towards the left side of the graph, producing a phase distortion pattern like the Casio CZ's "sawtooth waveform". As it changes from zero (0) to positive one (1), the pivot point

moves toward the right, producing an inverted pattern.

If the input to *pdhalf* is a phasor and the output is used to index a table, values for *kShapeAmount* that are less than zero will cause the first half of the table to be read more quickly than the second half. The reverse is true for values of *kShapeAmount* greater than zero. The rates at which the halves are read are calculated so that the frequency of the phasor is unchanged. Thus, this method of phase distortion can only produce higher partials in a harmonic series. It cannot produce inharmonic sidebands in the way that frequency modulation does.

pdhalf can work in either unipolar or bipolar modes. Unipolar mode is appropriate for signals like phasors that range between zero and some maximum value (selectable with *ifullscale*). Bipolar mode is appropriate for signals that range above and below zero by roughly equal amounts such as most audio signals. Applying *pdhalf* directly to an audio signal in this way results in a crude but adjustable sort of waveshaping/distortion.

A typical example of the use of *pdhalf* is

```
aphase    phasor    ifreq
apd       pdhalf    aphase, kamount
aout      tablei    apd, 1, 1
```

More information about Phase distortion synthesis can be found on Wikipedia at http://en.wikipedia.org/wiki/Phase_distortion_synthesis [ht-
[tp://en.wikipedia.org/wiki/Phase_distortion_synthesis](http://en.wikipedia.org/wiki/Phase_distortion_synthesis)]

See Also

pdhalfy, *pdclip*

Examples

Here is an example of the *pdhalf* opcode. It uses the file *pdhalf.csd* [examples/*pdhalf.csd*].

Example 352. Example of the *pdhalf* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o abs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
; test instrument for pdhalf opcode
instr 4

    idur      = p3
    iamp      = p4
    ifreq     = p5
    itable    = p6

    aenv      linseg      0, .001, 1.0, idur - .051, 1.0, .05, 0
    aosc      phasor      ifreq
    kamount    linseg      0.0, 0.02, -0.99, 0.05, -0.9, idur-0.06, 0.0
    apd       pdhalf      aosc, kamount
    aout      tablei      apd, itable, 1

                                out      aenv*aout*iamp
endin
</CsInstruments>
```

```
<CsScore>
f1 0 16385 10 1
f2 0 16385 10 1 .5 .3333 .25 .5
f3 0 16385 9 1 1 270 ; inverted cosine

; descending "just blues" scale

; pdhalf with cosine table
; (imitates the CZ-101 "sawtooth waveform")
t 0 100
i4 0 .333 10000 512 3
i. + . . 448
i. + . . 384
i. + . . 358.4
i. + . . 341.33
i. + . . 298.67
i. + 2 . 256
e
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Anthony Kozar
January 2008

New in Csound version 5.08

pdhalfy

pdhalfy — Distorts a phasor for reading two unequal portions of a table in equal periods.

Description

The *pdhalfy* opcode is a variation on the phase distortion synthesis method of the *pdhalf* opcode. It is useful for distorting a phasor in order to read two unequal portions of a table in the same number of samples.

Syntax

```
out pdhalfy in, kShapeAmount [, ibipolar [, ifullscale]]
```

Initialization

ibipolar -- an optional parameter specifying either unipolar (0) or bipolar (1) mode. Defaults to unipolar mode.

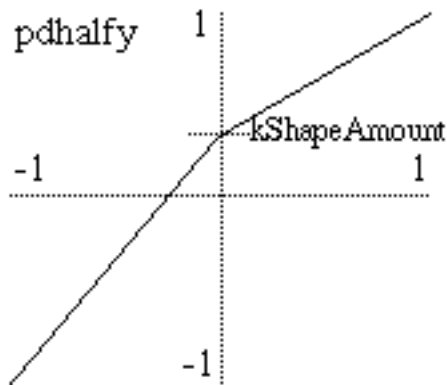
ifullscale -- an optional parameter specifying the range of input and output values. The maximum will be *ifullscale*. The minimum depends on the mode of operation: zero for unipolar or *-ifullscale* for bipolar. Defaults to 1.0 -- you should set this parameter to the maximum expected input value.

Performance

ain -- the input signal to be distorted.

out -- the output signal.

kShapeAmount -- the amount of distortion applied to the input. Must be between negative one and one (-1 to 1). An amount of zero means no distortion.



Transfer function created by *pdhalfy* and a negative *kShapeAmount*.

The *pdhalfy* opcode calculates a transfer function that is composed of two linear segments (see the graph). These segments meet at a "pivot point" which always lies on the same vertical axis. (In unipolar mode, the axis is $x = 0.5$, and for bipolar mode it is the y axis). So, *pdhalfy* is a variation of the *pdhalf* opcode that places the pivot point of the phase distortion pattern on a vertical axis instead of a horizontal axis.

The *kShapeAmount* parameter specifies where on the vertical axis this point falls. When *kShapeAmount* is zero, the pivot point is in the middle of the output range, forming a straight line

for the transfer function and thus causing no change in the input signal. As *kShapeAmount* changes from zero (0) to negative one (-1), the pivot point downward towards the bottom of the graph. As it changes from zero (0) to positive one (1), the pivot point moves upward, producing an inverted pattern.

If the input to *pdhalfy* is a phasor and the output is used to index a table, the use of *pdhalfy* will divide the table into two segments of different sizes with each segment being mapped to half of the oscillator period. Values for *kShapeAmount* that are less than zero will cause less than half of the table to be read in the first half of the period of oscillation. The rest of the table will be read in the second half of the period. The reverse is true for values of *kShapeAmount* greater than zero. Note that the frequency of the phasor is always unchanged. Thus, this method of phase distortion can only produce higher partials in a harmonic series. It cannot produce inharmonic sidebands in the way that frequency modulation does. *pdhalfy* tends to have a milder quality to its distortion than *pdhalf*.

pdhalfy can work in either unipolar or bipolar modes. Unipolar mode is appropriate for signals like phasors that range between zero and some maximum value (selectable with *ifullscale*). Bipolar mode is appropriate for signals that range above and below zero by roughly equal amounts such as most audio signals. Applying *pdhalfy* directly to an audio signal in this way results in a crude but adjustable sort of waveshaping/distortion.

A typical example of the use of *pdhalfy* is

```
aphase    phasor    ifreq
apd       pdhalfy   aphase, kamount
aout      tablei    apd, 1, 1
```

More information about Phase distortion synthesis can be found on Wikipedia at http://en.wikipedia.org/wiki/Phase_distortion_synthesis [http://en.wikipedia.org/wiki/Phase_distortion_synthesis]

See Also

pdhalf, *pdclip*

Examples

Here is an example of the *pdhalfy* opcode. It uses the file *pdhalfy.csd* [examples/*pdhalfy.csd*].

Example 353. Example of the *pdhalfy* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o abs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
; test instrument for pdhalfy opcode
instr 5

    idur          = p3
    iamp          = p4
    ifreq         = p5
    iamtinit      = p6      ; initial amount of phase distortion
    iatt          = p7      ; attack time
    isuslvl       = p8      ; sustain amplitude
    idistdec      = p9      ; time for distortion amount to reach zero
    itable        = p10
```

```
idec      = idistdec - iatt
irel      = .05
isus      = idur - (idistdec + irel)

aenv      linseg      0, iatt, 1.0, idec, isuslvl, isus, isuslvl, irel, 0, 0, 0
kamount   linseg      -iamtinit, idistdec, 0.0, idur-idistdec, 0.0
aosc      phasor
apd       pdhalfy     aosc, kamount
aout      tablei      apd, itable, 1

out        aenv*aout*iamp

endin

</CsInstruments>
<CsScore>
f1 0 16385 10 1          ; sine
f3 0 16385 9 1 1 270     ; inverted cosine

; descending "just blues" scale

; pdhalfy with cosine table
t 0 100
i5 0 .333 10000 512      1.0 .02 0.5 .12 3
i. + . . 448 <
i. + . . 384 <
i. + . . 358.4 <
i. + . . 341.33 <
i. + . . 298.67 <
i. + 2 . 256 0.5
s

; pdhalfy with sine table
t 0 100
i5 0 .333 10000 512      1.0 .001 0.1 .07 1
i. + . . 448 <
i. + . . 384 <
i. + . . 358.4 <
i. + . . 341.33 <
i. + . . 298.67 <
i. + 2 . 256 0.5
e
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Anthony Kozar
January 2008

New in Csound version 5.08

peak

peak — Maintains the output equal to the highest absolute value received.

Description

These opcodes maintain the output k-rate variable as the peak absolute level so far received.

Syntax

```
kres peak asig
```

```
kres peak ksig
```

Performance

kres -- Output equal to the highest absolute value received so far. This is effectively an input to the opcode as well, since it reads *kres* in order to decide whether to write something higher into it.

ksig -- k-rate input signal.

asig -- a-rate input signal.

Examples

Here is an example of the peak opcode. It uses the file *peak.csd* [examples/peak.csd], and *beats.wav* [examples/beats.wav].

Example 354. Example of the peak opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o peak.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1 - play an audio file.
instr 1
; Capture the highest amplitude in the "beats.wav" file.
asig soundin "beats.wav"
kp peak asig

; Print out the peak value once per second.
printk 1, kp

out asig
endin

</CsInstruments>
```

```
<CsScore>

; Play Instrument #1, the audio file, for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
i 1 time 0.00002: 4835.00000
i 1 time 1.00002: 29312.00000
i 1 time 2.00002: 32767.00000
```

Credits

Author: Robin Whittle
Australia
May 1997

Example written by Kevin Conder.

peakk

peakk — Deprecated.

Description

Deprecated as of version 3.63. Use the *peak* opcode instead.

pgmassign

pgmassign — Assigns an instrument number to a specified MIDI program.

Description

Assigns an instrument number to a specified (or all) MIDI program(s).

By default, the instrument is the same as the program number. If the selected instrument is zero or negative or does not exist, the program change is ignored. This opcode is normally used in the orchestra header. Although, like *massign*, it also works in instruments.

Syntax

```
pgmassign ipgm, inst[, ichn]
```

```
pgmassign ipgm, "insname"[, ichn]
```

Initialization

ipgm -- MIDI program number (1 to 128). A value of zero selects all programs.

inst -- instrument number. If set to zero, or negative, MIDI program changes to *ipgm* are ignored. Currently, assignment to an instrument that does not exist has the same effect. This may be changed in a later release to print an error message.

"insname" -- A string (in double-quotes) representing a named instrument.

"ichn" (optional, defaults to zero) -- channel number. If zero, program changes are assigned on all channels.

You can disable the turning on of any instruments by using the following in the header:

```
massign 0, 0  
pgmassign 0, 0
```

Examples

Here is an example of the *pgmassign* opcode. It uses the file *pgmassign.csd* [examples/pg-massign.csd].

Example 355. Example of the *pgmassign* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsoundOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  No messages  MIDI in  
-odac      -iadc      -d      -M0    ;;RT audio I/O with MIDI in  
; For Non-realtime output leave only the line below:  
; -o pgmassign.wav -W ;; for file output any platform  
</CsoundOptions>  
<CsoundInstruments>  
  
; Initialize the global variables.
```

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Program 55 (synth vox) uses Instrument #10.
pgmassign 55, 10

; Instrument #10.
instr 10
; Just an example, no working code in here!
endin

</CsInstruments>
<CsScore>

; Play Instrument #10 for one second.
i 10 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the `pgmassign` opcode that will ignore program change events. It uses the file `pgmassign_ignore.csd` [examples/pgmassign_ignore.csd].

Example 356. Example of the `pgmassign` opcode that will ignore program change events.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac      -iadc      -d      -M0      ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o pgmassign_ignore.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Ignore all program change events.
pgmassign 0, -1

; Instrument #1.
instr 1
; Just an example, no working code in here!
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Here is an advanced example of the `pgmassign` opcode. It uses the file `pgmassign_advanced.csd` [examples/pgmassign_advanced.csd].

Don't forget that you must include the `-F` flag when using an external MIDI file like “pgmassign_advanced.mid”.

Example 357. An advanced example of the pgmassign opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d          -M0    ;;RT audio I/O with MIDI in
; For Non-realtime output leave only the line below:
; -o pgmassign_advanced.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 10
nchnls = 1

    massign 1, 1 ; channels 1 to 4 use instr 1 by default
    massign 2, 1
    massign 3, 1
    massign 4, 1

; pgmassign.mid has 4 notes with these parameters:
;
;   Start time Channel Program
;
; note 1 0.5      1      10
; note 2 1.5      2      11
; note 3 2.5      3      12
; note 4 3.5      4      13

    pgmassign 0, 0          ; disable program changes
    pgmassign 11, 3         ; program 11 uses instr 3
    pgmassign 12, 2         ; program 12 uses instr 2

; waveforms for instruments
itmp ftgen 1, 0, 1024, 10, 1
itmp ftgen 2, 0, 1024, 10, 1, 0.5, 0.3333, 0.25, 0.2, 0.1667, 0.1429, 0.125
itmp ftgen 3, 0, 1024, 10, 1, 0, 0.3333, 0, 0.2, 0, 0.1429, 0, 0.10101

    instr 1                /* sine */

kcps cpsmidib 2 ; note frequency
asnd oscili 30000, kcps, 1
    out asnd

    endin

    instr 2                /* band-limited sawtooth */

kcps cpsmidib 2 ; note frequency
asnd oscili 30000, kcps, 2
    out asnd

    endin

    instr 3                /* band-limited square */

kcps cpsmidib 2 ; note frequency
asnd oscili 30000, kcps, 3
    out asnd

    endin

</CsInstruments>
<CsScore>

t 0 120
f 0 8.5 2 -2 0
e

</CsScore>
</CsoundSynthesizer>
```

See Also

midichn and *massign*

Credits

Author: Istvan Varga
May 2002

New in version 4.20

phaser1

phaser1 — First-order allpass filters arranged in a series.

Description

An implementation of *iord* number of first-order allpass filters in series.

Syntax

```
ares phaser1 asig, kfreq, kord, kfeedback [, iskip]
```

Initialization

iskip (optional, default=0) -- used to control initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

kfreq -- frequency (in Hz) of the filter(s). This is the frequency at which each filter in the series shifts its input by 90 degrees.

kord -- the number of allpass stages in series. These are first-order filters and can range from 1 to 4999.



Note

Although *kord* is listed as k-rate, it is in fact accessed only at init-time. So if you are using a k-rate argument, it must be assigned with *init*.

kfeedback -- amount of the output which is fed back into the input of the allpass chain. With larger amounts of feedback, more prominent notches appear in the spectrum of the output. *kfeedback* must be between -1 and +1. for stability.

phaser1 implements *iord* number of first-order allpass sections, serially connected, all sharing the same coefficient. Each allpass section can be represented by the following difference equation:

$$y(n) = C * x(n) + x(n-1) - C * y(n-1)$$

where $x(n)$ is the input, $x(n-1)$ is the previous input, $y(n)$ is the output, $y(n-1)$ is the previous output, and C is a coefficient which is calculated from the value of *kfreq*, using the bilinear z-transform.

By slowly varying *kfreq*, and mixing the output of the allpass chain with the input, the classic "phase shifter" effect is created, with notches moving up and down in frequency. This works best with *iord* between 4 and 16. When the input to the allpass chain is mixed with the output, 1 notch is generated for every 2 allpass stages, so that with *iord* = 6, there will be 3 notches in the output. With higher values for *iord*, modulating *kfreq* will result in a form of nonlinear pitch modulation.

Examples

Here is an example of the phaser1 opcode. It uses the file *phaser1.csd* [examples/phaser1.csd].

Example 358. Example of the phaser1 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o phaser1.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; demonstration of phase shifting abilities of phaser1.
instr 1
; Input mixed with output of phaser1 to generate notches.
; Shows the effects of different iorder values on the sound
idur = p3
iamp = p4 * .05
iorder = p5          ; number of 1st-order stages in phaser1 network.
                        ; Divide iorder by 2 to get the number of notches.
ifreq = p6           ; frequency of modulation of phaser1
ifeed = p7           ; amount of feedback for phaser1

kamp    linseg 0, .2, iamp, idur - .2, iamp, .2, 0

iharms = (sr*.4) / 100

asig    gbuzz 1, 100, iharms, 1, .95, 2 ; "Sawtooth" waveform modulation oscillator for phaser1 u
kfreq    oscili 5500, ifreq, 1
kmod     = kfreq + 5600

aphs    phaser1 asig, kmod, iorder, ifeed

out      (asig + apha) * iamp
endin

</CsInstruments>
<CsScore>

; inverted half-sine, used for modulating phaser1 frequency
f1 0 16384 9 .5 -1 0
; cosine wave for gbuzz
f2 0 8192 9 1 1 .25

; phaser1
i1 0 5 7000 4 .2 .9
i1 6 5 7000 6 .2 .9
i1 12 5 7000 8 .2 .9
i1 18 5 7000 16 .2 .9
i1 24 5 7000 32 .2 .9
i1 30 5 7000 64 .2 .9
e

</CsScore>
</CsoundSynthesizer>
```

Technical History

A general description of the differences between flanging and phasing can be found in Hartmann [1]. An early implementation of first-order allpass filters connected in series can be found in Beigel [2], where the bilinear z-transform is used for determining the phase shift frequency of each stage. Cronin [3] presents a similar implementation for a four-stage phase shifting network. Chamberlin [4] and Smith [5] both discuss using second-order allpass sections for greater control over notch depth, width, and frequency.

References

1. Hartmann, W.M. "Flanging and Phasers." Journal of the Audio Engineering Society, Vol. 26, No. 6, pp. 439-443, June 1978.
2. Beigel, Michael I. "A Digital 'Phase Shifter' for Musical Applications, Using the Bell Labs (Alles-Fischer) Digital Filter Module." Journal of the Audio Engineering Society, Vol. 27, No. 9, pp. 673-676, September 1979.
3. Cronin, Dennis. "Examining Audio DSP Algorithms." Dr. Dobb's Journal, July 1994, p. 78-83.
4. Chamberlin, Hal. Musical Applications of Microprocessors. Second edition. Indianapolis, Indiana: Hayden Books, 1985.
5. Smith, Julius O. "An Allpass Approach to Digital Phasing and Flanging." Proceedings of the 1984 ICMC, p. 103-108.

See Also

phaser2

Credits

Author: Sean Costello
Seattle, Washington
1999

November 2002. Added a note about the *kord* parameter, thanks to Rasmus Ekman.

New in Csound version 4.0

phaser2

phaser2 — Second-order allpass filters arranged in a series.

Description

An implementation of *iord* number of second-order allpass filters in series.

Syntax

ares **phaser2** asig, kfreq, kq, kord, kmode, ksep, kfeedback

Initialization

iskip (optional, default=0) -- used to control initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

kfreq -- frequency (in Hz) of the filter(s). This is the center frequency of the notch of the first allpass filter in the series. This frequency is used as the base frequency from which the frequencies of the other notches are derived.

kq -- Q of each notch. Higher Q values result in narrow notches. A Q between 0.5 and 1 results in the strongest "phasing" effect, but higher Q values can be used for special effects.

kord -- the number of allpass stages in series. These are second-order filters, and *iord* can range from 1 to 2499. With higher orders, the computation time increases.

kfeedback -- amount of the output which is fed back into the input of the allpass chain. With larger amounts of feedback, more prominent notches appear in the spectrum of the output. *kfeedback* must be between -1 and +1. for stability.

kmode -- used in calculation of notch frequencies.



Note

Although *kord* and *kmode* are listed as k-rate, they are in fact accessed only at init-time. So if you are using k-rate arguments, they must be assigned with *init*.

ksep -- scaling factor used, in conjunction with *imode*, to determine the frequencies of the additional notches in the output spectrum.

phaser2 implements *iord* number of second-order allpass sections, connected in series. The use of second-order allpass sections allows for the precise placement of the frequency, width, and depth of notches in the frequency spectrum. *iord* is used to directly determine the number of notches in the spectrum; e.g. for *iord* = 6, there will be 6 notches in the output spectrum.

There are two possible modes for determining the notch frequencies. When *imode* = 1, the notch frequencies are determined the following function:

frequency of notch N = kbf + (ksep * kbf * N-1)

For example, with *imode* = 1 and *ksep* = 1, the notches will be in harmonic relationship with the notch frequency determined by *kfreq* (i.e. if there are 8 notches, with the first at 100 Hz, the next notches will be at 200, 300, 400, 500, 600, 700, and 800 Hz). This is useful for generating a "comb filtering" effect, with the number of notches determined by *iord*. Different values of *ksep* allow for inharmonic notch frequencies and other special effects. *ksep* can be swept to create an expansion or contraction of the notch frequencies. A useful visual analogy for the effect of sweeping *ksep* would be the bellows of an accordion as it is being played - the notches will be separated, then compressed together, as *ksep* changes.

When *imode* = 2, the subsequent notches are powers of the input parameter *ksep* times the initial notch frequency specified by *kfreq*. This can be used to set the notch frequencies to octaves and other musical intervals. For example, the following lines will generate 8 notches in the output spectrum, with the notches spaced at octaves of *kfreq*:

```
aphs phaser2 ain, kfreq, 0.5, 8, 2, 2, 0
aout =      ain + aphis
```

When *imode* = 2, the value of *ksep* must be greater than 0. *ksep* can be swept to create a compression and expansion of notch frequencies (with more dramatic effects than when *imode* = 1).

Examples

Here is an example of the phaser2 opcode. It uses the file *phaser2.csd* [examples/phaser2.csd].

Example 359. Example of the phaser2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o phaser2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 2                                ; demonstration of phase shifting abilities of phaser2.
; Input mixed with output of phaser2 to generate notches.
; Demonstrates the interaction of imode and ksep.
idur = p3
iamp = p4 * .04
iorder = p5                            ; number of 2nd-order stages in phaser2 network
ifreq = p6                             ; not used
ifeed = p7                             ; amount of feedback for phaser2
imode = p8                             ; mode for frequency scaling
isep = p9                              ; used with imode to determine notch frequencies
kamp = linseg 0, .2, iamp, idur - .2, iamp, .2, 0
iharms = (sr*.4) / 100

; "Sawtooth" waveform exponentially decaying function, to control notch frequencies
asig gbuzz 1, 100, iharms, 1, .95, 2
kline expseg 1, idur, .005
aphs phaser2 asig, kline * 2000, .5, iorder, imode, isep, ifeed

out (asig + aphis) * iamp
endin

</CsInstruments>
<CsScore>
```

```
; cosine wave for gbuzz
f2 0 8192 9 1 1 .25

; phaser2, imode=1
i2 00 10 7000 8 .2 .9 1 .33
i2 11 10 7000 8 .2 .9 1 2

; phaser2, imode=2
i2 22 10 7000 8 .2 .9 2 .33
i2 33 10 7000 8 .2 .9 2 2
e

</CsScore>
</CsoundSynthesizer>
```

Technical History

A general description of the differences between flanging and phasing can be found in Hartmann [1]. An early implementation of first-order allpass filters connected in series can be found in Beigel [2], where the bilinear z-transform is used for determining the phase shift frequency of each stage. Cronin [3] presents a similar implementation for a four-stage phase shifting network. Chamberlin [4] and Smith [5] both discuss using second-order allpass sections for greater control over notch depth, width, and frequency.

References

1. Hartmann, W.M. "Flanging and Phasers." Journal of the Audio Engineering Society, Vol. 26, No. 6, pp. 439-443, June 1978.
2. Beigel, Michael I. "A Digital 'Phase Shifter' for Musical Applications, Using the Bell Labs (Alles-Fischer) Digital Filter Module." Journal of the Audio Engineering Society, Vol. 27, No. 9, pp. 673-676, September 1979.
3. Cronin, Dennis. "Examining Audio DSP Algorithms." Dr. Dobb's Journal, July 1994, p. 78-83.
4. Chamberlin, Hal. Musical Applications of Microprocessors. Second edition. Indianapolis, Indiana: Hayden Books, 1985.
5. Smith, Julius O. "An Allpass Approach to Digital Phasing and Flanging." Proceedings of the 1984 ICMC, p. 103-108.

See Also

phaser1

Credits

Author: Sean Costello
Seattle, Washington
1999

November 2002. Added a note about the *kord* and *kmode* parameters, thanks to Rasmus Ekman.

New in Csound version 4.0

phasor

phasor — Produce a normalized moving phase value.

Description

Produce a normalized moving phase value.

Syntax

```
ares phasor xcps [, iphs]
```

```
kres phasor kcps [, iphs]
```

Initialization

iphs (optional) -- initial phase, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is zero.

Performance

An internal phase is successively accumulated in accordance with the *kcps* or *xcps* frequency to produce a moving phase value, normalized to lie in the range $0 \leq \text{phs} < 1$.

When used as the index to a *table* unit, this phase (multiplied by the desired function table length) will cause it to behave like an oscillator.

Note that *phasor* is a special kind of integrator, accumulating phase increments that represent frequency settings.

Examples

Here is an example of the phasor opcode. It uses the file *phasor.csd* [examples/phasor.csd].

Example 360. Example of the phasor opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o phasor.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create an index that repeats once per second.
kcps init 1
kndx phasor kcps
```

```
; Read Table #1 with our index.
ifn = 1
ixmode = 1
kfreq table kndx, ifn, ixmode

; Generate a sine waveform, use our table values
; to vary its frequency.
a1 oscil 20000, kfreq, 2
out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a line from 200 to 2,000.
f 1 0 1025 -7 200 1024 2000
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See also

The *Table Access* opcodes like: *table*, *tablei*, *table3* and *tab*.

Also: *phasorbnk*.

Credits

Example written by Kevin Conder.

phasorbnk

phasorbnk — Produce an arbitrary number of normalized moving phase values.

Description

Produce an arbitrary number of normalized moving phase values, accessible by an index.

Syntax

```
ares phasorbnk xcps, kndx, icnt [, iphs]
```

```
kres phasorbnk kcps, kndx, icnt [, iphs]
```

Initialization

icnt -- maximum number of phasors to be used.

iphs -- initial phase, expressed as a fraction of a cycle (0 to 1). If -1 initialization is skipped. If *iphs*>1 each phasor will be initialized with a random value.

Performance

kndx -- index value to access individual phasors

For each independent phasor, an internal phase is successively accumulated in accordance with the *kcps* or *xcps* frequency to produce a moving phase value, normalized to lie in the range $0 \leq \text{phs} < 1$. Each individual phasor is accessed by index *kndx*.

This phasor bank can be used inside a k-rate loop to generate multiple independent voices, or together with the *adsynt* opcode to change parameters in the tables used by *adsynt*.

Examples

Here is an example of the phasorbnk opcode. It uses the file *phasorbnk.csd* [examples/phasorbnk.csd].

Example 361. Example of the phasorbnk opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o phasorbnk.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Generate a sinewave table.
```

```
giwave ftgen 1, 0, 1024, 10, 1

; Instrument #1
instr 1
  ; Generate 10 voices.
  icnt = 10
  ; Empty the output buffer.
  asum = 0
  ; Reset the loop index.
  kindex = 0

  ; This loop is executed every k-cycle.
loop:
  ; Generate non-harmonic partials.
  kcps = (kindex+1)*100+30
  ; Get the phase for each voice.
  aphas phasorbnk kcps, kindex, icnt
  ; Read the wave from the table.
  asig table aphas, giwave, 1
  ; Accumulate the audio output.
  asum = asum + asig

  ; Increment the index.
  kindex = kindex + 1

  ; Perform the loop until the index (kindex) reaches
  ; the counter value (icnt).
  if (kindex < icnt) kgoto loop

  out asum*3000
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Generate multiple voices with independent partials. This example is better with *adsynt*. See also the example under *adsynt*, for k-rate use of *phasorbnk*.

Credits

Author: Peter Neubäcker
Munich, Germany
August 1999

New in Csound version 3.58

pindex

pindex — Returns the value of a specified pfield.

Description

pindex returns the value of a specified pfield.

Syntax

ivalue **pindex** ipfieldIndex

Initialization

ipfieldIndex - pfield number to query.

ivalue - value of the pfield.

Examples

Here is an example of the pindex opcode. It uses the file *pindex.csd* [examples/pindex.csd].

Example 362. Example of the pindex opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      ; -d        -M0    ;;RT audio I/O with MIDI in
; For Non-realtime output leave only the line below:
;-o pindex.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
;Example by Anthony Kozar Dec 2006

instr 1
  inum      pcount
  index     init 1
  loop1:
    ivalue  pindex index
    printf_i "p%d = %f\n", 1, index, ivalue
    index   = index + 1
    if (index <= inum) igoto loop1
  print inum
endin

</CsInstruments>
<CsScore>
i1  0 3 40 50      ; has 5 pfields
i1  1 2 80         ; has 5 due to carry
i1  2 1 40 50 60 70 ; has 7
e
</CsScore>
</CsoundSynthesizer>
```

The example will produce the following output:

```
new alloc for instr 1:
WARNING: instr 1 uses 3 p-fields but is given 5
```



```
p1 = 1.000000
p2 = 0.000000
p3 = 3.000000
p4 = 40.000000
p5 = 50.000000
instr 1: inum = 5.000
B 0.000 .. 1.000 T 1.000 TT 1.000 M: 0.0
new alloc for instr 1:
WARNING: instr 1 uses 3 p-fields but is given 5
p1 = 1.000000
p2 = 1.000000
p3 = 2.000000
p4 = 80.000000
p5 = 50.000000
instr 1: inum = 5.000
B 1.000 .. 2.000 T 2.000 TT 2.000 M: 0.0
new alloc for instr 1:
WARNING: instr 1 uses 3 p-fields but is given 7
p1 = 1.000000
p2 = 2.000000
p3 = 1.000000
p4 = 40.000000
p5 = 50.000000
p6 = 60.000000
p7 = 70.000000
instr 1: inum = 7.000
```

The warnings can be ignored, because the pfields are used indirectly through pindex instead of explicitly through p4, p5, etc.

See Also

pcount

Credits

Example by: Anthony Kozar

Dec. 2006

pinkish

pinkish — Generates approximate pink noise.

Description

Generates approximate pink noise (-3dB/oct response) by one of two different methods:

- a multirate noise generator after Moore, coded by Martin Gardner
- a filter bank designed by Paul Kellet

Syntax

```
ares pinkish xin [, imethod] [, inumbands] [, iseed] [, iskip]
```

Initialization

imethod (optional, default=0) -- selects filter method:

- 0 = Gardner method (default).
- 1 = Kellet filter bank.
- 2 = A somewhat faster filter bank by Kellet, with less accurate response.

inumbands (optional) -- only effective with Gardner method. The number of noise bands to generate. Maximum is 32, minimum is 4. Higher levels give smoother spectrum, but above 20 bands there will be almost DC-like slow fluctuations. Default value is 20.

iseed (optional, default=0) -- only effective with Gardner method. If non-zero, seeds the random generator. If zero, the generator will be seeded from current time. Default is 0.

iskip (optional, default=0) -- if non-zero, skip (re)initialization of internal state (useful for tied notes). Default is 0.

Performance

xin -- for Gardner method: k- or a-rate amplitude. For Kellet filters: normally a-rate uniform random noise from *rand* (31-bit) or *unirand*, but can be any a-rate signal. The output peak value varies widely ($\pm 15\%$) even over long runs, and will usually be well below the input amplitude. Peak values may also occasionally overshoot input amplitude or noise.

pinkish attempts to generate pink noise (i.e., noise with equal energy in each octave), by one of two different methods.

The first method, by Moore & Gardner, adds several (up to 32) signals of white noise, generated at octave rates (sr, sr/2, sr/4 etc). It obtains pseudo-random values from an internal 32-bit generator. This random generator is local to each opcode instance and seedable (similar to *rand*).

The second method is a lowpass filter with a response approximating -3dB/oct. If the input is uniform white noise, it outputs pink noise. Any signal may be used as input for this method. The high quality filter is slower, but has less ripple and a slightly wider operating frequency range than less computationally intense versions. With the Kellet filters, seeding is not used.

The Gardner method output has some frequency response anomalies in the low-mid and high-mid frequency ranges. More low-frequency energy can be generated by increasing the number of bands. It is also a bit faster. The refined Kellet filter has very smooth spectrum, but a more limited effective range. The level increases slightly at the high end of the spectrum.

Examples

Here is an example of the pinkish opcode. It uses the file *pinkish.csd* [examples/pinkish.csd].

Example 363. Example of the pinkish opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o pinkish.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  awhite unirand 2.0

  ; Normalize to +/-1.0
  awhite = awhite - 1.0

  apink pinkish awhite, 1, 0, 0, 1

  out apink * 30000
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Kellet-filtered noise for a tied note (*iskip* is non-zero).

Credits

Authors: Phil Burk and John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

Adapted for Csound by Rasmus Ekman

The noise bands method is due to F. R. Moore (or R. F. Voss), and was presented by Martin Gard-

ner in an oft-cited article in Scientific American. The present version was coded by Phil Burk as the result of discussion on the music-dsp mailing list, with significant optimizations suggested by James McCartney.

The filter bank was designed by Paul Kellet, posted to the music-dsp mailing list.

The whole pink noise discussion was collected on a HTML page by Robin Whittle, which is currently available at <http://www.firstpr.com.au/dsp/pink-noise/>.

Added notes by Rasmus Ekman on September 2002.

pitch

pitch — Tracks the pitch of a signal.

Description

Using the same techniques as *spectrum* and *specptrk*, pitch tracks the pitch of the signal in octave point decimal form, and amplitude in dB.

Syntax

```
koct, kamp pitch asig, iupdt, ilo, ihi, idbthresh [, ifrqs] [, iconf] \  
[, istr] [, iocts] [, iq] [, inptls] [, irolloff] [, iskip]
```

Initialization

iupdt -- length of period, in seconds, that outputs are updated

ilo, *ihi* -- range in which pitch is detected, expressed in octave point decimal

idbthresh -- amplitude, expressed in decibels, necessary for the pitch to be detected. Once started it continues until it is 6 dB down.

ifrqs (optional) -- number of divisions of an octave. Default is 12 and is limited to 120.

iconf (optional) -- the number of conformations needed for an octave jump. Default is 10.

istr (optional) -- starting pitch for tracker. Default value is $(ilo + ihi)/2$.

iocts (optional) -- number of octave decimations in spectrum. Default is 6.

iq (optional) -- Q of analysis filters. Default is 10.

inptls (optional) -- number of harmonics, used in matching. Computation time increases with the number of harmonics. Default is 4.

irolloff (optional) -- amplitude rolloff for the set of filters expressed as fraction per octave. Values must be positive. Default is 0.6.

iskip (optional) -- if non-zero, skips initialization. Default is 0.

Performance

koct -- The pitch output, given in the octave point decimal format.

kamp -- The amplitude output.

pitch analyzes the input signal, *asig*, to give a pitch/amplitude pair of outputs, for the strongest frequency in the signal. The value is updated every *iupdt* seconds.

The number of partials and rolloff fraction can effect the pitch tracking, so some experimentation may be necessary. Suggested values are 4 or 5 harmonics, with rolloff 0.6, up to 10 or 12 harmonics with rolloff 0.75 for complex timbres, with a weak fundamental.

Examples

Here is an example of the pitch opcode. It uses the file *pitch.csd* [examples/pitch.csd] and *mary.wav* [examples/mary.wav].

Example 364. Example of the pitch opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o pitch.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1 - play an audio file without effects.
instr 1
  asig soundin "mary.wav"
  out asig
endin

; Instrument #2 - track the pitch of an audio file.
instr 2
  iupdt = 0.01
  ilo = 7
  ihi = 9
  idbthresh = 10
  ifrqs = 12
  iconf = 10
  istr = 8

  asig soundin "mary.wav"

  ; Follow the audio file, get its pitch and amplitude.
  koct, kamp pitch asig, iupdt, ilo, ihi, idbthresh, ifrqs, iconf, istr

  ; Re-synthesize the audio file with a different sounding waveform.
  kamp2 = kamp * 10
  kcps = cpsoct(koct)
  al oscil kamp2, kcps, 1

  out al
endin

</CsInstruments>
<CsScore>

; Table #1: A different sounding waveform.
f 1 0 32768 11 7 3 .7

; Play Instrument #1, the audio file, for three seconds.
i 1 0 3
; Play Instrument #2, the "re-synthesized" waveform, for three seconds.
i 2 3 3
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch
University of Bath, Codemist Ltd.
Bath, UK
April 1999

Example written by Kevin Conder.

New in Csound version 3.54

pitchamdf

pitchamdf — Follows the pitch of a signal based on the AMDF method.

Description

Follows the pitch of a signal based on the AMDF method (Average Magnitude Difference Function). Outputs pitch and amplitude tracking signals. The method is quite fast and should run in real-time. This technique usually works best for monophonic signals.

Syntax

```
kcps, krms pitchamdf asig, imincps, imaxcps [, icps] [, imedi] \  
[, idowns] [, iexcps] [, irmsmedi]
```

Initialization

imincps -- estimated minimum frequency (expressed in Hz) present in the signal

imaxcps -- estimated maximum frequency present in the signal

icps (optional, default=0) -- estimated initial frequency of the signal. If 0, $icps = (imincps + imaxcps) / 2$. The default is 0.

imedi (optional, default=1) -- size of median filter applied to the output *kcps*. The size of the filter will be $imedi * 2 + 1$. If 0, no median filtering will be applied. The default is 1.

idowns (optional, default=1) -- downsampling factor for *asig*. Must be an integer. A factor of *idowns* > 1 results in faster performance, but may result in worse pitch detection. Useful range is 1 - 4. The default is 1.

iexcps (optional, default=0) -- how frequently pitch analysis is executed, expressed in Hz. If 0, *iexcps* is set to *imincps*. This is usually reasonable, but experimentation with other values may lead to better results. Default is 0.

irmsmedi (optional, default=0) -- size of median filter applied to the output *krms*. The size of the filter will be $irmsmedi * 2 + 1$. If 0, no median filtering will be applied. The default is 0.

Performance

kcps -- pitch tracking output

krms -- amplitude tracking output

pitchamdf usually works best for monophonic signals, and is quite reliable if appropriate initial values are chosen. Setting *imincps* and *imaxcps* as narrow as possible to the range of the signal's pitch, results in better detection and performance.

Because this process can only detect pitch after an initial delay, setting *icps* close to the signal's real initial pitch prevents spurious data at the beginning.

The median filter prevents *kcps* from jumping. Experiment to determine the optimum value for *imedi* for a given signal.

Other initial values can usually be left at the default settings. Lowpass filtering of *asig* before passing it to *pitchamdf*, can improve performance, especially with complex waveforms.

Examples

Here is an example of the `pitchamdf` opcode. It uses the file `pitchamdf.csd` [examples/pitchamdf.csd] and `mary.wav` [examples/mary.wav].

Example 365. Example of the `pitchamdf` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc       -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o pitchamdf.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; synth waveform
giwave ftgen 2, 0, 1024, 10, 1, 1, 1, 1

; Instrument #1 - play an audio file with no effects.
instr 1
; get input signal with original freq.
asig soundin "mary.wav"

out asig
endin

; Instrument #2 - play the synth waveform using the
; same pitch and amplitude as the audio file.
instr 2
; get input signal with original freq.
asig soundin "mary.wav"

; lowpass-filter
asig tone asig, 1000
; extract pitch and envelope
kcps, krms pitchamdf asig, 150, 500, 200
; "re-synthesize" with the synth waveform, giwave.
asigl oscil krms, kcps, giwave

out asigl
endin

</CsInstruments>
<CsScore>

; Play Instrument #1, the audio file, for three seconds.
i 1 0 3
; Play Instrument #2, the "re-synthesized" waveform, for three seconds.
i 2 3 3
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Peter Neubäcker
Munich, Germany
August 1999

New in Csound version 3.59

planet

planet — Simulates a planet orbiting in a binary star system.

Description

planet simulates a planet orbiting in a binary star system. The outputs are the x, y and z coordinates of the orbiting planet. It is possible for the planet to achieve escape velocity by a close encounter with a star. This makes this system somewhat unstable.

Syntax

```
ax, ay, az planet kmass1, kmass2, ksep, ix, iy, iz, ivx, ivy, ivz, idelta \
[, ifriction] [, iskip]
```

Initialization

ix, iy, iz -- the initial x, y and z coordinates of the planet

ivx, ivy, ivz -- the initial velocity vector components for the planet.

idelta -- the step size used to approximate the differential equation.

ifriction (optional, default=0) -- a value for friction, which can be used to keep the system from blowing up

iskip (optional, default=0) -- if non zero skip the initialisation of the filter. (New in Csound version 4.23f13 and 5.0)

Performance

ax, ay, az -- the output x, y, and z coordinates of the planet

kmass1 -- the mass of the first star

kmass2 -- the mass of the second star

Examples

Here is an example of the planet opcode. It uses the file *planet.csd* [examples/planet.csd].

Example 366. Example of the planet opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o planet.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
```

```
kr = 44100
ksmps = 1
nchnls = 2

; Instrument #1 - a planet orbiting in 3D space.
instr 1
; Create a basic tone.
kamp init 5000
kcps init 440
ifn = 1
asnd oscil kamp, kcps, ifn

; Figure out its X, Y, Z coordinates.
kml init 0.5
km2 init 0.35
ksep init 2.2
ix = 0
iy = 0.1
iz = 0
ivx = 0.5
ivy = 0
ivz = 0
ih = 0.0003
ifric = -0.1
ax1, ay1, az1 planet kml, km2, ksep, ix, iy, iz, \
                    ivx, ivy, ivz, ih, ifric

; Place the basic tone within 3D space.
kx downsamp ax1
ky downsamp ay1
kz downsamp az1
idist = 1
ift = 0
imode = 1
imdel = 1.018853416
iovr = 2
aw2, ax2, ay2, az2 spat3d asnd, kx, ky, kz, idist, \
                        ift, imode, imdel, iovr

; Convert the 3D sound to stereo.
aleft = aw2 + ay2
aright = aw2 - ay2

outs aleft, aright
endin

</CsInstruments>
<CsScore>

; Table #1 a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 10 seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Hans Mikelson
December 1998

New in Csound version 3.50

pluck

pluck — Produces a naturally decaying plucked string or drum sound.

Description

Audio output is a naturally decaying plucked string or drum sound based on the Karplus-Strong algorithms.

Syntax

```
ares pluck kamp, kcps, icps, ifn, imeth [, iparm1] [, iparm2]
```

Initialization

icps -- intended pitch value in Hz, used to set up a buffer of 1 cycle of audio samples which will be smoothed over time by a chosen decay method. *icps* normally anticipates the value of *kcps*, but may be set artificially high or low to influence the size of the sample buffer.

ifn -- table number of a stored function used to initialize the cyclic decay buffer. If *ifn* = 0, a random sequence will be used instead.

imeth -- method of natural decay. There are six, some of which use parameters values that follow.

1. simple averaging. A simple smoothing process, uninfluenced by parameter values.
2. stretched averaging. As above, with smoothing time stretched by a factor of *iparm1* (=1).
3. simple drum. The range from pitch to noise is controlled by a 'roughness factor' in *iparm1* (0 to 1). Zero gives the plucked string effect, while 1 reverses the polarity of every sample (octave down, odd harmonics). The setting .5 gives an optimum snare drum.
4. stretched drum. Combines both roughness and stretch factors. *iparm1* is roughness (0 to 1), and *iparm2* the stretch factor (=1).
5. weighted averaging. As method 1, with *iparm1* weighting the current sample (the status quo) and *iparm2* weighting the previous adjacent one. *iparm1* + *iparm2* must be <= 1.
6. 1st order recursive filter, with coefs .5. Unaffected by parameter values.

iparm1, *iparm2* (optional) -- parameter values for use by the smoothing algorithms (above). The default values are both 0.

Performance

kamp -- the output amplitude.

kcps -- the resampling frequency in cycles-per-second.

An internal audio buffer, filled at i-time according to *ifn*, is continually resampled with periodicity *kcps* and the resulting output is multiplied by *kamp*. Parallel with the sampling, the buffer is smoothed to simulate the effect of natural decay.

Plucked strings (1,2,5,6) are best realized by starting with a random noise source, which is rich in initial harmonics. Drum sounds (methods 3,4) work best with a flat source (wide pulse), which produces a deep noise attack and sharp decay.

The original Karplus-Strong algorithm used a fixed number of samples per cycle, which caused serious quantization of the pitches available and their intonation. This implementation resamples a buffer at the exact pitch given by *kcps*, which can be varied for vibrato and glissando effects. For low values of the orch sampling rate (e.g. $sr = 10000$), high frequencies will store only very few samples ($sr / icps$). Since this may cause noticeable noise in the resampling process, the internal buffer has a minimum size of 64 samples. This can be further enlarged by setting *icps* to some artificially lower pitch.

Examples

Here is an example of the pluck opcode. It uses the file *pluck.csd* [examples/pluck.csd].

Example 367. Example of the pluck opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o pluck.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 20000
  kcps = 440
  icps = 440
  ifn = 0
  imeth = 1

  al pluck kamp, kcps, icps, ifn, imeth
  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Example written by Kevin Conder.

poisson

poisson — Poisson distribution random number generator (positive values only).

Description

Poisson distribution random number generator (positive values only). This is an x-class noise generator.

Syntax

```
ares poisson klambda
```

```
ires poisson klambda
```

```
kres poisson klambda
```

Performance

ares, *kres*, *ires* - number of events occurring (always an integer).

klambda - the expected number of occurrences that occur during the rate interval.

Adapted from Wikipedia:

In probability theory and statistics, the Poisson distribution is a discrete probability distribution. It expresses the probability of a number of events occurring in a fixed period of time if these events occur with a known average rate, and are independent of the time since the last event.

The Poisson distribution describing the probability that there are exactly k occurrences (k being a non-negative integer, $k = 0, 1, 2, \dots$) is:

$$f(k; \lambda) = \frac{e^{-\lambda} \lambda^k}{k!},$$

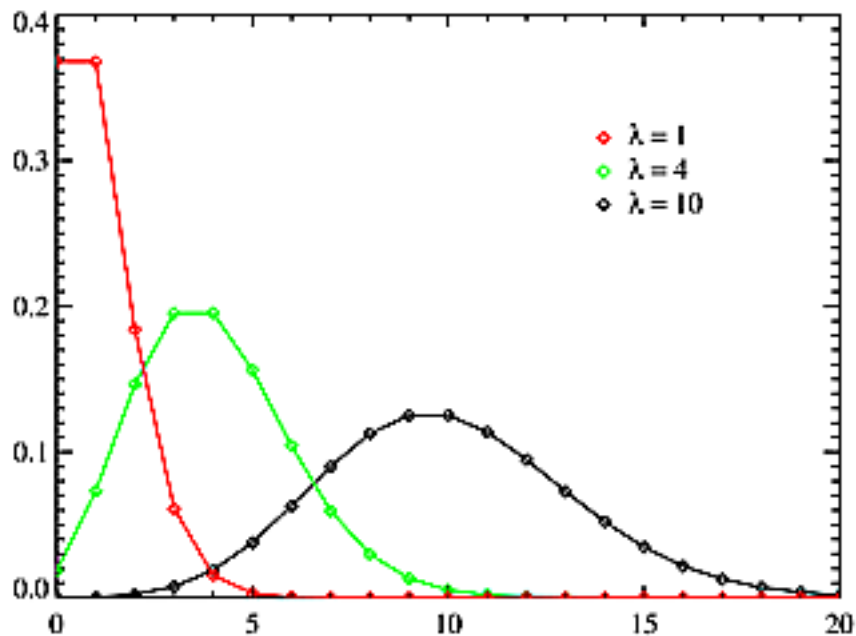
where:

- λ is a positive real number, equal to the expected number of occurrences that occur during the given interval. For instance, if the events occur on average every 4 minutes, and you are interested in the number of events occurring in a 10 minute interval, you would use as model a Poisson distribution with $\lambda = 10/4 = 2.5$. This parameter is called *klambda* on the *poisson* opcodes.
- k refers to the number of i- , k- or a- periods elapsed.

The Poisson distribution arises in connection with Poisson processes. It applies to various phenomena of discrete nature (that is, those that may happen 0, 1, 2, 3, ... times during a given period of time or in a given area) whenever the probability of the phenomenon happening is constant in time or space. Examples of events that can be modelled as Poisson distributions include:

- The number of cars that pass through a certain point on a road (sufficiently distant from traffic lights) during a given period of time.
- The number of spelling mistakes one makes while typing a single page.

- The number of phone calls at a call center per minute.
- The number of times a web server is accessed per minute.
- The number of roadkill (animals killed) found per unit length of road.
- The number of mutations in a given stretch of DNA after a certain amount of radiation.
- The number of unstable nuclei that decayed within a given period of time in a piece of radioactive substance. The radioactivity of the substance will weaken with time, so the total time interval used in the model should be significantly less than the mean lifetime of the substance.
- The number of pine trees per unit area of mixed forest.
- The number of stars in a given volume of space.
- The distribution of visual receptor cells in the retina of the human eye.
- The number of viruses that can infect a cell in cell culture.



A diagram showing the Poisson distribution.

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the poisson opcode. It uses the file *poisson.csd* [examples/poisson.csd]. It is written for *NIX systems, and will generate errors on Windows.

Example 368. Example of the poisson opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o poisson.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
ksmps = 441 ;ksmps set deliberately high to have few k-periods per second
nchnls = 1

; Instrument #1.
instr 1
; Generates a random number in a poisson distribution.
; klambda = 1

il poisson 1

print il
endin

instr 2

kres poisson p4
printk (ksmps/sr),kres ;prints every k-period
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
i 2 1 0.2 0.5
i 2 2 0.2 4 ;average 4 events per k-period
i 2 3 0.2 20 ;average 20 events per k-period
e

</CsScore>
</CsoundSynthesizer>
```

See Also

seed, betarand, bexpnrd, cauchy, exprand, gauss, linrand, pcauchy, trirand, unirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder and Andrés Cabrera

polyaft

polyaft — Returns the polyphonic after-touch pressure of the selected note number.

Description

polyaft returns the polyphonic pressure of the selected note number, optionally mapped to an user-specified range.

Syntax

```
ires polyaft inote [, ilow] [, ihigh]
```

```
kres polyaft inote [, ilow] [, ihigh]
```

Initialization

inote -- note number. Normally set to the value returned by *notnum*

ilow (optional, default: 0) -- lowest output value

ihigh (optional, default: 127) -- highest output value

Performance

kres -- polyphonic pressure (aftertouch).

Examples

Here is an example of the *polyaft* opcode. It uses the file *polyaft.csd* [examples/polyaft.csd].

Don't forget that you must include the *-F flag* when using an external MIDI file like “polyaft.mid”.

Example 369. Example of the *polyaft* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d          -M0    ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o polyaft.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 10
nchnls = 1

        massign 1, 1
itmp ftgen 1, 0, 1024, 10, 1           ; sine wave

        instr 1

kcps cpsmidib 2           ; note frequency
inote notnum              ; note number
```

```
kaft polyaft inote, 0, 127 ; aftertouch  
; interpolate aftertouch to eliminate clicks  
ktmp phasor 40  
ktmp trigger 1 - ktmp, 0.5, 0  
kaft tlineto kaft, 0.025, ktmp  
; map to sine curve for crossfade  
kaft = sin(kaft * 3.14159 / 254) * 22000  
  
asnd oscili kaft, kcps, 1  
  
    out asnd  
  
    endin  
  
</CsInstruments>  
<CsScore>  
  
t 0 120  
f 0 9 2 -2 0  
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

Credits

Added thanks to an email from Istvan Varga

New in version 4.12

polynomial

polynomial — Efficiently evaluates a polynomial of arbitrary order.

Description

The *polynomial* opcode calculates a polynomial with a single a-rate input variable. The polynomial is a sum of any number of terms in the form $k_n * x^n$ where k_n is the n th coefficient of the expression. These coefficients are k-rate values.

Syntax

```
aout polynomial ain, k0 [, k1 [, k2 [...]]]
```

Performance

ain -- the input signal used as the independent variable of the polynomial ("x").

aout -- the output signal ("y").

k0, *k1*, *k2*, ... -- the coefficients for each term of the polynomial.

If we consider the input parameter *ain* to be "x" and the output *aout* to be "y", then the *polynomial* opcode calculates the following equation:

$$y = k_0 + k_1 * x + k_2 * x^2 + k_3 * x^3 + \dots$$

See Also

chebyshevpoly, *mac maca*

Examples

Here is an example of the polynomial opcode. It uses the file *polynomial.csd* [examples/polynomi-al.csd].

Example 370. Example of the polynomial opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
  ; no sound output
  -n
</CsOptions>
<CsInstruments>

sr = 10 ; audio rate is not important
kr = 10 ; execute the statements in instr 1 ten times per second

instr 1
  ; ax will vary from 1 to 10
  ax          init      1

  ; ay = ax^3 + 2ax^2 + 3ax + 4
```

```
ay      polynomial  ax, 4, 3, 2, 1

; convert our a-rate signals to k-rate values so that we can print
ky      downsamp    ay
kx      downsamp    ax
printf  "%d:  %d\n", kx, kx, ky

ax      =            ax + 1

endin

</CsInstruments>
<CsScore>

i1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Anthony Kozar
January 2008

New in Csound version 5.08

pop

push — Pops values from the global stack.

Description

Pops values from the global stack.

Syntax

```
xval1, [xval2, ... , xval31] pop
```

```
ival1, [ival2, ... , ival31] pop
```

Initialization

ival1 ... ival31 - values to be popped from the stack.

Performance

xval1 ... xval31 - values to be popped from the stack.

The given values are popped from the stack. The global stack works in LIFO order: after multiple *push* calls, *pop* should be used in reverse order.

Each *push* or *pop* operation can work on a "bundle" of multiple variables. When using *pop*, the number, type, and order of items must match those used by the corresponding *push*. That is, after a 'push Sfoo, ibar', you must call something like 'pop Sbar, ifoo', and not e.g. two separate 'pop' statements.

push and *pop* opcodes can take variables of any type (i-, k-, a- and strings). Use of any combination of i, k, a, and S types is allowed. Variables of type 'a' and 'k' are passed at performance time only, while 'i' and 'S' are passed at init time only.

push/pop for a, k, i, and S types copy data by value. By contrast, *push_f* only pushes a "reference" to the f-signal, and then the corresponding *pop_f* will copy directly from the original variable to its output signal. For this reason, changing the source f-signal of *push_f* before *pop_f* is called is not recommended, and if the instrument instance owning the variable that was passed by *push_f* is deactivated before *pop_f* is called, undefined behavior may occur.

Any stack errors (trying to push when there is no more space, or pop from an empty stack, inconsistent number or type of arguments, etc.) are fatal and terminate performance.

See also

stack, *push*, *pop_f* and *push_f*.

Credits

By: Istvan Varga.

2006

pop_f

pop_f — Pops an f-sig frame from the global stack.

Description

Pops an f-sig frame from the global stack.

Syntax

*f*sig **pop_f**

Performance

*f*sig - f-signal to be popped from the stack.

The values are popped the stack. The global stack must be initialized before used, and its size must be set. The global stack works in LIFO order: after multiple *push_f* calls, *pop_f* should be used in reverse order.

push/pop for a, k, i, and S types copy data by value. By contrast, *push_f* only pushes a "reference" to the f-signal, and then the corresponding *pop_f* will copy directly from the original variable to its output signal. For this reason, changing the source f-signal of *push_f* before *pop_f* is called is not recommended, and if the instrument instance owning the variable that was passed by *push_f* is deactivated before *pop_f* is called, undefined behavior may occur.

push_f and *pop_f* can only take a single argument, and the data is passed both at init and performance time.

Any stack errors (trying to push when there is no more space, or pop from an empty stack, inconsistent number or type of arguments, etc.) are fatal and terminate performance.

See also

stack, *push*, *pop* and *push_f*.

Credits

By: Istvan Varga.

2006

port

port — Applies portamento to a step-valued control signal.

Description

Applies portamento to a step-valued control signal.

Syntax

```
kres port ksig, ihtim [, isig]
```

Initialization

*ih**tim* -- half-time of the function, in seconds.

isig (optional, default=0) -- initial (i.e. previous) value for internal feedback. The default value is 0. Negative value will cause initialization to be skipped and last value from previous instance to be used as initial value for note.

Performance

kres -- the output signal at control-rate.

ksig -- the input signal at control-rate.

port applies portamento to a step-valued control signal. At each new step value, *ksig* is low-pass filtered to move towards that value at a rate determined by *ih**tim*. *ih**tim* is the “half-time” of the function (in seconds), during which the curve will traverse half the distance towards the new value, then half as much again, etc., theoretically never reaching its asymptote. With *portk*, the half-time can be varied at the control rate.

See Also

areson, *aresonk*, *atone*, *atonek*, *portk*, *reson*, *resonk*, *tone*, *tonek*

portk

portk — Applies portamento to a step-valued control signal.

Description

Applies portamento to a step-valued control signal.

Syntax

```
kres portk ksig, khtim [, isig]
```

Initialization

isig (optional, default=0) -- initial (i.e. previous) value for internal feedback. The default value is 0.

Performance

kres -- the output signal at control-rate.

ksig -- the input signal at control-rate.

khtim -- half-time of the function in seconds.

portk is like *port* except the half-time can be varied at the control rate.

Examples

Here is an example of the portk opcode. It uses the file *portk.csd* [examples/portk.csd].

Example 371. Example of the portk opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      ; -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o portk.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 128
nchnls = 1

;Example by Andres Cabrera 2007

FLpanel "Slider", 650, 140, 50, 50
  gkval1, gislider1 FLslider "Watch me", 0, 127, 0, 5, -1, 580, 30, 25, 20
  gkval2, gislider2 FLslider "Move me", 0, 127, 0, 5, -1, 580, 30, 25, 80
  gkhtim, gislider3 FLslider "khtim", 0.1, 1, 0, 6, -1, 30, 100, 610, 10
FLpanelEnd
FLrun

FLsetVal_i 0.1, gislider3 ;set initial time to 0.1

instr 1
  kval portk gkval2, gkhtim ; take the value of slider 2 and apply portamento
```

```
FLsetVal 1, kval, gislider1 ;set the value of slider 1 to kval
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one minute.
i 1 0 60
e

</CsScore>
</CsoundSynthesizer>
```

See Also

areson, aresonk, atone, atonek, port, reson, resonk, tone, tonek

Credits

Author: Robin Whittle
Australia
May 1997

poscil

poscil — High precision oscillator.

Description

High precision oscillator.

Syntax

```
ares poscil aamp, acps, ifn [, iphs]
```

```
ares poscil aamp, kcps, ifn [, iphs]
```

```
ares poscil kamp, acps, ifn [, iphs]
```

```
ares poscil kamp, kcps, ifn [, iphs]
```

```
ires poscil kamp, kcps, ifn [, iphs]
```

```
kres poscil kamp, kcps, ifn [, iphs]
```

Initialization

ifn -- function table number

iphs (optional, default=0) -- initial phase (normalized table index 0-1)

Performance

ares -- output signal

kamp, *aamp* -- the amplitude of the output signal.

kcps, *acps* -- the frequency of the output signal in cycles per second.

poscil (precise oscillator) is the same as *oscili*, but allows much more precise frequency control, especially when using long tables and low frequency values. It uses floating-point table indexing, instead of integer math, like *oscil* and *oscili*. It is only a bit slower than *oscili*.

Since Csound 4.22, *poscil* can accept also negative frequency values and use a-rate values both for amplitude and frequency. So both AM and FM are allowed using this opcode.

The opcode *poscil3* is the same as *poscil*, but uses cubic interpolation.

Examples

Here is an example of the *poscil* opcode. It uses the file *poscil.csd* [examples/poscil.csd].

Example 372. Example of the *poscil* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o poscil.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a basic oscillator.
instr 1
  kamp = 10000
  kcps = 440
  ifn = 1

  al poscil kamp, kcps, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

poscil3

Credits

Author: Gabriel Maldonado
Italy
1998

Example written by Kevin Conder.

November 2002. Added a note about the changes to Csound version 4.22, thanks to Rasmus Ekman.

New in Csound version 3.52

poscil3

poscil3 — High precision oscillator with cubic interpolation.

Description

High precision oscillator with cubic interpolation.

Syntax

```
ares poscil3 kamp, kcps, ifn [, iphs]
```

```
kres poscil3 kamp, kcps, ifn [, iphs]
```

Initialization

ifn -- function table number

iphs (optional, default=0) -- initial phase (normalized table index 0-1)

Performance

ares -- output signal

kamp -- the amplitude of the output signal.

kcps -- the frequency of the output signal in cycles per second.

poscil3 works like *poscil*, but uses cubic interpolation.

Examples

Here is an example of the poscil3 opcode. It uses the file *poscil3.csd* [examples/poscil3.csd].

Example 373. Example of the poscil3 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o poscil3.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a basic oscillator.
instr 1
  kamp = 10000
  kcps = 440
  ifn = 1
```

```
    al poscil3 kamp, kcps, ifn
    out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the poscil3 opcode, which uses a table filled from a sound file. It uses the file *poscil3-file.csd* [examples/poscil3-file.csd].

Example 374. Another example of the poscil3 opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o poscil3.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a basic oscillator.
instr 1
    kamp = 10000
    kcps = 440
    ifn = 1

    al poscil3 kamp, kcps, ifn
    out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

poscil

Credits

Authors: John ffitich, Gabriel Maldonado
Italy

Example written by Kevin Conder.

New in Csound version 3.52

pow

pow — Computes one argument to the power of another argument.

Description

Computes *xarg* to the power of *kpow* (or *ipow*) and scales the result by *inorm*.

Syntax

```
ares pow aarg, kpow [, inorm]
```

```
ires pow iarg, ipow [, inorm]
```

```
kres pow karg, kpow [, inorm]
```

Initialization

inorm (optional, default=1) -- The number to divide the result (default to 1). This is especially useful if you are doing powers of a- or k- signals where samples out of range are extremely common!

Performance

aarg, *iarg*, *karg* -- the base.

ipow, *kpow* -- the exponent.



Note

Use ^ with caution in arithmetical statements, as the precedence may not be correct.
New in Csound version 3.493.

Examples

Here is an example of the pow opcode. It uses the file *pow.csd* [examples/pow.csd].

Example 375. Example of the pow opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o pow.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```



```
; Instrument #1.
instr 1
; This could also be expressed as: i1 = 2 ^ 12
i1 pow 2, 12

    print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 4096.000
```

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder.

powershape

powershape — Waveshapes a signal by raising it to a variable exponent.

Description

The *powershape* opcode raises an input signal to a power with pre- and post-scaling of the signal so that the output will be in a predictable range. It also processes negative inputs in a symmetrical way to positive inputs, calculating a dynamic transfer function that is useful for waveshaping.

Syntax

```
aout powershape ain, kShapeAmount [, ifullscale]
```

Initialization

ifullscale -- optional parameter specifying the range of input values from *-ifullscale* to *ifullscale*. Defaults to 1.0 -- you should set this parameter to the maximum expected input value.

Performance

ain -- the input signal to be shaped.

aout -- the output signal.

kShapeAmount -- the amount of the shaping effect applied to the input; equal to the power that the input signal is raised.

The *powershape* opcode is very similar to the *pow* unit generators for calculating the mathematical "power of" operation. However, it introduces a couple of twists that can make it much more useful for waveshaping audio-rate signals. The *kShapeAmount* parameter is the exponent to which the input signal is raised.

To avoid discontinuities, the *powershape* opcode treats all input values as positive (by taking their absolute value) but preserves their original sign in the output signal. This allows for smooth shaping of any input signal while varying the exponent over any range. (*powershape* also (hopefully) deals intelligently with discontinuities that could arise when the exponent and input are both zero. Note though that negative exponents will usually cause the signal to exceed the maximum amplitude specified by the *ifullscale* parameter and should normally be avoided).

The other adaptation involves the *ifullscale* parameter. The input signal is divided by *ifullscale* before being raised to *kShapeAmount* and then multiplied by *ifullscale* before being output. This normalizes the input signal to the interval [-1,1], guaranteeing that the output (before final scaling) will also be within this range for positive shaping amounts and providing a smoothly evolving transfer function while varying the amount of shaping. Values of *kShapeAmount* between (0,1) will make the signal more "convex" while values greater than 1 will make it more "concave". A value of exactly 1.0 will produce no change in the input signal.

See Also

pow, *powoftwo*

Examples

Here is an example of the *powershape* opcode. It uses the file *powershape.csd* [examples/power-shape.csd].

Example 376. Example of the powershape opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o abs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
instr 1
    imaxamp      =      10000
    kshapeamt    line      p5, p3, p6
    aosc         oscili     1.0, cpspch(p4), 1
    aout         powershape aosc, kshapeamt
    adeclick     linseg     0.0, 0.01, 1.0, p3 - 0.06, 1.0, 0.05, 0.0

                                out      aout * adeclick * imaxamp
endin

</CsInstruments>
<CsScore>
f1 0 32768 10 1

i1 0 1      7.00  0.000001 0.8
i1 + 0.5    7.02  0.01    1.0
i1 + .      7.05  0.5     1.0
i1 + .      7.07  4.0     1.0
i1 + .      7.09  1.0     10.0
i1 + 2      7.06  1.0     25.0

e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Anthony Kozar
January 2008

New in Csound version 5.08

powoftwo

powoftwo — Performs a power-of-two calculation.

Description

Performs a power-of-two calculation.

Syntax

```
powoftwo(x)  (init-rate or control-rate args only)
```

Performance

powoftwo() function returns 2^x and allows positive and negatives numbers as argument. The range of values admitted in *powoftwo()* is -5 to +5 allowing a precision more fine than one cent in a range of ten octaves. If a greater range of values is required, use the slower opcode *pow*.

These functions are fast, because they read values stored in tables. Also they are very useful when working with tuning ratios. They work at i- and k-rate.

Examples

Here is an example of the *powoftwo* opcode. It uses the file *powoftwo.csd* [examples/powoftwo.csd].

Example 377. Example of the powoftwo opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o powoftwo.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  il = powoftwo(12)
  print il
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 4096.000
```

See Also

logbtwo, pow

Credits

Author: Gabriel Maldonado
Italy
June 1998

Author: John ffitch
University of Bath, Codemist, Ltd.
Bath, UK
July 1999

Example written by Kevin Conder.

New in Csound version 3.57

prealloc

prealloc — Creates space for instruments but does not run them.

Description

Creates space for instruments but does not run them.

Syntax

```
prealloc insnum, icount  
  
prealloc "insname", icount
```

Initialization

insnum -- instrument number

icount -- number of instrument allocations

"insname" -- A string (in double-quotes) representing a named instrument.

Performance

All instances of *prealloc* must be defined in the header section, not in the instrument body.

Examples

Here is an example of the *prealloc* opcode. It uses the file *prealloc.csd* [examples/prealloc.csd].

Example 378. Example of the prealloc opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  No messages  
-odac       -iadc      -d      ;;RT audio I/O  
; For Non-realtime output leave only the line below:  
; -o prealloc.wav -W ;; for file output any platform  
</CsOptions>  
<CsInstruments>  
  
; Initialize the global variables.  
sr = 44100  
kr = 4410  
ksmps = 10  
nchnls = 1  
  
; Pre-allocate memory for five instances of Instrument #1.  
prealloc 1, 5  
  
; Instrument #1  
instr 1  
; Generate a waveform, get the cycles per second from the 4th p-field.  
a1 oscil 6500, p4, 1  
out a1  
endin
```

```
</CsInstruments>
<CsScore>

; Just generate a nice, ordinary sine wave.
f 1 0 32768 10 1

; Play five instances of Instrument #1 for one second.
; Note that 4th p-field contains cycles per second.
i 1 0 1 220
i 1 0 1 440
i 1 0 1 880
i 1 0 1 1320
i 1 0 1 1760
e

</CsScore>
</CsoundSynthesizer>
```

See Also

cpuprc, maxalloc

Credits

Author: Gabriel Maldonado
Italy
July 1999

Example written by Kevin Conder.

New in Csound version 3.57

prepiano

prepiano — Creates a tone similar to a piano string prepared in a Cageian fashion.

Description

Audio output is a tone similar to a piano string, prepared with a number of rubbers and rattles. The method uses a physical model developed from solving the partial differential equation.

Syntax

```
ares prepiano ifreq, iNS, iD, iK, \  
    iT30, iB, kbcl, kbcr, imass, ifreq, iinit, ipos, ivel, isfreq, \  
    isspread[, irattles, irubbers]  
  
al,ar prepiano ifreq, iNS, iD, iK, \  
    iT30, iB, kbcl, kbcr, imass, ifreq, iinit, ipos, ivel, isfreq, \  
    isspread[, irattles, irubbers]
```

Initialization

ifreq -- The base frequency of the string.

iNS -- the number of strings involved. In a real piano 1, 2 or 3 strings are found in different frequency regions.

iD -- the amount each string other than the first is detuned from the main frequency, measured in cents.

iK -- dimensionless stiffness parameter.

iT30 -- 30 db decay time in seconds.

ib -- high-frequency loss parameter (keep this small).

imass -- the mass of the piano hammer.

ifreq -- the frequency of the natural vibrations of the hammer.

iinit -- the initial position of the hammer.

ipos -- position along the string that the strike occurs.

ivel -- normalized strike velocity.

isfreq -- scanning frequency of the reading place.

isspread -- scanning frequency spread.

irattles -- table number giving locations of any rattle(s).

irubbers -- table number giving locations of any rubbers(s).

The rattles and rubbers tables are collections of four values, preceded by a count. In the case of a rattle the four are position, mass density ratio of rattle/string, frequency of rattle and vertical length of the rattle. For the rubber the four are position, mass density ratio of rubber/string, frequency of rubber and the loss parameter.

Performance

A note is played on a piano string, with the arguments as below.

kbcL -- Boundary condition at left end of string (1 is clamped, 2 pivoting and 3 free).

kbcR -- Boundary condition at right end of string (1 is clamped, 2 pivoting and 3 free).

Note that changing the boundary conditions during playing may lead to glitches and is made available as an experiment.

Examples

Here is an example of the prepiano opcode. It uses the file *prepiano.csd* [examples/prepiano.csd].

Example 379. Example of the prepiano opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o prepiano.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2;

; Instrument #1.
instr 1
;;      fund NS detune stiffness decay loss (bndry) (hammer) scan prep
aa,ab prepiano 60, 3, 10, p4, 3, 0.002, 2, 2, 1, 5000, -0.01, p5, p6, 0, 0.1, 1, 2
outs aa*.75, ab*.75
endin
</CsInstruments>
<CsScore>
f1 0 8 2 1 0.6 10 100 0.001 ;; 1 rattle
f2 0 8 2 1 0.7 50 500 1000 ;; 1 rubber
i1 0.0 0.5 1 0.09 20
i1 0.5 . -1 0.09 40      ;; 1 -> skip initialisation
i1 1.0 . -1 0.09 60
i1 1.5 . -1 0.09 80
i1 2.0 1.8 -1 0.09 100
e
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Stefan Bilbao
University of Edinburgh, UK
Author: John ffitch
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 5.05

print

print — Displays the values init (i-rate) variables.

Description

These units will print orchestra init-values.

Syntax

```
print iarg [, iarg1] [, iarg2] [...]
```

Initialization

iarg, *iarg2*, ... -- i-rate arguments.

Performance

print -- print the current value of the i-time arguments (or expressions) *iarg* at every i-pass through the instrument.



Note

The *print* opcode will truncate decimal places and may not show the complete value. Csound's precision depends on whether it is the floats (32-bit) or double (64-bit) *version*, since most internal calculations use one of these formats. If you need more resolution in the console output, you can try *printf*.

Examples

Here is an example of the print opcode. It uses the file *print.csd* [examples/print.csd].

Example 380. Example of the print opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o print.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print the fourth p-field.
print p4
endin
```

```
</CsInstruments>
<CsScore>

; p4 = value to be printed.
; Play Instrument #1 for one second, p4 = 50.375.
i 1 0 1 50.375
; Play Instrument #1 for one second, p4 = 300.
i 1 1 1 300
; Play Instrument #1 for one second, p4 = -999.
i 1 2 1 -999
e

</CsScore>
</CsSoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  p4 = 50.375
instr 1:  p4 = 300.000
instr 1:  p4 = -999.000
```

See Also

disppfft, display, printk, printk2, printks , printf and prints

Credits

Example written by Kevin Conder.

Comments about the *inprds* parameter contributed by Rasmus Ekman.

printf

printf — printf-style formatted output

Description

printf and **printf_i** write formatted output, similarly to the C function printf(). **printf_i** runs at i-time only, while **printf** runs both at initialization and performance time.

Syntax

```
printf_i Sfmt, itrig, [iarg1[, iarg2[, ... ]]]
```

```
printf Sfmt, ktrig, [xarg1[, xarg2[, ... ]]]
```

Initialization

Sfmt -- format string, has the same format as in printf() and other similar C functions, except length modifiers (l, ll, h, etc.) are not supported. The following conversion specifiers are allowed:

- d, i, o, u, x, X, e, E, f, F, g, G, c, s

iarg1, *iarg2*, ... -- input arguments (max. 30) for format. Integer formats like %d round the input values to the nearest integer.

Performance

itrig -- if greater than zero the opcode performs the printing; otherwise it is a null operation.

ktrig -- if greater than zero and different from the value on the previous control cycle the opcode performs the requested printing. Initially this previous value is taken as zero.

xarg1, *xarg2*, ... -- input arguments (max. 30) for format. Integer formats like %d round the input values to the nearest integer. Note that only k-rate and i-rate arguments are valid (no a-rate printing)

Example

Credits

Author: Istvan Varga
2005

printk

printk — Prints one k-rate value at specified intervals.

Description

Prints one k-rate value at specified intervals.

Syntax

```
printk itime, kval [, ispace]
```

Initialization

itime -- time in seconds between printings.

ispace (optional, default=0) -- number of spaces to insert before printing. (default: 0, max: 130)

Performance

kval -- The k-rate values to be printed.

printk prints one k-rate value on every k-cycle, every second or at intervals specified. First the instrument number is printed, then the absolute time in seconds, then a specified number of spaces, then the *kval* value. The variable number of spaces enables different values to be spaced out across the screen - so they are easier to view.

This opcode can be run on every k-cycle it is run in the instrument. To every accomplish this, set *itime* to 0.

When *itime* is not 0, the opcode print on the first k-cycle it is called, and subsequently when every *itime* period has elapsed. The time cycles start from the time the opcode is initialized - typically the initialization of the instrument.

Examples

Here is an example of the printk opcode. It uses the file *printk.csd* [examples/printk.csd].

Example 381. Example of the printk opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o printk.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1
```

```
; Instrument #1.
instr 1
; Change a value linearly from 0 to 100,
; over the period defined by p3.
kval line 0, p3, 100

; Print the value of kval, once per second.
printk 1, kval
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 5 seconds.
i 1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

i	1	time	0.00002:	0.00000
i	1	time	1.00002:	20.01084
i	1	time	2.00002:	40.02999
i	1	time	3.00002:	60.04914
i	1	time	4.00002:	79.93327

See Also

printk2 and *printks*

Credits

Author: Robin Whittle
Australia
May 1997

Example written by Kevin Conder.

Thanks goes to Luis Jure for pointing out a mistake wit the *itime* parameter.

printk2

printk2 — Prints a new value every time a control variable changes.

Description

Prints a new value every time a control variable changes.

Syntax

```
printk2 kvar [, inumspaces]
```

Initialization

inumspaces (optional, default=0) -- number of space characters printed before the value of *kvar*

Performance

kvar -- signal to be printed

Derived from Robin Whittle's *printk*, prints a new value of *kvar* each time *kvar* changes. Useful for monitoring MIDI control changes when using sliders.



Warning

WARNING! Don't use this opcode with normal, continuously variant k-signals, because it can hang the computer, as the rate of printing is too fast.

Examples

Here is an example of the printk2 opcode. It uses the file *printk2.csd* [examples/printk2.csd].

Example 382. Example of the printk2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o printk2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1.
instr 1
; Change a value linearly from 0 to 10,
; over the period defined by p3.
kva11 line 0, p3, 10
```

```
    ; If kval1 is greater than or equal to 5,  
    ; then kval=2, else kval=1.  
    kval2 = (kval1 >= 5 ? 2 : 1)  
  
    ; Print the value of kval2 when it changes.  
    printk2 kval2  
endin  
  
</CsInstruments>  
<CsScore>  
  
    ; Play Instrument #1 for 5 seconds.  
    i 1 0 5  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

Its output should include a line like this:

```
i1      1.00000  
i1      2.00000
```

See Also

printk and *printks*

Credits

Author: Gabriel Maldonado
Italy
1998

Example written by Kevin Conder.

New in Csound version 3.48

printks

printks — Prints at k-rate using a printf() style syntax.

Description

Prints at k-rate using a printf() style syntax.

Syntax

```
printks "string", itime [, kval1] [, kval2] [...]
```

Initialization

"string" -- the text string to be printed. Can be up to 8192 characters and must be in double quotes.

itime -- time in seconds between printings.

Performance

kval1, *kval2*, ... (optional) -- The k-rate values to be printed. These are specified in *"string"* with the standard C value specifier (%f, %d, etc.) in the order given.

In Csound version 4.23, you can use as many *kval* variables as you like. In versions prior to 4.23, you must specify 4 and only 4 kvals (using 0 for unused kvals).

printks prints numbers and text which can be i-time or k-rate values. *printks* is highly flexible, and if used together with cursor positioning codes, could be used to write specific values to locations in the screen as the Csound processing proceeds.

A special mode of operation allows this *printks* to convert *kval1* input parameter into a 0 to 255 value and to use it as the first character to be printed. This enables a Csound program to send arbitrary characters to the console. To achieve this, make the first character of the string a # and then, if desired continue with normal text and format specifiers.

This opcode can be run on every k-cycle it is run in the instrument. To every accomplish this, set *itime* to 0.

When *itime* is not 0, the opcode print on the first k-cycle it is called, and subsequently when every *itime* period has elapsed. The time cycles start from the time the opcode is initialized - typically the initialization of the instrument.

Print Output Formatting

All standard C language printf() control characters may be used. For example, if *kval1* = 153.26789 then some common formatting options are:

1. %f prints with full precision: 153.26789
2. %5.2f prints: 153.26
3. %d prints integers-only: 153
4. %c treats *kval1* as an ascii character code.

In addition to all the `printf()` codes, `printks` supports these useful character codes:

printks Code	Character Code
\\r, \\R, %r, or %R	return character (\r)
\\n, \\N, %n, %N	newline character (\n)
\\t, \\T, %t, or %T	tab character (\t)
%!	semicolon character (;) This was needed because a “;” is interpreted as an comment.
^	escape character (0x1B)
^ ^	caret character (^)
~	ESC[(escape+[is the escape sequence for ANSI consoles)
~~	tilde (~)

For more information about `printf()` formatting, consult any C language documentation.



Note

Prior to version 4.23, only the `%f` format code was supported.

Examples

Here is an example of the `printks` opcode. It uses the file *printks.csd* [examples/printks.csd].

Example 383. Example of the `printks` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o printks.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1.
instr 1
; Change a value linearly from 0 to 100,
; over the period defined by p3.
kup line 0, p3, 100
; Change a value linearly from 30 to 10,
; over the period defined by p3.
kdown line 30, p3, 10

; Print the value of kup and kdown, once per second.
printks "kup = %f, kdown = %f\n", 1, kup, kdown
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 5 seconds.
i 1 0 5
```

e

```
</CsScore>  
</CsoundSynthesizer>
```

Its output should include lines like this:

```
kup = 0.000000, kdown = 30.000000  
kup = 20.010843, kdown = 25.962524  
kup = 40.029991, kdown = 21.925049  
kup = 60.049141, kdown = 17.887573  
kup = 79.933266, kdown = 13.872493
```

See Also

printk2 and *printk*

Credits

Author: Robin Whittle
Australia
May 1997

Example written by Kevin Conder.

Thanks goes to Luis Jure for pointing out a mistake with the *itime* parameter.

Thanks to Matt Ingalls, updated the documentation for version 4.23.

prints

prints — Prints at init-time using a printf() style syntax.

Description

Prints at init-time using a printf() style syntax.

Syntax

```
prints "string" [, kval1] [, kval2] [...]
```

Initialization

"string" -- the text string to be printed. Can be up to 8192 characters and must be in double quotes.

Performance

kval1, kval2, ... (optional) -- The k-rate values to be printed. These are specified in *"string"* with the standard C value specifier (%f, %d, etc.) in the order given. Use 0 for those which are not used.

prints is similar to the *printks* opcode except it operates at init-time instead of k-rate. For more information about output formatting, please look at *printks's documentation*.

Examples

Here is an example of the prints opcode. It uses the file *prints.csd* [examples/prints.csd].

Example 384. Example of the prints opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o prints.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Matt Ingalls, edited by Kevin Conder. */
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Init-time print.
prints "%2.3f\\t%!%!%!%!%;semicolons!\\n", 1234.56789
endin

</CsInstruments>
<CsScore>

/* Written by Matt Ingalls, edited by Kevin Conder. */
; Play instrument #1.
i 1 0 0.004
```

```
</CsScore>  
</CsoundSynthesizer>
```

Its output should include a line like this:

```
1234.568          ;;;;;;semicolons!
```

See Also

printks

Credits

Author: Matt Ingalls
January 2003

product

product — Multiplies any number of a-rate signals.

Description

Multiplies any number of a-rate signals.

Syntax

```
ares product asig1, asig2 [, asig3] [...]
```

Performance

asig1, asig2, asig3, ... -- a-rate signals to be multiplied.

Credits

Author: Gabriel Maldonado
Italy
April 1999

New in Csound version 3.54

pset

pset — Defines and initializes numeric arrays at orchestra load time.

Description

Defines and initializes numeric arrays at orchestra load time.

Syntax

```
pset icon1 [, icon2] [...]
```

Initialization

icon1, icon2, ... -- preset values for a MIDI instrument

pset (optional) defines and initializes numeric arrays at orchestra load time. It may be used as an orchestra header statement (i.e. instrument 0) or within an instrument. When defined within an instrument, it is not part of its i-time or performance operation, and only one statement is allowed per instrument. These values are available as i-time defaults. When an instrument is triggered from MIDI it only gets p1 and p2 from the event, and p3, p4, etc. will receive the actual preset values.

Examples

The example below illustrates *pset* as used within an instrument.

```
instr 1  
  pset 0,0,3,4,5,6 ; pfield substitutes  
  al oscil 10000, 440, p6
```

See Also

strset

ptrack

ptrack — Tracks the pitch of a signal.

Description

ptrack takes an input signal, splits it into *ihopsize* blocks and using a STFT method, extracts an estimated pitch for its fundamental frequency as well as estimating the total amplitude of the signal in dB, relative to full-scale (0dB). The method implies an analysis window size of $2*ihopsize$ samples (overlapping by 1/2 window), which has to be a power-of-two, between 128 and 8192 (hopsize between 64 and 4096). Smaller windows will give better time precision, but worse frequency accuracy (esp. in low fundamentals). This opcode is based on an original algorithm by M. Puckette.

Syntax

```
kcps, kamp ptrack asig, ihopsize[,ipeaks]
```

Initialization

ihopsize -- size of the analysis 'hop', in samples, required to be power-of-two (min 64, max 4096). This is the period between measurements.

ipeaks, ihi -- number of spectral peaks to use in the analysis, defaults to 20 (optional)

Performance

kcps -- estimated pitch in Hz.

kamp -- estimated amplitude in dB relative to full-scale (0dB) (ie. always ≤ 0).

ptrack analyzes the input signal, *asig*, to give a pitch/amplitude pair of outputs, for the fundamental of a monophonic signal. The output is updated every *sr/ihopsize* seconds.

Examples

Here is an example of the *ptrack* opcode. This example uses the files *ptrack.csd* [examples/ptrack.csd].

Example 385. Example of the ptrack opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No display
-odac        -iadc     -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>
sr= 44100
ksmps = 16
nchnls= 1

;Example by Victor Lazzarini 2007

instr 1
  al inch 1          ; take an input signal
  kf,ka ptrack al, 512 ; pitch track with winsize=1024
  kcps port kf, 0.01  ; smooth freq
```



```
kamp port ka, 0.01 ; smooth amp
; drive an oscillator
aout oscili ampdb(kamp)*0dbfs, kcps, 1

out aout
endin

</CsInstruments>
<CsScore>
i1 0 3600
e
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Victor Lazzarini;
NUI, Maynooth.
Maynooth, Ireland
March, 2007

New in Csound version 5.05

puts

puts — Print a string constant or variable

Description

puts prints a string with an optional newline at the end whenever the trigger signal is positive and changes.

Syntax

```
puts Sstr, ktrig[, inonl]
```

Initialization

Sstr -- string to be printed

inonl (optional, defaults to 0) -- if non-zero, disables the default printing of a newline character at the end of the string

Performance

ktrig -- trigger signal, should be valid at i-time. The string is printed at initialization time if ktrig is positive, and at performance time whenever ktrig is both positive and different from the previous value. Use a constant value of 1 to print once at note initialization.

Credits

Author: Istvan Varga
2005

push

push — Pushes a value into the global stack.

Description

Pushes a value into the global stack.

Syntax

```
push  xval1, [xval2, ... , xval31]
```

```
push  ival1, [ival2, ... , ival31]
```

Initialization

ival1 ... ival31 - values to be pushed into the stack.

Performance

xval1 ... xval31 - values to be pushed into the stack.

The given values are pushed into the global stack as a bundle. The global stack works in LIFO order: after multiple *push* calls, *pop* should be used in reverse order.

Each *push* or *pop* operation can work on a "bundle" of multiple variables. When using *pop*, the number, type, and order of items must match those used by the corresponding *push*. That is, after a 'push Sfoo, ibar', you must call something like 'pop Sbar, ifoo', and not e.g. two separate 'pop' statements.

push and *pop* opcodes can take variables of any type (i-, k-, a- and strings). Use of any combination of i, k, a, and S types is allowed. Variables of type 'a' and 'k' are passed at performance time only, while 'i' and 'S' are passed at init time only.

push/pop for a, k, i, and S types copy data by value. By contrast, *push_f* only pushes a "reference" to the f-signal, and then the corresponding *pop_f* will copy directly from the original variable to its output signal. For this reason, changing the source f-signal of *push_f* before *pop_f* is called is not recommended, and if the instrument instance owning the variable that was passed by *push_f* is deactivated before *pop_f* is called, undefined behavior may occur.

Any stack errors (trying to push when there is no more space, or pop from an empty stack, inconsistent number or type of arguments, etc.) are fatal and terminate performance.

See also

stack, *pop*, *pop_f* and *push_f*.

Credits

By: Istvan Varga.

2006

push_f

`push_f` — Pushes an f-sig frame into the global stack.

Description

Pushes an f-sig frame into the global stack.

Syntax

```
push_f fsig
```

Performance

fsig - f-signal to be pushed into the stack.

The values are pushed into the global stack. The global stack works in LIFO order: after multiple *push_f* calls, *pop_f* should be used in reverse order.

push/pop for a, k, i, and S types copy data by value. By contrast, *push_f* only pushes a "reference" to the f-signal, and then the corresponding *pop_f* will copy directly from the original variable to its output signal. For this reason, changing the source f-signal of *push_f* before *pop_f* is called is not recommended, and if the instrument instance owning the variable that was passed by *push_f* is deactivated before *pop_f* is called, undefined behavior may occur.

pop_f and *push_f* can only take a single argument, and the data is passed both at init and performance time.

Any stack errors (trying to push when there is no more space, or pop from an empty stack, inconsistent number or type of arguments, etc.) are fatal and terminate performance.

See also

stack, *push*, *pop* and *pop_f*.

Credits

By: Istvan Varga.

2006

pvadd

pvadd — Reads from a *pvoc* file and uses the data to perform additive synthesis.

Description

pvadd reads from a *pvoc* file and uses the data to perform additive synthesis using an internal array of interpolating oscillators. The user supplies the wave table (usually one period of a sine wave), and can choose which analysis bins will be used in the re-synthesis.

Syntax

```
ares pvadd ktimpnt, kfmod, ifilcod, ifn, ibins [, ibinoffset] \  
      [, ibinincr] [, iextractmode] [, ifreqlim] [, igatefn]
```

Initialization

ifilcod -- integer or character-string denoting a control-file derived from *pvanal* analysis of an audio signal. An integer denotes the suffix of a file *pvoc.m*; a character-string (in double quotes) gives a filename, optionally a full pathname. If not fullpath, the file is sought first in the current directory, then in the one given by the environment variable *SADIR* (if defined). *pvoc* control files contain data organized for fft resynthesis. Memory usage depends on the size of the files involved, which are read and held entirely in memory during computation but are shared by multiple calls (see also *lpread*).

ifn -- table number of a stored function containing a sine wave.

ibins -- number of bins that will be used in the resynthesis (each bin counts as one oscillator in the re-synthesis)

ibinoffset (optional) -- is the first bin used (it is optional and defaults to 0).

ibinincr (optional) -- sets an increment by which *pvadd* counts up from *ibinoffset* for *ibins* components in the re-synthesis (see below for a further explanation).

ieextractmode (optional) -- determines if spectral extraction will be carried out and if so whether components that have changes in frequency below *ifreqlim* or above *ifreqlim* will be discarded. A value for *ieextractmode* of 1 will cause *pvadd* to synthesize only those components where the frequency difference between analysis frames is greater than *ifreqlim*. A value of 2 for *ieextractmode* will cause *pvadd* to synthesize only those components where the frequency difference between frames is less than *ifreqlim*. The default values for *ieextractmode* and *ifreqlim* are 0, in which case a simple resynthesis will be done. See examples below.

igatefn (optional) -- is the number of a stored function which will be applied to the amplitudes of the analysis bins before resynthesis takes place. If *igatefn* is greater than 0 the amplitudes of each bin will be scaled by *igatefn* through a simple mapping process. First, the amplitudes of all of the bins in all of the frames in the entire analysis file are compared to determine the maximum amplitude value. This value is then used create normalized amplitudes as indeces into the stored function *igatefn*. The maximum amplitude will map to the last point in the function. An amplitude of 0 will map to the first point in the function. Values between 0 and 1 will map accordingly to points along the function table. This will be made clearer in the examples below.

Performance

ktimpnt and *kfmod* are used in the same way as in *pvoc*.

Examples

```
ptime line 0, p3, p3
```

```
asig pvadd ptime, 1, "oboe.pvoc", 1, 100, 2
```

In the above, *ibins* is 100 and *ibinoffset* is 2. Using these settings the resynthesis will contain 100 components beginning with bin #2 (bins are counted starting with 0). That is, resynthesis will be done using bins 2-101 inclusive. It is usually a good idea to begin with bin 1 or 2 since the 0th and often 1st bin have data that is neither necessary nor even helpful for creating good clean resynthesis.

```
ptime line 0, p3, p3
asig pvadd ptime, 1, "oboe.pvoc", 1, 100, 2, 2
```

The above is the same as the previous example with the addition of the value 2 used for the optional *ibinincr* argument. This result will still result in 100 components in the resynthesis, but *pvadd* will count through the bins by 2 instead of by 1. It will use bins 2, 4, 6, 8, 10, and so on. For *ibins*=10, *ibinoffset*=10, and *ibinincr*=10, *pvadd* would use bins 10, 20, 30, 40, up to and including 100.

Below is an example using spectral extraction. In this example *iextractmode* is one and *ifreqlim* is 9. This will cause *pvadd* to synthesize only those bins where the frequency deviation, averaged over 6 frames, is greater than 9.

```
ptime line 0, p3, p3
asig pvadd ptime, 1, "oboe.pvoc", 1, 100, 2, 2, 1, 9
```

If *iextractmode* were 2 in the above, then only those bins with an average frequency deviation of less than 9 would be synthesized. If tuned correctly, this technique can be used to separate the pitched parts of the spectrum from the noisy parts. In practice this depends greatly on the type of sound, the quality of the recording and digitization, and also on the analysis window size and frame increment.

Next is an example using amplitude gating. The last 2 in the argument list stands for *f2* in the score.

```
asig pvadd ptime, 1, "oboe.pvoc", 1, 100, 2, 2, 0, 0, 2
```

Suppose the score for the above were to contain:

```
f2 0 512 7 0 256 1 256 1
```

Then those bins with amplitudes of 50% of the maximum or greater would be left unchanged, while those with amplitudes less than 50% of the maximum would be scaled down. In this case the lower the amplitude the more severe the scaling down would be. But suppose the score contains:

```
f2 0 512 5 1 512 .001
```

In this case lower amplitudes will be left unchanged and greater ones will be scaled down, turning the sound “upside-down” in terms of the amplitude spectrum! Functions can be arbitrarily complex. Just remember that the normalized amplitude values of the analysis are themselves the indices into the function.

Finally, both spectral extraction and amplitude gating can be used together. The example below will synthesize only those components that with a frequency deviation of less than 5Hz per frame and it will scale the amplitudes according to F2.

```
asig pvadd ktime, 1, "oboe.pvoc", 1, 100, 1, 1, 2, 5, 2
```



USEFUL HINTS

By using several *pvadd* units together, one can gradually fade in different parts of the resynthesis, creating various “filtering” effects. The author uses *pvadd* to synthesize one bin at a time to have control over each separate component of the re-synthesis.

If any combination of *ibins*, *ibinoffset*, and *ibinincr*, creates a situation where *pvadd* is asked to use a bin number greater than the number of bins in the analysis, it will just use all of the available bins, and give no complaint. So to use every bin just make *ibins* a big number (ie. 2000).

Expect to have to scale up the amplitudes by factors of 10-100, by the way.

Credits

Author: Richard Karpen
Seattle, WA USA
1998

New in Csound version 3.48, additional arguments version 3.56

pvbufread

pvbufread — Reads from a phase vocoder analysis file and makes the retrieved data available.

Description

pvbufread reads from a *pvoc* file and makes the retrieved data available to any following *pvinterp* and *pvcross* units that appear in an instrument before a subsequent *pvbufread* (just as *lpread* and *lpreson* work together). The data is passed internally and the unit has no output of its own.

Syntax

```
pvbufread ktimepnt, ifile
```

Initialization

ifile -- the *pvoc* number (n in *pvoc.n*) or the name in quotes of the analysis file made using *pvanal*. (See *pvoc*.)

Performance

ktimepnt -- the passage of time, in seconds, through this file. *ktimepnt* must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the analysis file.

Examples

The example below shows an example using *pvbufread* with *pvinterp* to interpolate between the sound of an oboe and the sound of a clarinet. The value of *kinterp* returned by a *linseg* is used to determine the timing of the transitions between the two sounds. The interpolation of frequencies and amplitudes are controlled by the same factor in this example, but for other effects it might be interesting to not have them synchronized in this way. In this example the sound will begin as a clarinet, transform into the oboe and then return again to the clarinet sound. The value of *kfreqscale2* is 1.065 because the oboe in this case is a semitone higher in pitch than the clarinet and this brings them approximately to the same pitch. The value of *kampscale2* is .75 because the analyzed clarinet was somewhat louder than the analyzed oboe. The setting of these two parameters make the transition quite smooth in this case, but such adjustments are by no means necessary or even advocated.

```
ktime1 line      0, p3, 3.5 ; used as index in the "oboe.pvoc" file
ktime2 line      0, p3, 4.5 ; used as index in the "clar.pvoc" file
kinterp linseg   1, p3*.15, 1, p3*.35, 0, p3*.25, 0, p3*.15, 1, p3*.1, 1
          pvbufread ktime1, "oboe.pvoc"
apv          pvinterp ktime2,1,"clar.pvoc",1,1.065,1,.75,1-kinterp,1-kinterp
```

Below is an example using *pvbufread* with *pvcross*. In this example the amplitudes used in the re-synthesis gradually change from those of the oboe to those of the clarinet. The frequencies, of course, remain those of the clarinet throughout the process since *pvcross* does not use the frequency data from the file read by *pvbufread*.

```
ktime1 line      0, p3, 3.5 ; used as index in the "oboe.pvoc" file
ktime2 line      0, p3, 4.5 ; used as index in the "clar.pvoc" file
kcross expon     .001, p3, 1
```



```
apv      pvbufread ktimel, "oboe.pvoc"  
         pvcross   ktime2, 1, "clar.pvoc", 1-kcross, kcross
```

See Also

pvcross, pvinterp, pvread, tableseg, tablexseg

Credits

Author: Richard Karpen
Seattle, WA USA
1997

pvcross

pvcross — Applies the amplitudes from one phase vocoder analysis file to the data from a second file.

Description

pvcross applies the amplitudes from one phase vocoder analysis file to the data from a second file and then performs the resynthesis. The data is passed, as described above, from a previously called *pvbufread* unit. The two k-rate amplitude arguments are used to scale the amplitudes of each files separately before they are added together and used in the resynthesis (see below for further explanation). The frequencies of the first file are not used at all in this process. This unit simply allows for cross-synthesis through the application of the amplitudes of the spectra of one signal to the frequencies of a second signal. Unlike *pvinterp*, *pvcross* does allow for the use of the *ispecwp* as in *pvoc* and *vpvoc*.

Syntax

```
ares pvcross ktmpnt, kfmod, ifile, kampscale1, kampscale2 [, ispecwp]
```

Initialization

ifile -- the *pvoc* number (n in *pvoc.n*) or the name in quotes of the analysis file made using *pvanal*. (See *pvoc*.)

ispecwp (optional, default=0) -- if non-zero, attempts to preserve the spectral envelope while its frequency content is varied by *kfmod*. The default value is zero.

Performance

ktmpnt -- the passage of time, in seconds, through this file. *ktmpnt* must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the analysis file.

kfmod -- a control-rate transposition factor: a value of 1 incurs no transposition, 1.5 transposes up a perfect fifth, and .5 down an octave.

kampscale1, *kampscale2* -- used to scale the amplitudes stored in each frame of the phase vocoder analysis file. *kampscale1* scale the amplitudes of the data from the file read by the previously called *pvbufread*. *kampscale2* scale the amplitudes of the file named by *ifile*.

By using these arguments, it is possible to adjust these values before applying the interpolation. For example, if file1 is much louder than file2, it might be desirable to scale down the amplitudes of file1 or scale up those of file2 before interpolating. Likewise one can adjust the frequencies of each to bring them more in accord with one another (or just the opposite, of course!) before the interpolation is performed.

Examples

Below is an example using *pvbufread* with *pvcross*. In this example the amplitudes used in the resynthesis gradually change from those of the oboe to those of the clarinet. The frequencies, of course, remain those of the clarinet throughout the process since *pvcross* does not use the frequency data from the file read by *pvbufread*.

```
ktime1  line    0, p3, 3.5 ; used as index in the "oboe.pvoc" file
ktime2  line    0, p3, 4.5 ; used as index in the "clar.pvoc" file
kcross  expon   .001, p3, 1
        pvbufread ktime1, "oboe.pvoc"
apv      pvcross  ktime2, 1, "clar.pvoc", 1-kcross, kcross
```

See Also

pvbufread, pvinterp, pvread, tableseg, tablexseg

Credits

Author: Richard Karpen
Seattle, Wash
1997

New in version 3.44

pvinterp

pvinterp — Interpolates between the amplitudes and frequencies of two phase vocoder analysis files.

Description

pvinterp interpolates between the amplitudes and frequencies, on a bin by bin basis, of two phase vocoder analysis files (one from a previously called *pdbufread* unit and the other from within its own argument list), allowing for user defined transitions between analyzed sounds. It also allows for general scaling of the amplitudes and frequencies of each file separately before the interpolated values are calculated and sent to the resynthesis routines. The *kfmod* argument in *pvinterp* performs its frequency scaling on the frequency values after their derivation from the separate scaling and subsequent interpolation is performed so that this acts as an overall scaling value of the new frequency components.

Syntax

```
ares pvinterp ktmpnt, kfmod, ifile, kfreqscale1, kfreqscale2, \
      kampscale1, kampscale2, kfreqinterp, kampinterp
```

Initialization

ifile -- the *pvoc* number (n in *pvoc.n*) or the name in quotes of the analysis file made using *pvanal*. (See *pvoc*.)

Performance

ktmpnt -- the passage of time, in seconds, through this file. *ktmpnt* must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the analysis file.

kfmod -- a control-rate transposition factor: a value of 1 incurs no transposition, 1.5 transposes up a perfect fifth, and .5 down an octave.

kfreqscale1, *kfreqscale2*, *kampscale1*, *kampscale2* -- used in *pvinterp* to scale the frequencies and amplitudes stored in each frame of the phase vocoder analysis file. *kfreqscale1* and *kampscale1* scale the frequencies and amplitudes of the data from the file read by the previously called *pdbufread* (this data is passed internally to the *pvinterp* unit). *kfreqscale2* and *kampscale2* scale the frequencies and amplitudes of the file named by *ifile* in the *pvinterp* argument list and read within the *pvinterp* unit.

By using these arguments, it is possible to adjust these values before applying the interpolation. For example, if file1 is much louder than file2, it might be desirable to scale down the amplitudes of file1 or scale up those of file2 before interpolating. Likewise one can adjust the frequencies of each to bring them more in accord with one another (or just the opposite, of course!) before the interpolation is performed.

kfreqinterp, *kampinterp* -- used in *pvinterp*, determine the interpolation distance between the values of one phase vocoder file and the values of a second file. When the value of *kfreqinterp* is 1, the frequency values will be entirely those from the first file (read by the *pdbufread*), post scaling by the *kfreqscale1* argument. When the value of *kfreqinterp* is 0 the frequency values will be those of the second file (read by the *pvinterp* unit itself), post scaling by *kfreqscale2*. When *kfreqinterp* is between 0 and 1 the frequency values will be calculated, on a bin, by bin basis, as the percentage between each pair of frequencies (in other words, *kfreqinterp*=.5 will cause the frequencies values to be half way between the values in the set of data from the first file and the set of data from the second file).

kampinterp works in the same way upon the amplitudes of the two files. Since these are k-rate argu-

ments, the percentages can change over time making it possible to create many kinds of transitions between sounds.

Examples

The example below shows an example using *pvbufread* with *pvinterp* to interpolate between the sound of an oboe and the sound of a clarinet. The value of *kinterp* returned by a *linseg* is used to determine the timing of the transitions between the two sounds. The interpolation of frequencies and amplitudes are controlled by the same factor in this example, but for other effects it might be interesting to not have them synchronized in this way. In this example the sound will begin as a clarinet, transform into the oboe and then return again to the clarinet sound. The value of *kfreqscale2* is 1.065 because the oboe in this case is a semitone higher in pitch than the clarinet and this brings them approximately to the same pitch. The value of *kampscale2* is .75 because the analyzed clarinet was somewhat louder than the analyzed oboe. The setting of these two parameters make the transition quite smooth in this case, but such adjustments are by no means necessary or even advocated.

```
ptime1  line      0, p3, 3.5 ; used as index in the "oboe.pvoc" file
ptime2  line      0, p3, 4.5 ; used as index in the "clar.pvoc" file
kinterp linseg    1, p3*.15, 1, p3*.35, 0, p3*.25, 0, p3*.15, 1, p3*.1, 1
pvbufread kptime1, "oboe.pvoc"
apv      pvinterp kptime2,1,"clar.pvoc",1,1.065,1,.75,1-kinterp,1-kinterp
```

See Also

pvbufread, *pvcross*, *pvread*, *tableseg*, *tablexseg*

Credits

Author: Richard Karpen
Seattle, Wash
1997

pvoc

pvoc — Implements signal reconstruction using an fft-based phase vocoder.

Description

Implements signal reconstruction using an fft-based phase vocoder.

Syntax

```
ares pvoc ktmpnt, kfmmod, ifilcod [, ispecwp] [, iextractmode] \  
      [, ifreqlim] [, igatefn]
```

Initialization

ifilcod -- integer or character-string denoting a control-file derived from analysis of an audio signal. An integer denotes the suffix of a file *pvoc.m*; a character-string (in double quotes) gives a filename, optionally a full pathname. If not fullpath, the file is sought first in the current directory, then in the one given by the environment variable *SADIR* (if defined). *pvoc* control contains breakpoint amplitude and frequency envelope values organized for fft resynthesis. Memory usage depends on the size of the files involved, which are read and held entirely in memory during computation but are shared by multiple calls (see also *lpread*).

ispecwp (optional) -- if non-zero, attempts to preserve the spectral envelope while its frequency content is varied by *kfmmod*. The default value is zero.

extractmode (optional) -- determines if spectral extraction will be carried out and if so whether components that have changes in frequency below *ifreqlim* or above *ifreqlim* will be discarded. A value for *extractmode* of 1 will cause *pvadd* to synthesize only those components where the frequency difference between analysis frames is greater than *ifreqlim*. A value of 2 for *extractmode* will cause *pvadd* to synthesize only those components where the frequency difference between frames is less than *ifreqlim*. The default values for *extractmode* and *ifreqlim* are 0, in which case a simple resynthesis will be done. See examples under *pvadd* for how to use spectral extraction.

igatefn (optional) -- the number of a stored function which will be applied to the amplitudes of the analysis bins before resynthesis takes place. If *igatefn* is greater than 0 the amplitudes of each bin will be scaled by *igatefn* through a simple mapping process. First, the amplitudes of all of the bins in all of the frames in the entire analysis file are compared to determine the maximum amplitude value. This value is then used create normalized amplitudes as indices into the stored function *igatefn*. The maximum amplitude will map to the last point in the function. An amplitude of 0 will map to the first point in the function. Values between 0 and 1 will map accordingly to points along the function table. See examples under *pvadd* for how to use amplitude gating.

Performance

ktmpnt -- The passage of time, in seconds, through the analysis file. *ktmpnt* must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the analysis file.

kfmmod -- a control-rate transposition factor: a value of 1 incurs no transposition, 1.5 transposes up a perfect fifth, and .5 down an octave.

pvoc implements signal reconstruction using an fft-based phase vocoder. The control data stems from a precomputed analysis file with a known frame rate.

This implementation of *pvoc* was originally written by Dan Ellis. It is based in part on the system of Mark Dolson, but the pre-analysis concept is new. The spectral extraction and amplitude gating (new in Csound version 3.56) were added by Richard Karpen based on functions in SoundHack by

Tom Erbe.

See Also

vpvoc, *PVANAL*.

Credits

Authors: Dan Ellis and Richard Karpen
Seattle, Wash
1997

pvread

pvread — Reads from a phase vocoder analysis file and returns the frequency and amplitude from a single analysis channel or bin.

Description

pvread reads from a *pvoc* file and returns the frequency and amplitude from a single analysis channel or bin. The returned values can be used anywhere else in the Csound instrument. For example, one can use them as arguments to an oscillator to synthesize a single component from an analyzed signal or a bank of *pvreads* can be used to resynthesize the analyzed sound using additive synthesis by passing the frequency and magnitude values to a bank of oscillators.

Syntax

```
kfreq, kamp pvread ktmpnt, ifile, ibin
```

Initialization

ifile -- the *pvoc* number (n in *pvoc.n*) or the name in quotes of the analysis file made using *pvanal*. (See *pvoc*.)

ibin -- the number of the analysis channel from which to return frequency in Hz and magnitude.

Performance

kfreq, *kamp* -- outputs of the *pvread* unit. These values, retrieved from a phase vocoder analysis file, represent the values of frequency and amplitude from a single analysis channel specified in the *ibin* argument. Interpolation between analysis frames is performed at k-rate resolution and dependent of course upon the rate and direction of *ktmpnt*.

ktmpnt -- the passage of time, in seconds, through this file. *ktmpnt* must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the analysis file.

Examples

The example below shows the use *pvread* to synthesize a single component from a phase vocoder analysis file. It should be noted that the *kfreq* and *kamp* outputs can be used for any kind of synthesis, filtering, processing, and so on.

```
ktime      line    0, p3, 3
kfreq, kamp  pvread ktime, "pvoc.file", 7 ; read
                                     ;data from 7th analysis bin.
asig        oscili  kamp, kfreq, 1      ; function 1
                                     ;is a stored sine
```

See Also

pvbufread, *pvcross*, *pvinterp*, *tableseg*, *tablexseg*

Credits

Author: Richard Karpen
Seattle, Wash
1997

New in version 3.44

pvsadsyn

pvsadsyn — Resynthesize using a fast oscillator-bank.

Description

Resynthesize using a fast oscillator-bank.

Syntax

```
ares pvsadsyn fsrc, inoscs, kfmod [, ibinoffset] [, ibinincr] [, iinit]
```

Initialization

inoscs -- The number of analysis bins to synthesise. Cannot be larger than the size of *fsrc* (see *pvsinfo*), e.g. as created by *pvsanal*. Processing time is directly proportional to *inoscs*.

ibinoffset (optional, default=0) -- The first (lowest) bin to resynthesise, counting from 0 (default = 0).

ibinincr (optional) -- Starting from bin *ibinoffset*, resynthesize bins *ibinincr* apart.

iinit (optional) -- Skip reinitialization. This is not currently implemented for any of these opcodes, and it remains to be seen if it is even practical.

Performance

kfmod -- Scale all frequencies by factor *kfmod*. 1.0 = no change, 2 = up one octave.

pvsadsyn is experimental, and implements the oscillator bank using a fast direct calculation method, rather than a lookup table. This takes advantage of the fact, empirically arrived at, that for the analysis rates generally used, (and presuming analysis using *pvsanal*, where frequencies in a bin change only slightly between frames) it is not necessary to interpolate frequencies between frames, only amplitudes. Accurate resynthesis is often contingent on the use of *pvsanal* with *iwinsize* = *ifftsize**2.

This opcode is the most likely to change, or be much extended, according to feedback and advice from users. It is likely that a full interpolating table-based method will be added, via a further optional *iarg*. The parameter list to *pvsadsyn* mimics that for *pvadd*, but excludes spectral extraction.

Examples

```
; resynth the first 100 odd-numbered bins, with pitch scaling envelope.
kpch linseg 1,p3/3,1,p3/3,1.5,p3/3,1
aout pvsadsyn fsrc, 100,kpch,1,2
```

See Also

pvsynth

Credits

Author: Richard Dobson
August 2001

New in version 4.13

pvsanal

pvsanal — Generate an fsig from a mono audio source ain, using phase vocoder overlap-add analysis.

Description

Generate an fsig from a mono audio source ain, using phase vocoder overlap-add analysis.

Syntax

```
fsig pvsanal ain, ifftsize, ioverlap, iwinsize, iwintype [, iformat] [, iinit]
```

Initialization

ifftsize -- The FFT size in samples. Need not be a power of two (though these are especially efficient), but must be even. Odd numbers are rounded up internally. *ifftsize* determines the number of analysis bins in *fsig*, as $\text{ifftsize}/2 + 1$. For example, where *ifftsize* = 1024, *fsig* will contain 513 analysis bins, ordered linearly from the fundamental to Nyquist. The fundamental of analysis (which in principle gives the lowest resolvable frequency) is determined as $\text{sr}/\text{ifftsize}$. Thus, for the example just given and assuming $\text{sr} = 44100$, the fundamental of analysis is 43.07Hz. In practice, due to the phase-preserving nature of the phase vocoder, the frequency of any bin can deviate bilaterally, so that DC components are recorded. Given a strongly pitched signal, frequencies in adjacent bins can bunch very closely together, around partials in the source, and the lowest bins may even have negative frequencies.

As a rule, the only reason to use a non power-of-two value for *ifftsize* would be to match the known fundamental frequency of a strongly pitched source. Values with many small factors can be almost as efficient as power-of-two sizes; for example: 384, for a source pitched at around low A=110Hz.

ioverlap -- The distance in samples (“hop size”) between overlapping analysis frames. As a rule, this needs to be at least $\text{ifftsize}/4$, e.g. 256 for the example above. *ioverlap* determines the underlying analysis rate, as $\text{sr}/\text{ioverlap}$. *ioverlap* does not require to be a simple factor of *ifftsize*; for example a value of 160 would be legal. The choice of *ioverlap* may be dictated by the degree of pitch modification applied to the *fsig*, if any. As a rule of thumb, the more extreme the pitch shift, the higher the analysis rate needs to be, and hence the smaller the value for *ioverlap*. A higher analysis rate can also be advantageous with broadband transient sounds, such as drums (where a small analysis window gives less smearing, but more frequency-related errors).

Note that it is possible, and reasonable, to have distinct *fsigs* in an orchestra (even in the same instrument), running at different analysis rates. Interactions between such *fsigs* is currently unsupported, and the *fsig* assignment opcode does not allow copying between *fsigs* with different properties, even if the only difference is in *ioverlap*. However, this is not a closed issue, as it is possible in theory to achieve crude rate conversion (especially with regard to in-memory analysis files) in ways analogous to time-domain techniques.

iwinsize -- The size in samples of the analysis window filter (as set by *iwintype*). This must be at least *ifftsize*, and can usefully be larger. Though other proportions are permitted, it is recommended that *iwinsize* always be an integral multiple of *ifftsize*, e.g. 2048 for the example above. Internally, the analysis window (Hamming, von Hann) is multiplied by a sinc function, so that amplitudes are zero at the boundaries between frames. The larger analysis window size has been found to be especially important for oscillator bank resynthesis (e.g. using *pvsadsyn*), as it has the effect of increasing the frequency resolution of the analysis, and hence the accuracy of the resynthesis. As noted above, *iwinsize* determines the overall latency of the analysis/resynthesis system. In many cases, and especially in the absence of pitch modifications, it will be found that setting *iwinsize*=*ifftsize* works very well, and offers the lowest latency.

iwintype -- The shape of the analysis window. Currently only two choices are implemented:

- 0 = Hamming window
- 1 = von Hann window

Both are also supported by the PVOC-EX file format. The window type is stored as an internal attribute of the fsig, together with the other parameters (see *pvsinfo*). Other types may be implemented later on (e.g. the Kaiser window, also supported by PVOC-EX), though an obvious alternative is to enable windows to be defined via a function table. The main issue here is the constraint of f-tables to power-of-two sizes, so this method does not offer a complete solution. Most users will find the Hamming window meets all normal needs, and can be regarded as the default choice.

iformat -- (optional) The analysis format. Currently only one format is implemented by this opcode:

- 0 = amplitude + frequency

This is the classic phase vocoder format; easy to process, and a natural format for oscillator-bank re-synthesis. It would be very easy (tempting, one might say) to treat an fsig frame not purely as a phase vocoder frame but as a generic additive synthesis frame. It is indeed possible to use an fsig this way, but it is important to bear in mind that the two are not, strictly speaking, directly equivalent.

Other important formats (supported by PVOC-EX) are:

- 1 = amplitude + phase
- 2 = complex (real + imaginary)

iformat is provided in case it proves useful later to add support for these other formats. Formats 0 and 1 are very closely related (as the phase is “wrapped” in both cases - it is a trivial matter to convert from one to the other), but the complex format might warrant a second explicit signal type (a “csig”) specifically for convolution-based processes, and other processes where the full complement of arithmetic operators may be useful.

init -- (optional) Skip reinitialization. This is not currently implemented for any of these opcodes, and it remains to be seen if it is even practical.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

```
ain    in                ; live source
ffin   pvsanal   ain,1024,256,2048,0 ; analyze, using Hamming
ffout  pvsmask   ffin,1,0.75         ; apply eq from f-table
aout   pvsynth   ffout              ; and resynthesize
```

Credits

Author: Richard Dobson

August 2001

New in version 4.13

pvsarp

pvsarp — Arpeggiate the spectral components of a streaming pv signal.

Description

This opcode arpeggiates spectral components, by amplifying one bin and attenuating all the others around it. Used with an LFO it will provide a spectral arpeggiator similar to Trevor Wishart's CDP program specarp.

Syntax

`fsig pvsarp fsigin, kbin, kdepth, kgain`

Performance

fsig -- output pv stream

fsigin -- input pv stream

kbin -- target bin, normalised 0 - 1 (0Hz - Nyquist).

kdepth -- depth of attenuation of surrounding bins

kgain -- gain boost applied to target bin



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Here is an example of the butterbp opcode. It uses the file *pvsarp.csd* [examples/pvsarp.csd]. The example shows a spectral arpeggiator working in the 220.5 - 2425.5 range (sr=44100). The LFO outputs a positive-only signal, so its ftable will be defined in the 0 - 1 range (a hanning window can be used, for instance).

Example 386. Example of the pvsarp opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      ;;-d      RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o pvsarp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 300
nchnls = 1
```

```
instr 1
asig in ; get the signal in
idepth = p4

fsig pvsanal asig, 1024, 256, 1024, 1 ; analyse it
kbin oscili 0.1, 0.5, 1 ; ftable 1 in the 0-1 range
ftps pvsarp fsig, kbin+0.01, idepth, 2 ; arpeggiate it (range 220.5 - 2425.5)
atps pvsynth ftps ; synthesise it

out atps
endin

</CsInstruments>
<CsScore>
; Table #1, a sine wave.
f 1 0 4096 10 1
i 1 0 10 0.5
i 1 + 10 0.9
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Victor Lazzarini
April 2005

New plugin in version 5

April 2005.

pvsbandp

pvsbandp — A band pass filter working in the spectral domain.

Description

Filter the pvoc frames, passing bins whose frequency is within a band, and with linear interpolation for transitional bands.

Syntax

```
fsig pvsbandp fsigin, xlowcut,  
xlowfull, xhighfull, xhighcut[, ktype]
```

Performance

f_{sig} -- output pv stream

f_{sigin} -- input pv stream.

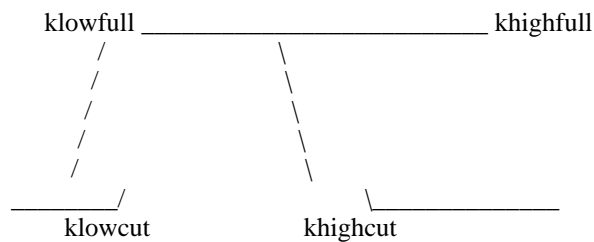
x_{lowcut}, *x_{lowfull}*, *x_{highfull}*, *x_{highcut}* -- define a trapezium shape for the band that is passed. The a-rate versions only apply to the sliding case.

k_{type} -- specifies the shape of the transitional band. If at the default value of zero the shape is as below, with linear transition in amplitude. Other values yield an exponential shape:

$$(1 - \exp(r * \text{type})) / (1 - \exp(\text{type}))$$

This includes a linear dB shape when *k_{type}* is $\log(10)$ or about 2.30.

The opcode performs a band-pass filter with a spectral envelope shaped like



Examples

Example 387. Example

```
asig in ; get the signal in  
fsig pvsanal asig, 1024, 256, 1024, 1 ; analyse it  
ftps pvsbandp fsig, 200, 400, 1000, 2000 ; filter 300-1500Hz  
atps pvsynth ftps ; synthesise it
```

Credits

Author: John ffitch
December 2007

pvsbandr

pvsbandr — A band reject filter working in the spectral domain.

Description

Filter the pvoc frames, rejecting bins whose frequency is within a band, and with linear interpolation for transitional bands.

Syntax

```
fsig pvsbandr fsigin, xlowcut,  
             xlowfull, xhighfull, xhighcut[, ktype]
```

Performance

fsig -- output pv stream

fsigin -- input pv stream.

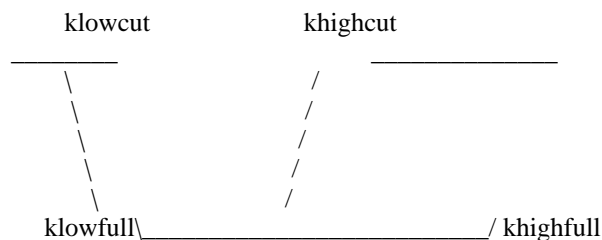
xlowcut, *xlowfull*, *xhighfull*, *xhighcut* -- define a trapezium shape for the band that is rejected. The a-rate versions only apply to the sliding case.

ktype -- specifies the shape of the transitional band. If at the default value of zero the shape is as below, with linear transition in amplitude. Other values give an exponential curve

$$(1 - \exp(r * \text{type})) / (1 - \exp(\text{type}))$$

This includes a linear dB shape when *ktype* is $\log(10)$ or about 2.30.

The opcode performs a band-reject filter with a spectral envelope shaped like



Examples

Example 388. Example

```
asig in                                ; get the signal in  
fsig pvsanal asig, 1024, 256, 1024, 1 ; analyse it  
ftps pvsbandr fsig, 200, 400, 1000, 2000 ; filter 300-1500Hz  
atps pvsynth ftps                      ; synthesise it
```

Credits

Author: John ffitch
December 2007

pvsbin

pvsbin — Obtain the amp and freq values off a PVS signal bin.

Description

Obtain the amp and freq values off a PVS signal bin as k-rate variables.

Syntax

```
kamp, kfr pvsbin fsig, kbin
```

Performance

kamp -- bin amplitude

kfr -- bin frequency

fsig -- an input pv stream

kbin -- bin number

Examples

Here is an example of the pvsbin opcode. It uses the file *pvsbin.csd* [examples/pvsbin.csd]. This example uses realtime input, but you can also use it for soundfile input.

Example 389. Example of the pvsbin opcode

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o pvsbin.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 1
ifftsize = 1024
iwtype = 1      /* cleaner with hanning window */

;al soundin "input.wav" ;select a soundifle
al inch 1      ;Use realtime input

fsig pvsanal  al, ifftsize, ifftsize/4, ifftsize, iwtype
kamp, kfr pvsbin  fsig, 10
adm oscil      kamp, kfr, 1

      out      adm
endin

</CsInstruments>
<CsScore>
```

```
i 1 0 30  
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

Credits

Author: Victor Lazzarini
August 2006

pvsblur

pvsblur — Average the amp/freq time functions of each analysis channel for a specified time.

Description

Average the amp/freq time functions of each analysis channel for a specified time (truncated to number of frames). As a side-effect the input pvoc stream will be delayed by that amount.

Syntax

```
fsig pvsblur fsigin, kblurtime, imaxdel
```

Performance

fsig -- output pv stream

fsigin -- input pv stream.

kblurtime -- time in secs during which windows will be averaged .

imaxdel -- maximum delay time, used for allocating memory used in the averaging operation.

This opcode will blur a pvstream by smoothing the amplitude and frequency time functions (a type of low-pass filtering); the amount of blur will depend on the length of the averaging period, larger blurtimes will result in a more pronounced effect.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 390. Example

```
asig  in                                ; get the signal in
fsig  pvsanal  asig, 1024, 256, 1024, 1 ; analyse it
ftps  pvsblur  fsig, 0.2, 0.2           ; blur it for 200 ms
atps  pvsynth  ftps                     ; synthesise it
```

Credits

Author: Victor Lazzarini;
November 2004

New plugin in version 5

November 2004.

pvsbuffer

pvsbuffer — This opcode creates and writes to a circular buffer for streaming PV signals.

Description

This opcode sets up and writes to a circular buffer of length *ilen* (secs), giving a handle for the buffer and a time pointer, which holds the current write position (also in seconds). It can be used with one or more pvsbufread opcodes. Writing is circular, wrapping around at the end of the buffer.

Syntax

```
ihandle, ktime pvsbuffer fsig, ilen
```

Initialisation

Initialisation

ihandle -- handle identifying this particular buffer, which should be passed to a reader opcode.

ilen -- buffer length in seconds.

fsig -- an input pv stream

ktime -- the current time of writing in the buffer

Examples

Here is an example of the pvsbuffer opcode.

Example 391. Example of the pvsbuffer opcode

With this opcode and pvsbufread, it is possible to, among other things: 1) time-stretch/compress a fsig stream, by reading it at different rates 2) delay a fsig or portions of it. 3) 'brassage' two or more fsigs by switching buffers, since the reading handles are k-rate. Note that, when using k-rate handles, it is important to initialise the k-rate variable to a given handle (so that the fsig initialisation can take place) and it is only possible to switch handles between compatible fsig buffers (with the same fftsize and overlap), eg.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
fsig1  pvsanal  asig1,1024,256,1024,1
fsig2  pvsanal  asig2,1024,256,1024,1

ibuf1,kt1  pvsbuffer  fsig1, 10  ; 10-sec buf with fsig1
ibuf2,kt2  pvsbuffer  fsig2, 7   ; 7-sec buf with fsig2

khan init ibuf1      ; initialise handle to buf1

if ktrig > 0 then    ; switch buffers according to trigger
khan = ibuf2
else
khan = ibuf1
endif
```

```
fsb pvsbufread kt1, khan ; read buffer
```

Credits

Author: Victor Lazzarini
July 2007

pvsbufread

pvsbufread — This opcode creates and writes to a circular buffer for streaming PV signals.

Description

This opcode sets up and writes to a circular buffer of length *ilen* (secs), giving a handle for the buffer and a time pointer, which holds the current write position (also in seconds). It can be used with one or more pvsbufread opcodes. Writing is circular, wrapping around at the end of the buffer.

Syntax

```
fsig pvsbufread ktime, khandle[, ilo, ihi]
```

Initialisation

Initialisation

ilo, ihi -- set the lowest and highest freqs to be read from the buffer (defaults to 0, Nyquist).

fsig -- output pv stream

ktime -- time position of reading pointer (in secs).

khandle -- handle identifying the buffer to be read. When using k-rate handles, it is important to initialise the k-rate variable to a given existing handle. When changing buffers, fsig buffers need to be compatible (same fsig format).

Examples

Here is an example of the pvsbufread opcode.

Example 392. Example of the pvsbufread opcode

With this opcode and pvsbuffer, it is possible to, among other things: 1) time-stretch/compress a fsig stream, by reading it at different rates 2) delay a fsig or portions of it. 3) 'brassage' two or more fsigs by switching buffers, since the reading handles are k-rate. Note that, when using k-rate handles, it is important to initialise the k-rate variable to a given handle (so that the fsig initialisation can take place) and it is only possible to switch handles between compatible fsig buffers (with the same fft-size and overlap), eg.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
fsig1  pvsanal  asig1,1024,256,1024,1
fsig2  pvsanal  asig2,1024,256,1024,1

ibuf1,kt1  pvsbuffer  fsig1, 10  ; 10-sec buf with fsig1
ibuf2,kt2  pvsbuffer  fsig2,  7  ;  7-sec buf with fsig2

khan init ibuf1      ; initialise handle to buf1

if ktrig > 0 then    ; switch buffers according to trigger
khan = ibuf2
else
```

```
khan = ibuf1  
endif  
  
fsb pvsbufread kt1, khan ; read buffer
```

Credits

Author: Victor Lazzarini
July 2007

pvscale

pvscale — Scale the frequency components of a pv stream.

Description

Scale the frequency components of a pv stream, resulting in pitch shift. Output amplitudes can be optionally modified in order to attempt formant preservation.

Syntax

```
fsig pvscale fsigin, kscal[, ikeepform, igain]]
```

Performance

f_{sig} -- output pv stream

f_{sigin} -- input pv stream

kscal -- scaling ratio.

ikeepform -- attempt to keep input signal -- -- formants; 0: do not keep formants; 1: keep formants by imposing original amps; 2: keep formants by filtering using the original spec envelope (defaults to 0).

igain -- amplitude scaling (defaults to 1).

The quality of the pitch shift will be improved with the use of a Hanning window in the pvoc analysis. Formant preservation is only successful with strong-formant sounds, such as voices and certain instrumental sounds, but also can be used for interesting transformation effects.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 393. Example

```
asig in                                ; get the signal in
fsig pvsanal asig, 1024, 256, 1024, 1 ; analyse it
ftps pvscale fsig, 1.5, 1, 2          ; transpose it keeping formants
atps pvsynth ftps                    ; synthesise it

adp delayr .1                          ; delay original signal
adel deltapn 1024                      ; by 1024 samples
      delayw asig

      out atps+adel                  ; add transposed and original
```

The example above shows a vocal harmoniser. The delay is necessary to time-align the signals, as

the analysis-synthesis process will imply a delay of 1024 samples between the analysis input and the synthesis output.

Credits

Author: Victor Lazzarini;
November 2004

New plugin in version 5

November 2004.

pvscent

`pvscent` — Calculate the spectral centroid of a signal.

Description

Calculate the spectral centroids of a signal from its discrete Fourier transform.

Syntax

```
kcent pvscent fsig
```

Performance

kcent -- the spectral centroid

fsig -- an input pv stream

Examples

Example 394. Example

```
ifftsize = 1024
iwtype = 1      /* cleaner with hanning window */

a1  soundin "input.wav"

fsig pvsanal    a1, ifftsize, ifftsize/4, ifftsize, iwtype
kcen pvscent    fsig
adm  oscil      32000, kcen, 1

      out      adm
```

Credits

Author: John ffitch;
March 2005

New plugin in version 5

March 2005.

pvcross

pvcross — Performs cross-synthesis between two source fsigs.

Description

Performs cross-synthesis between two source fsigs.

Syntax

fsig **pvcross** fsrc, fdest, kamp1, kamp2

Performance

The operation of this opcode is identical to that of *pvcross* (q.v.), except in using *fsigs* rather than analysis files, and the absence of spectral envelope preservation. The amplitudes from *fsrc* and *fdest* (using scale factors *kamp1* for *fsrc* and *kamp2* for *fdest*) are applied to the frequencies of *fsrc*. *kamp1* and *kamp2* must not exceed the range 0 to 1.

With this opcode, cross-synthesis can be performed on real-time audio input, by using *pvsanal* to generate *fsrc* and *fdest*. These must have the same format.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

```
kcross  linseg      0,p3/3,0,p3/3,1,p3/3,1 ; progressive cross-synthesis
fcross  pvcross     fsig1,fsig2,1-kcross,kcross
across  pvsynth     fcross
```

Credits

Author: Richard Dobson
August 2001

November 2003. Thanks to Kanata Motohashi, fixed the link to the *pvcross* opcode.

New in version 4.13

pvsdemix

pvsdemix — Spectral azimuth-based de-mixing of stereo sources.

Description

Spectral azimuth-based de-mixing of stereo sources, with a reverse-panning result. This opcode implements the Azimuth Discrimination and Resynthesis (ADResS) algorithm, developed by Dan Barry (Barry et Al. "Sound Source Separation Azimuth Discrimination and Resynthesis". DAFx'04, Univ. of Napoli). The source separation, or de-mixing, is controlled by two parameters: an azimuth position (*kpos*) and a subspace width (*kwidth*). The first one is used to locate the spectral peaks of individual sources on a stereo mix, whereas the second widens the 'search space', including/excluding the peaks around *kpos*. These two parameters can be used interactively to extract source sounds from a stereo mix. The algorithm is particularly successful with studio recordings where individual instruments occupy individual panning positions; it is, in fact, a reverse-panning algorithm.



Warning

It is unsafe to use the same *f*-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Syntax

```
fsig pvsdemix fleft, fright, kpos, kwidth, ipoints
```

Performance

fsig -- output pv stream

fleft -- left channel input pv stream.

fright -- right channel pv stream.

kpos -- the azimuth target centre position, which will be de-mixed, from left to right ($-1 \leq kpos \leq 1$). This is the reverse pan-pot control.

kwidth -- the azimuth subspace width, which will determine the number of points around *kpos* which will be used in the de-mixing process. ($1 \leq kwidth \leq ipoints$)

ipoints -- total number of discrete points, which will divide each pan side of the stereo image. This ultimately affects the resolution of the process.

Examples

The example below takes a stereo input and passes through a de-mixing process revealing a source located at *ipos* +/- *iwidth* points. These parameters can be controlled in realtime (e.g. using FLTK widgets or MIDI) for an interactive search of sound sources.

Example 395. Example

```
ifftsize = 1024
iwidth = 1      /* cleaner with hanning window */
ipos = -0.8    /* to the left of the stereo image */
```

```
iwidth = 20 /* use peaks of 20 points around it */  
al,ar  soundin "sinput.wav"  
  
flc  pvsanal    al, ifftsize, ifftsize/4, ifftsize, iwtype  
frc  pvsanal    ar, ifftsize, ifftsize/4, ifftsize, iwtype  
fdm  pvsdemix   flc, frc, kpos, kwidth, 100  
adm  pvsynth    fdm  
  
      outs      adm,adm
```

Credits

Author: Victor Lazzarini;
January 2005

New plugin in version 5

January 2005.

pvdiskin

pvdiskin — Read a selected channel from a PVOC-EX analysis file.

Description

Create an fsigr stream by reading a selected channel from a PVOC-EX analysis file, with frame interpolation.

Syntax

```
fsigr pvdiskin Sfname, ktscal, kgain[, ioffset, ichan]
```

Initialization

Sfname -- Name of the analysis file. This must have the .pvx file extension.

A multi-channel PVOC-EX file can be generated using the extended *pvanal* utility.

ichan -- (optional) The channel to read (counting from 1). Default is 1.

ioff -- start offset from beginning of file (secs) (default: 0) .

Performance

ktscal -- time scale, ie. the read pointer speed (1 is normal speed, negative is backwards, $0 < ktscal < 1$ is slower and $ktscal > 1$ is faster)

kgain -- gain scaling.

Examples

```
fsigr pvdiskin "test.pvx", 1, 1 ; read PVOC-EX file with tscale and gain = 1  
aout pvsynth fsigr ; resynthesise it
```

Credits

Author: Victor Lazzarini
May 2007
New in Csound 5.06

pvsdisp

pvsdisp — Displays a PVS signal as an amplitude vs. freq graph.

Description

This opcode will display a PVS signal fsig. Uses X11 or FLTK windows if enabled, else (or if -g flag is set) displays are approximated in ASCII characters.

Syntax

```
pvsdisp fsig[, ibins, iwtflg]
```

Initialization

iprd -- the period of pvsdisp in seconds.

ibins (optional, default=all bins) -- optionally, display only ibins bins.

iwtflg (optional, default=0) -- wait flag. If non-zero, each pvsdisp is held until released by the user. The default value is 0 (no wait).

Performance

pvsdisp -- displays the PVS signal frame-by-frame.

Examples

Here is an example of the pvsdisp opcode. It uses the file *pvsdisp.csd* [examples/pvsdisp.csd]. This example uses realtime input, but you can also use it for soundfile input.

Example 396. Example of the pvsdisp opcode

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o pvsdisp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 1
asig inch 1
;al soundin "input.wav" ;select a soundfile
fsig pvsanal asig, 1024,256, 1024, 1
pvsdisp fsig

endin

</CsInstruments>
<CsScore>
```

```
i 1 0 30  
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

dispfft, print

Credits

Author: V Lazzarini, 2006

pvsfilter

pvsfilter — Multiply amplitudes of a pvoc stream by those of a second pvoc stream, with dynamic scaling.

Description

Multiply amplitudes of a pvoc stream by those of a second pvoc stream, with dynamic scaling.

Syntax

```
fsig pvsfilter fsigin, fsigfil, kdepth[, igain]
```

Performance

fsig -- output pv stream

fsigin -- input pv stream.

fsigfil -- filtering pvoc stream.

kdepth -- controls the depth of filtering of fsigin by fsigfil .

igain -- amplitude scaling (optional, defaults to 1).

Here the input pvoc stream amplitudes are modified by the filtering stream, keeping its frequencies intact. As usual, both signals have to be in the same format.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 397. Example

```
kfreq expon 500, p3, 4000      ; 3-octave sweep
kdepth linseg 1, p3/2, 0.5, p3/2, 1 ; varying filter depth

asig in                        ; input
afil oscili 1, kfreq, 1       ; filter t-domain signal

fim pvsanal asig,1024,256,1024,0 ; pvoc analysis
fil pvsanal afil,1024,256,1024,0
fou pvsfilter fim, fil, kdepth ; filter signal
aout pvsynth fou              ; pvoc synthesis
```

In the example above the filter curve will depend on the spectral envelope of afil; in the simple case of a sinusoid, it will be equivalent to a narrowband band-pass filter.

Credits

Author: Victor Lazzarini;
November 2004

New plugin in version 5

November 2004.

pvsfread

pvsfread — Read a selected channel from a PVOC-EX analysis file.

Description

Create an fsig stream by reading a selected channel from a PVOC-EX analysis file loaded into memory, with frame interpolation. Only format 0 files (amplitude+frequency) are currently supported. The operation of this opcode mirrors that of pvoc, but outputs an fsig instead of a resynthesized signal.

Syntax

```
fsig pvsfread ktimpt, ifn [, ichan]
```

Initialization

ifn -- Name of the analysis file. This must have the .pvx file extension.

A multi-channel PVOC-EX file can be generated using the extended *pvanal* utility.

ichan -- (optional) The channel to read (counting from 0). Default is 0.

Performance

ktimpt -- Time pointer into analysis file, in seconds. See the description of the same parameter of *pvoc* for usage.

Note that analysis files can be very large, especially if multi-channel. Reading such files into memory will very likely incur breaks in the audio during real-time performance. As the file is read only once, and is then available to all other interested opcodes, it can be expedient to arrange for a dedicated instrument to preload all such analysis files at startup.

Examples

```
idur  filelen  "test.pvx"          ; find dur of (stereo) analysis file
kpos  line    0,p3,idur           ; to ensure we process whole file
fsigr  pvsfread kpos,"test.pvx",1 ; create fsig from R channel
```

(NB: as this example shows, the filelen opcode has been extended to accept both old and new analysis file formats).

Credits

Author: Richard Dobson
August 2001

New in version 4.13

pvsfreeze

pvsfreeze — Freeze the amplitude and frequency time functions of a pv stream according to a control-rate trigger.

Description

This opcode 'freezes' the evolution of pvs stream by locking into steady amplitude and/or frequency values for each bin. The freezing is controlled, independently for amplitudes and frequencies, by a control-rate trigger, which switches the freezing 'on' if equal to or above 1 and 'off' if below 1.

Syntax

```
fsig pvsfreeze fsigin, kfreeza, kfreezf
```

Performance

fsig -- output pv stream

fsigin -- input pv stream.

kfreeza -- freezing switch for amplitudes. Freezing is on if above or equal to 1 and off if below 1.

kfcf -- freezing switch for frequencies. Freezing is on if above or equal to 1 and off if below 1.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 398. Example

```
asig in                                ; input
ktrig oscil 1.5, 0.25, 1                ; trigger
fim pvsanal asigl,1024,256,1024,0 ; pvoc analysis
fou pvsfreeze fim, abs(ktrig), abs(ktrig) ; regular 'freeze' of spectra
aout pvsynth fou                        ; pvoc synthesis
```

In the example above the input signal will be regularly 'frozen' for a short while, as the trigger rises above 1 about every two seconds.

Credits

Author: Victor Lazzarini;
May 2006

New plugin in version 5

May 2006.

pvsftr

pvsftr — Reads amplitude and/or frequency data from function tables.

Description

Reads amplitude and/or frequency data from function tables.

Syntax

```
pvsftr fsrc, ifna [, ifnf]
```

Initialization

ifna -- A table, at least inbins in size, that stores amplitude data. Ignored if ifna = 0

ifnf (optional) -- A table, at least inbins in size, that stores frequency data. Ignored if ifnf = 0

Performance

fsrc -- a PVOC-EX formatted source.

Enables the contents of *fsrc* to be exchanged with function tables for custom processing. Except when the frame overlap equals *ksmps* (which will generally not be the case), the frame data is not updated each control period. The data in *ifna*, *ifnf* should only be processed when *kflag* is set to 1. To process only frequency data, set *ifna* to zero.

As the function tables are required only to store data from *fsrc*, there is no advantage in defining them in the score, and they should generally be created in the instrument, using *ftgen*.

By exporting amplitude data, say, from one fsig and importing it into another, basic cross-synthesis (as in *pvcross*) can be performed, with the option to modify the data beforehand using the table manipulation opodes.

Note that the format data in the source fsig is not written to the tables. This therefore offers a means of transferring amplitude and frequency data between non-identical fsigs. Used this way, these opcodes become potentially pathological, and can be relied upon to produce unexpected results. In such cases, resynthesis using *pvsadsyn* would almost certainly be required.

To perform a straight copy from one fsig to another one of identical format, the conventional assignment syntax can be used:

```
fsig1 = fsig2
```

It is not necessary to use function tables in this case.

Examples

```
ifn      ftgen      0,0,inbins,10,1      ; make ftable
kflag    pvsftw     fsrc,ifn             ; export amps to table,
kamp     init      0
if       kflag==0   kgoto contin         ; only proc when frame is ready
; kill   lowest bins, for obvious effect
         tablew     kamp,1,ifn
```

```
        tablew      kamp,2,ifn
        tablew      kamp,3,ifn
        tablew      kamp,4,ifn
; read modified data back to fsrc
        pvsftr      fsrc,ifn
contin:
; and resynth
aout    pvsynth     fsrc
```

See Also

pvsftw

Credits

Author: Richard Dobson
August 2001

New in version 4.13

pvsftw

pvsftw — Writes amplitude and/or frequency data to function tables.

Description

Writes amplitude and/or frequency data to function tables.

Syntax

```
kflag pvsftw fsrc, ifna [, ifnf]
```

Initialization

ifna -- A table, at least inbins in size, that stores amplitude data. Ignored if ifna = 0

ifnf -- A table, at least inbins in size, that stores frequency data. Ignored if ifnf = 0

Performance

kflag -- A flag that has the value of 1 when new data is available, 0 otherwise.

fsrc -- a PVOC-EX formatted source.

Enables the contents of *fsrc* to be exchanged with function tables, for custom processing. Except when the frame overlap equals *ksmps* (which will generally not be the case), the frame data is not updated each control period. The data in *ifna*, *ifnf* should only be processed when *kflag* is set to 1. To process only frequency data, set *ifna* to zero.

As the functions tables are required only to store data from *fsrc*, there is no advantage in defining them in the score. They should generally be created in the instrument using *ftgen*.

By exporting amplitude data, say, from one fsig and importing it into another, basic cross-synthesis (as in *pvsfcross*) can be performed, with the option to modify the data beforehand using the table manipulation opodes.

Note that the format data in the source fsig is not written to the tables. This therefore offers a means of transferring amplitude and frequency data between non-identical fsigs. Used this way, these opcodes become potentially pathological, and can be relied upon to produce unexpected results. In such cases, resynthesis using *pvsadsyn* would almost certainly be required.

To perform a straight copy from one fsig to another one of identical format, the conventional assignment syntax can be used:

```
fsig1 = fsig2
```

It is not necessary to use function tables in this case.

Examples

```
ifn      ftgen      0,0,inbins,10,1      ; make ftable
kflag    pvsftw     fsrc,ifn             ; export amps to table,
kamp     init       0
if       kflag==0   kgoto contin        ; only proc when frame is ready
```

```
; kill lowest bins, for obvious effect
      tablew      kamp,1,ifn
      tablew      kamp,2,ifn
      tablew      kamp,3,ifn
      tablew      kamp,4,ifn
; read modified data back to fsrc
      pvsftr      fsrc,ifn
contin:
; and resynth
aout      pvsynth      fsrc
```

See Also

pvsftr

Credits

Author: Richard Dobson
August 2001

New in version 4.13

pvsfwrite

pvsfwrite — Write a fsig to a PVOCEX file.

Description

This opcode writes a fsig to a PVOCEX file (which in turn can be read by pvsfread or other programs that support PVOCEX file input).

Syntax

```
pvsfwrite fsig, ifile
```

Initialisation

fsig -- fsig input data. *ifile* -- filename (a string in double-quotes) .

Examples

Here is an example of the pvsfwrite opcode. It uses the file *pvsfwrite.csd* [examples/pvsfwrite.csd]. This example uses realtime audio input.

Example 399. Example of the pvsfwrite opcode

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o pvsfwrite.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

;By Victor Lazzarini 2008

instr 1
asig oscili 10000, 440, 1
fss pvsanal asig, 1024,256,1024,0
pvsfwrite fss, "mypvs.pvx"
ase pvsynth fss
      out ase
endin

instr 2 ; must be called after instr 1 finishes
ktim timeinsts
fss pvsfread ktim, "mypvs.pvx"
asig pvsynth fss
      out asig
endin

</CsInstruments>
<CsScore>
f1 0 16384 10 1
i1 0 1
i2 1 1
e
</CsScore>
```

`</CsoundSynthesizer>`

Credits

Author: Victor Lazzarini;
November 2004

New plugin in version 5
November 2004.

pvshift

pvshift — Shift the frequency components of a pv stream, stretching/compressing its spectrum.

Description

Shift the frequency components of a pv stream, stretching/compressing its spectrum.

Syntax

```
fsig pvshift fsigin, kshift, klowest[, ikeepform, igain]
```

Performance

fsig -- output pv stream

fsigin -- input pv stream

kshift -- shift amount (in Hz, positive or negative).

klowest -- lowest frequency to be shifted.

ikeepform -- attempt to keep input signal formants; 0: do not keep formants; 1: keep formants by imposing original amps; 2: keep formants by filtering using the original spec envelope (defaults to 0).

igain -- amplitude scaling (defaults to 1).

This opcode will shift the components of a pv stream, from a certain frequency upwards, up or down a fixed amount (in Hz). It can be used to transform a harmonic spectrum into an inharmonic one. The ikeepform flag can be used to try and preserve formants for possibly interesting and unusual spectral modifications.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 400. Example

```
asig in                                ; get the signal in
fsig pvsanal   asig, 1024, 256, 1024, 1 ; analyse it
ftps pvshift   fsig, 100, 0             ; add 100 Hz to each component
atps pvsynth   ftps                    ; synthesise it
```

Depending on the input, this will transform a pitched sound into an inharmonic, bell-like sound.

Credits

Author: Victor Lazzarini
November 2004

New plugin in version 5

Nivember 2004.

pvsifd

pvsifd — Instantaneous Frequency Distribution, magnitude and phase analysis.

Description

The pvsifd opcode takes an input a-rate signal and performs an Instantaneous Frequency, magnitude and phase analysis, using the STFT and pvsifd (Instantaneous Frequency Distribution), as described in Lazzarini et al, "Time-stretching using the Instantaneous Frequency Distribution and Partial Tracking", Proc.of ICMC05, Barcelona. It generates two PV streaming signals, one containing the amplitudes and frequencies (a similar output to pvsanal) and another containing amplitudes and unwrapped phases.

Syntax

```
ffr,fphs pvsifd ain, ifftsize, ihopsize, iwintype[,iscal]
```

Performance

ffr -- output pv stream in AMP_FREQ format

fphs -- output pv stream in AMP_PHASE format

ifftsize -- FFT analysis size, must be power-of-two and integer multiple of the hopsize.

ihopsize -- hopsize in samples

iwintype -- window type (0: Hamming, 1: Hanning)

iscal -- amplitude scaling (defaults to 1).



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 401. Example

```
ain inch 1 ; input signal
fs1,fsi2 pvsifd ain,2048,512,1 ; pvsifd analysis
fst partials fs1,fsi2,.003,1,3,500 ; partial tracking
aout resyn fst, 1, 1.5, 500, 1 ; resynthesis (up a 5th)
out aout
```

The example above shows the pvsifd analysis feeding into partial tracking and cubic-phase additive resynthesis with pitch shifting.

Credits

Author: Victor Lazzarini;
June 2005

New plugin in version 5

November 2004.

pvsinfo

pvsinfo — Get information from a PVOC-EX formatted source.

Description

Get format information about fsrc, whether created by an opcode such as pvsanal, or obtained from a PVOC-EX file by pvsfread. This information is available at init time, and can be used to set parameters for other pvs opcodes, and in particular for creating function tables (e.g. for pvsftw), or setting the number of oscillators for pvsadsyn.

Syntax

```
ioverlap, inumbins, iwinsize, iformat pvsinfo fsrc
```

Initialization

ioverlap -- The stream overlap size.

inumbins -- The number of analysis bins (amplitude+frequency) in fsrc. The underlying FFT size is calculated as (inumbins -1) * 2.

iwinsize -- The analysis window size. May be larger than the FFT size.

iformat -- The analysis frame format. If fsrc is created by an opcode, iformat will always be 0, signifying amplitude+frequency. If fsrc is defined from a PVOC-EX file, iformat may also have the value 1 or 2 (amplitude+phase, complex).

Examples

```
fim      pvsfread  "test.pvx"      ; import pvocex file
iovl,inb,iws,ifmt  pvsinfo  fim      ; get inumbins info
ifn      ftgen     0,0,inb,10,1    ; and create f-table
```

Credits

Author: Richard Dobson
August 2001

New in version 4.13

pvsinit

pvsinit — Initialise a spectral (f) variable to zero.

Description

Fermorms the equavent to an init operation on an f-variable.

Syntax

```
fsig pvsinit isize[,iolap,iwinsize,iwintype, iformat]
```

Performance

f_{sig} -- output pv stream set to zero.

isize -- size of the DFT frame.

iolap -- size of the analysis overlap, defaults to isize/4.

iwinsize -- size of the analysis window, defaults to isize.

iwintype -- type of analysis window, defaults to 1, Hanning.

iformat -- pvsdata format, defaults to 0:PVS_AMP_FREQ.

Examples

Example 402. Example

```
fsig pvsinit 1024
```

Credits

Author: Victor Lazzarini;
November 2004

New plugin in version 5

November 2004.

pvsin

pvsin — Retrieve an fsig from the input software bus; a pvs equivalent to chani.

Description

This opcode retrieves an f-sig from the pvs in software bus, which can be used to get data from an external source, using the Csound 5 API. A channel is created if not already existing. The fsig channel is in that case initialised with the given parameters. It is important to note that the pvs input and output (pvsout opcode) busses are independent and data is not shared between them.

Syntax

```
fsig pvsin kchan[,isize,iolap,iwinsize,iwintype,iformat]
```

Initialisation

isize -- initial DFT size, defaults to 1024.

iolap -- size of overlap, defaults to isize/4.

isize -- size of analysis window, defaults to isize.

isize -- type of window, defaults to Hanning (1) (see pvsanal)

iformat -- data format, defaults 0 (PVS_AMP_FREQ). Other possible values are 1 (PVS_AMP_PHASE), 2 (PVS_COMPLEX) or 3 (PVS_TRACKS).

Performance

fsig -- output fsig.

kchan -- channel number. If non-existent, a channel will be created.

Examples

Example 403. Example

```
fsig pvsin 0 ; get data from pvs in bus channel 0
```

Credits

Author: Victor Lazzarini;
August 2006

pvsmaska

pvsmaska — Modify amplitudes using a function table, with dynamic scaling.

Description

Modify amplitudes of fsrc using function table, with dynamic scaling.

Syntax

```
fsig pvsmaska fsrc, ifn, kdepth
```

Initialization

ifn -- The f-table to use. Given fsrc has N analysis bins, table ifn must be of size N or larger. The table need not be normalized, but values should lie within the range 0 to 1. It can be supplied from the score in the usual way, or from within the orchestra by using *pvsinfo* to find the size of fsrc, (returned by pvsinfo in inbins), which can then be passed to ftgen to create the f-table.

Performance

kdepth -- Controls the degree of modification applied to fsrc, using simple linear scaling. 0 leaves amplitudes unchanged, 1 applies the full profile of ifn.

Note that power-of-two FFT sizes are particularly convenient when using table-based processing, as the number of analysis bins (inbins) is then a power-of-two plus one, for which an exactly matching f-table can be created. In this case it is important that the f-table be created with a size of inbins, rather than as a power of two, as the latter will copy the first table value to the guard point, which is inappropriate for this opcode.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 404. Example (using score-supplied f-table, assuming fsig fftsize = 1024)

```
; score f-table using cubic spline to define shaped peaks
f1 0 513 8 0 2 1 3 0 4 1 6 0 10 1 12 0 16 1 32 0 1 0 436 0

asig buzz      20000,199,50,1      ; pulsewave source
fsig pvsanal   asig,1024,256,1024,0 ; create fsig
kmod linseg    0,p3/2,1,p3/2,0     ; simple control sig

fsig2 pvsmaska fsig,2,kmod          ; apply weird eq to fsig
aout pvsynth   fsig2                ; resynthesize,
dispfft aout,0.1,1024              ; and view the effect
```


Credits

Author: Richard Dobson
August 2001

New in version 4.13

pvsmix

pvsmix — Mix 'seamlessly' two pv signals.

Description

Mix 'seamlessly' two pv signals. This opcode combines the most prominent components of two pvoc streams into a single mixed stream.

Syntax

```
fsig pvsmix fsigin1, fsigin2
```

Performance

fsig -- output pv stream

fsigin1 -- input pv stream.

fsigin2 -- input pv stream, which must have same format as fsigin1.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 405. Example

```
fsg1 pvsanal asig1,1024,256,1024,0 ; pvoc analysis  
fsg2 pvsanal asig2,1024,256,1024,0  
fsgout pvsmix fsg1, fsg2           ; mix signals  
aout pvsynth fsgout                ; pvoc synthesis
```

Depending on the input, this will transform a pitched sound into an inharmonic, bell-like sound.

Credits

Author: Victor Lazzarini
November 2004

New plugin in version 5

November 2004.

pvsmorph

pvsmorph — Performs morphing (or interpolation) between two source fsigs.

Description

Performs morphing (or interpolation) between two source fsigs.

Syntax

```
fsig pvsmorph fsig1, fsig2, kampint, kfrqint
```

Performance

The operation of this opcode is similar to that of *pvinterp* (q.v.), except in using *fsigs* rather than analysis files, and the absence of spectral envelope preservation. The amplitudes and frequencies of *fsig1* are interpolated with those of *fsig2*, depending on the values of *kampint* and *kfrqint*, respectively. These range between 0 and 1, where 0 means *fsig1* and 1, *fsig2*. Anything in between will interpolate amps and/or freqs of the two fsigs.

With this opcode, morphing can be performed on real-time audio input, by using *pvsanal* to generate *fsig1* and *fsig2*. These must have the same format.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

```
km linseg 0,p3/3,0,p3/3,1,p3/3,1 ; progressive morphing
fmo pvsmorph fsig1,fsig2,km,km
asig pvsynth fmo
```

Credits

Author: Victor Lazzarini
April 2007
New in Csound 5.06

pvsMOOTH

pvsMOOTH — Smooth the amplitude and frequency time functions of a pv stream using parallel 1st order lowpass IIR filters with time-varying cutoff frequency.

Description

Smooth the amplitude and frequency time functions of a pv stream using a 1st order lowpass IIR with time-varying cutoff frequency. This opcode uses the same filter as the 'tone' opcode, but this time acting separately on the amplitude and frequency time functions that make up a pv stream. The cutoff frequency parameter runs at the control-rate, but unlike tone and tonek, it is not specified in Hz, but as fractions of 1/2 frame-rate (actually the pv stream sampling rate), which is easier to understand. This means that the highest cutoff frequency is 1 and the lowest 0; the lower the frequency the smoother the functions and more pronounced the effect will be. This opcode produces effects that are more or less similar to pvsblur, but with two important differences: 1. smoothing of amplitudes and frequencies use separate sets of filters; and 2. there is no increase in computational cost when higher amounts of 'blurring' (smoothing) are desired.

Syntax

```
fsig pvsMOOTH fsign, kacf, kkcf
```

Performance

f_{sig} -- output pv stream

f_{sign} -- input pv stream.

k_{acf} -- amount of cutoff frequency for amplitude function filtering, between 0 and 1, in fractions of 1/2 frame-rate.

k_{kcf} -- amount of cutoff frequency for frequency function filtering, between 0 and 1, in fractions of 1/2 frame-rate.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 406. Example

```
asigl in ; input
fim pvsanal asigl,1024,256,1024,0 ; pvoc analysis
fou pvsMOOTH fim, 0.01, 0.01 ; smooth with cf at 1% of 1/2 frame-rate (ca 8.6 Hz)
aout pvsynth fou ; pvoc synthesis
```

In the example above the input signal will be smoothed/blurred by pvsMOOTH with a cutoff frequency of 1% of 1/2 frame-rate (which is about 172Hz, so the cf is about 8.6Hz) .

Credits

Author: Victor Lazzarini;
May 2006

New plugin in version 5

May 2006.

pvsout

pvsout — Write a fsig to the pvs output bus.

Description

This opcode writes a fsig to a channel of the pvs output bus. Note that the pvs out bus and the pvs in bus are separate and independent. A new channel is created if non-existent.

Syntax

```
pvsout fsig, kchan
```

Performance

fsig -- fsig input data.

kchan -- pvs out bus channel number.

Examples

Example 407. Example

```
asig in ; input
fsig pvsanal asig, 1024, 256, 1024, 1 ; analysis
pvsout fsig,0 ; write signal to pvs out bus channel 0
```

Credits

Author: Victor Lazzarini;
August 2006

pvsosc

pvsosc — PVS-based oscillator simulator.

Description

Generates periodic signal spectra in AMP-FREQ format, with the option of four wave types:

1. sawtooth-like (harmonic weight $1/n$, where n is partial number)
2. square-like (similar to 1., but only odd partials)
3. pulse (all harmonics with same weight)
4. cosine

Complex waveforms (ie. all types except cosine) contain all harmonics up to the Nyquist. This makes pvsosc an option for generation of band-limited periodic waves. In addition, types can be changed using a k-rate variable.

Syntax

```
fsig pvsosc kamp, kfreq, ktype, isize [,ioverlap] [, iwinsize] [, iwintype] [, iformat]
```

Initialisation

fsig -- output pv stream set to zero.

isize -- size of analysis frame and window.

ioverlap -- (Optional) size of overlap, defaults to *isize*/4.

iwinsize -- (Optional) window size, defaults to *isize*.

iwintype -- (Optional) window type, defaults to Hanning. The choices are currently:

- 0 = Hamming window
- 1 = von Hann window

iformat -- (Optional) data format, defaults to 0 which produces AMP:FREQ data. That is currently the only option.

Performance

kamp -- signal amplitude. Note that the actual signal amplitude can, depending on wave type and frequency, vary slightly above or below this value. Generally the amplitude will tend to exceed *kamp* on higher frequencies (> 1000 Hz) and be reduced on lower ones. Also due to the overlap-add process, when resynthesing with pvsynth, frequency glides will cause the output amplitude to fluctuate above and below *kamp*.

kfreq -- fundamental frequency in Hz.

ktype -- wave type: 1. sawtooth-like, 2.square-like, 3.pulse and any other value for cosine.

Examples

Here is an example of the pvsosc opcode. It uses the file *pvsosc.csd* [examples/pvsosc.csd].

Example 408. Example of the pvsosc opcode

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o pvsosc.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 1
; a band-limited sawtooth-wave oscillator
fsig pvsosc 10000, 440, 1, 1024 ; generate wave spectral signal
asig pvsynth fsig                ; resynthesise it
out asig
endin

instr 2
; a band-limited square-wave oscillator
fsig pvsosc 10000, 440, 2, 1024 ; generate wave spectral signal
asig pvsynth fsig                ; resynthesise it
out asig
endin

instr 3
; a pulse oscillator
fsig pvsosc 10000, 440, 3, 1024 ; generate wave spectral signal
asig pvsynth fsig                ; resynthesise it
out asig
endin

instr 4
; a cosine-wave oscillator
fsig pvsosc 10000, 440, 4, 1024 ; generate wave spectral signal
asig pvsynth fsig                ; resynthesise it
out asig
endin

</CsInstruments>
<CsScore>

i 1 0 1
i 2 2 1
i 3 4 1
i 4 6 1

e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Victor Lazzarini;
August 2006

pvspitch

pvspitch — Track the pitch and amplitude of a PVS signal.

Description

Track the pitch and amplitude of a PVS signal as k-rate variables.

Syntax

```
kfr, kamp pvspitch fsig, kthresh
```

Performance

kamp -- Amplitude of fundamental frequency

kfr -- Fundamental frequency

fsig -- an input pv stream

kthresh -- analysis threshold (between 0 and 1). Higher values will eliminate low-amplitude components from the analysis.

Performance

The pitch detection algorithm implemented by *pvspitch* is based upon J. F. Schouten's hypothesis of the neural processes of the brain used to determine the pitch of a sound after the frequency analysis of the basilar membrane. Except for some further considerations, *pvspitch* essentially seeks out the highest common factor of an incoming sound's spectral peaks to find the pitch that may be attributed to it.

In general, input sounds that exhibit pitch will also exhibit peaks in their spectrum according to where their harmonics lie. There are some the exceptions, however. Some sounds whose spectral representation is continuous can impart a sensation of pitch. Such sounds are explained by the centroid or center of gravity of the spectrum and are beyond the scope of the method of pitch detection implemented by *pvspitch* (Using opcodes like *pvscent* might be more appropriate in these cases).

pvspitch is able (using a previous analysis *fsig* generated by *pvsanal*) to locate the spectral peaks of a signal. The threshold parameter (*kthresh*) is of utmost importance, as adjusting it can introduce weak yet significant harmonics into the calculation of the fundamental. However, bringing *kthresh* too low would allow harmonically unrelated partials into the analysis algorithm and this will compromise the method's accuracy. These initial steps emulate the response of the basilar membrane by identifying physical characteristics of the input sound. The choice of *kthresh* depends on the actual level of the input signal, since its range (from 0 to 1) spans the whole dynamic range of an analysis bin (from -inf to 0dBFS).

It is important to remember that the input to the *pvspitch* opcode is assumed to be characterised by strong partials within its spectrum. If this is not the case, the results outputted by the opcode may not bear any relation to the pitch of the input signal. If a spectral frame with many unrelated partials was analysed, the greatest common factor of these frequency values that allows for adjacent “harmonics” would be chosen. Thus, noisy frames can be characterised by low frequency outputs of *pvspitch*. This fact allows for a primitive type of instrumental transient detection, as the attack portion of some instrumental tones contain inharmonic components. Should the lowest frequency of the analysed melody be known, then all frequencies detected below this threshold are inaccurate readings, due to the presence of unrelated partials.

In order to facilitate efficient testing of the *pvspitch* algorithm, an amplitude value proportional to the one in the observed in the signal frame is also outputted (*kamp*). The results of *pvspitch* can then be employed to drive an oscillator whose pitch can be audibly compared with that of the original signal (In the example below, an oscillator generates a signal which appears a fifth above the detected pitch).

Examples

Here is an example of the *pvspitch* opcode. It uses the file *pvspitch.csd* [examples/pvspitch.csd]. This example uses realtime audio input but can be used for audiofile input as well.

Example 409. Example of the *pvspitch* opcode

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o pvspitch.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 128
nchnls = 1

giwave ftgen 0, 0, 4096, 10, 1, 0.5, 0.333, 0.25, 0.2, 0.1666

instr 1

ifftsize = 1024
iwtype = 1      /* cleaner with hanning window */

a1 inch 1 ;Realtime audio input
;a1 soundin "input.wav" ;Use this line for file input

fsig pvsanal a1, ifftsize, ifftsize/4, ifftsize, iwtype
kfr, kamp pvspitch fsig, 0.01

adm oscil      kamp, kfr * 1.5, giwave ;Generate note a fifth above detected pitch

      out      adm
endin

</CsInstruments>
<CsScore>

i 1 0 30

e

</CsScore>
</CsoundSynthesizer>
```

See also

pvsanal, *pvscent*

Credits

Author: Alan OCinneide
August 2005, added by V Lazzarini, August 2006

Part of the text has been adapted from the Csound Journal winter 2006 issue's article "Introducing PVSPITCH: A pitch tracking opcode for Csound" by Alan OCinneide. The article is available at: www.csounds.com/journal/2006winter/pvspitch.html [http://www.csounds.com/journal/2006winter/pvspitch.html]

pvstencil

pvstencil — Transforms a pvoc stream according to a masking function table.

Description

Transforms a pvoc stream according to a masking function table; if the pvoc stream amplitude falls below the value of the function for a specific pvoc channel, it applies a gain to that channel.

The pvoc stream amplitudes are compared to a masking table, if they fall below the table values, they are scaled by *kgain*. Prior to the operation, table values are scaled by *klevel*, which can be used as masking depth control.

Tables have to be at least $\text{fftsize}/2$ in size; for most GENS it is important to use an extended-guard point (size power-of-two plus one), however this is not necessary with GEN43.

One of the typical uses of *pvstencil* would be in noise reduction. A noise print can be analysed with *pvanal* into a PVOCEX file and loaded in a table with GEN43. This then can be used as the masking table for *pvstencil* and the amount of reduction would be controlled by *kgain*. Skipping post-normalisation will keep the original noise print average amplitudes. This would provide a good starting point for a successful noise reduction (so that *klevel* can be generally set to close to 1).

Other possible transformation effects are possible, such as filtering and 'inverse-masking'.

Syntax

```
fsig pvstencil fsigin, kgain, klevel, iftable
```

Performance

fsig -- output pv stream

fsigin -- input pv stream.

kgain -- 'stencil' gain.

klevel -- masking function level (scales the ftable prior to 'stenciling').

iftable -- masking function table.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 410. Example

```
fsig    pvsanal    asig, 1024, 256, 1024, 1  
fclean pvstencil fsig, 0, 1, 1  
aclean pvsynth fclean
```

Credits

Author: Victor Lazzarini;
November 2004

New plugin in version 5

Nivember 2004.

pvsvoc

pvsvoc — Combine the spectral envelope of one fsig with the excitation (frequencies) of another.

Description

This opcode provides support for cross-synthesis of amplitudes and frequencies. It takes the amplitudes of one input fsig and combines with frequencies from another. It is a spectral version of the well-known channel vocoder.

Syntax

fsig **pvsvoc** famp, fexc, kdepth, kgain

Performance

fsig -- output pv stream

famp -- input pv stream from which the amplitudes will be extracted

fexc -- input pv stream from which the frequencies will be taken

kdepth -- depth of effect, affecting how much of the frequencies will be taken from the second fsig: 0, the output is the famp signal, 1 the output is the famp amplitudes and fexc frequencies.

kgain -- gain boost/attenuation applied to the output.



Warning

It is unsafe to use the same f-variable for both input and output of pvs opcodes. Using the same one might lead to undefined behavior on some opcodes. Use a different one on the left and right sides of the opcode.

Examples

Example 411. Example

```
asig in ; get the signal in
asyn oscili 16000, 150, 1 ; excitation signal

famp pvsanal asig, 1024, 256, 1024, 1 ; analyse in signal
fexc pvsanal asyn, 1024, 256, 1024, 1 ; analyse excitation signal
ftps pvsvoc famp, fexc, 1, 1 ; cross it
atps pvsynth ftps ; synthesise it

out atps
```

The example above shows a typical cross-synthesis operation. The input signal (say a vocal sound) is used for its amplitude spectrum. An oscillator with an arbitrary complex waveform produces the excitation signal, giving the vocal sound its pitch.

Credits

Author: Victor Lazzarini;
April 2005

New plugin in version 5

April 2005.

pvsynth

pvsynth — Resynthesise using a FFT overlap-add.

Description

Resynthesise using a FFT overlap-add.

Syntax

```
ares pvsynth fsrc, [iinit]
```

Performance

ares -- output audio signal

fsrc -- input signal

iinit -- not yet implemented.

Examples

Example 412. Example (using score-supplied f-table, assuming fsig fftsize = 1024)

```
; score f-table using cubic spline to define shaped peaks
f1 0 513 8 0 2 1 3 0 4 1 6 0 10 1 12 0 16 1 32 0 1 0 436 0

asig  buzz      20000,199,50,1      ; pulswave source
fsig  pvsanal  asig,1024,256,1024,0 ; create fsig
kmod  linseg    0,p3/2,1,p3/2,0     ; simple control sig

fsig  pvsmaska fsig,2,kmod          ; apply weird eq to fsig
aout  pvsynth  fsig                 ; resynthesize,
      dispfft  aout,0.1,1024        ; and view the effect
```

This also illustrates that the usual Csound behaviour applies to fsigs; the same name can be used for both input and output.

See Also

pvsadsyn

Credits

Author: Richard Dobson
August 2001

New in version 4.13

February 2004. Thanks to a note from Francisco Vila, updated the example.

pyassign Opcodes

pyassign — Assign the value of the given Csound variable to a Python variable possibly destroying its previous content.

Syntax

```
pyassign "variable", kvalue
```

```
pyassigni "variable", ivalue
```

```
pylassign "variable", kvalue
```

```
pylassigni "variable", ivalue
```

```
pyassignt ktrigger, "variable", kvalue
```

```
pylassignt ktrigger, "variable", kvalue
```

Description

Assign the value of the given Csound variable to a Python variable possibly destroying its previous content. The resulting Python object will be a float.

Credits

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pycall Opcodes

pycall — Invoke the specified Python callable at k-time and i-time (i suffix), passing the given arguments. The call is performed in the global environment, and the result (the returning value) is copied into the Csound output variables specified.

Syntax

kresult	pycall	"callable", karg1, ...
kresult1, kresult2	pycall1	"callable", karg1, ...
kr1, kr2, kr3	pycall2	"callable", karg1, ...
kr1, kr2, kr3, kr4	pycall3	"callable", karg1, ...
kr1, kr2, kr3, kr4, kr5	pycall4	"callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6	pycall5	"callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6, kr7	pycall6	"callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6, kr7, kr8	pycall7	"callable", karg1, ...
	pycall8	"callable", karg1, ...
	pycallt	ktrigger, "callable", karg1, ...
kresult	pycall1t	ktrigger, "callable", karg1, ...
kresult1, kresult2	pycall2t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3	pycall3t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4	pycall4t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4, kr5	pycall5t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6	pycall6t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6, kr7	pycall7t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6, kr7, kr8	pycall8t	ktrigger, "callable", karg1, ...
	pycalli	"callable", karg1, ...
iresult	pycall1i	"callable", iarg1, ...
iresult1, iresult2	pycall2i	"callable", iarg1, ...
ir1, ir2, ir3	pycall3i	"callable", iarg1, ...
ir1, ir2, ir3, ir4	pycall4i	"callable", iarg1, ...
ir1, ir2, ir3, ir4, ir5	pycall5i	"callable", iarg1, ...
ir1, ir2, ir3, ir4, ir5, ir6	pycall6i	"callable", iarg1, ...
ir1, ir2, ir3, ir4, ir5, ir6, ir7	pycall7i	"callable", iarg1, ...
ir1, ir2, ir3, ir4, ir5, ir6, ir7, ir8	pycall8i	"callable", iarg1, ...
pycalln	"callable", nresults, kresult1, ..., kresultn, karg1, ...	
pycallni	"callable", nresults, iresult1, ..., iresultn, iarg1, ...	
	pylcall	"callable", karg1, ...
kresult	pylcall1	"callable", karg1, ...
kresult1, kresult2	pylcall2	"callable", karg1, ...
kr1, kr2, kr3	pylcall3	"callable", karg1, ...
kr1, kr2, kr3, kr4	pylcall4	"callable", karg1, ...
kr1, kr2, kr3, kr4, kr5	pylcall5	"callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6	pylcall6	"callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6, kr7	pylcall7	"callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6, kr7, kr8	pylcall8	"callable", karg1, ...
	pylcallt	ktrigger, "callable", karg1, ...
kresult	pylcall1t	ktrigger, "callable", karg1, ...
kresult1, kresult2	pylcall2t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3	pylcall3t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4	pylcall4t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4, kr5	pylcall5t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6	pylcall6t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6, kr7	pylcall7t	ktrigger, "callable", karg1, ...
kr1, kr2, kr3, kr4, kr5, kr6, kr7, kr8	pylcall8t	ktrigger, "callable", karg1, ...
	pylcalli	"callable", karg1, ...
iresult	pylcall1i	"callable", iarg1, ...
iresult1, iresult2	pylcall2i	"callable", iarg1, ...
ir1, ir2, ir3	pylcall3i	"callable", iarg1, ...
ir1, ir2, ir3, ir4	pylcall4i	"callable", iarg1, ...
ir1, ir2, ir3, ir4, ir5	pylcall5i	"callable", iarg1, ...
ir1, ir2, ir3, ir4, ir5, ir6	pylcall6i	"callable", iarg1, ...
ir1, ir2, ir3, ir4, ir5, ir6, ir7	pylcall7i	"callable", iarg1, ...
ir1, ir2, ir3, ir4, ir5, ir6, ir7, ir8	pylcall8i	"callable", iarg1, ...
pylcalln	"callable", nresults, kresult1, ..., kresultn, karg1, ...	
pylcallni	"callable", nresults, iresult1, ..., iresultn, iarg1, ...	

Description

This family of opcodes call the specified Python callable at k-time and i-time (i suffix), passing the given arguments. The call is performed in the global environment and the result (the returning value) is copied into the Csound output variables specified.

They pass any number of parameters which are cast to float inside the Python interpreter.

The *pycall/pycalli*, *pycall1/pycall1i* ... *pycall8/pycall8i* opcodes can accomodate for a number of results ranging from 0 to 8 according to their numerical prefix (0 is omitted).

The *pycalln/pycallni* opcodes can accomodate for any number of results: the callable name is followed by the number of output arguments, then come the list of Csound output variable and the list of parameters to be passed.

The returning value of the callable must be *None* for *pycall* or *pycalli*, a float for *pycall1i* or *pycall1i* and a tuple (with proper size) of floats for the *pycall2/pycall2i* ... *pycall8/pycall8i* and *pycalln/pycallni* opcodes.

Examples

Example 413. Calling a C or Python function

Supposing we have previously defined or imported a function named `get_number_from_pool` as:

```
from random import random, choice

# a pool of 100 numbers
pool = [i ** 1.3 for i in range(100)]

def get_number_from_pool(n, p):
    # substitute an old number with the new number?
    if random() < p:
        i = choice(range(len(pool)))
        pool[i] = n

    # return a random number from the pool
    return choice(pool)
```

then the following orchestra code

```
k2    pycall1 "get_number_from_pool", k1, p6
```

would set k2 randomly from a pool of numbers changing in time. You can pass new pools elements and control the change rate from the orchestra.

Example 414. Calling a Function Object

A more generic implementation of the previous example makes use of a simple function object:

```
from random import random, choice

class GetNumberFromPool:
    def __init__(self, e, begin=0, end=100, step=1):
        self.pool = [i ** e for i in range(begin, end, step)]

    def __call__(self, n, p):
        # substitute an old number with the new number?
        if random() < p:
            i = choice(range(len(pool)))
            pool[i] = n
```

```
# return a random number from the pool
return choice(pool)

get_number_from_pool1 = GetNumberFromPool(1.3)
get_number_from_pool2 = GetNumberFromPool(1.5, 50, 250, 2)
```

Then the following orchestra code:

```
k2    pycall1 "get_number_from_pool1", k1, p6
k4    pycall1 "get_number_from_pool2", k3, p7
```

would set k2 and k3 randomly from a pool of numbers changing in time. You can pass new pools elements (here k1 and k3) and control the change rate (here p6 and p7) from the orchestra.

As you can see in the first snippet, you can customize the initialization of the pool as well as create several pools.

Credits

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pyeval Opcodes

pyeval — Evaluate a generic Python expression and store the result in a Csound variable at k-time or i-time (i suffix).

Syntax

```
kresult pyeval "expression"

iresult pyevali "expression"

kresult pyleval "expression"

iresult pylevali "expression"

kresult pyevalt ktrigger, "expression"

kresult pylevalt ktrigger, "expression"
```

Description

These opcodes evaluate a generic Python expression and store the result in a Csound variable at k-time or i-time (i suffix).

The expression must evaluate in a float or an object that can be cast to a float.

They can be used effectively to transfer data from a Python object into a Csound variable.

Example of the pyleval Opcode Group

The code:

```
k1          pyleval      "v1"
```

will copy the content of the Python variable v1 into the Csound variable k1 at each k-time.

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pyexec Opcodes

pyexec — Execute a script from a file at k-time or i-time (i suffix).

Syntax

```
pyexec "filename"

pyexeci "filename"

pylexec "filename"

pylexeci "filename"

pyexec ktrigger, "filename"

pylexec ktrigger, "filename"
```

Description

Execute a script from a file at k-time or i-time (i suffix).

This is not the same as calling the script with the `system()` call, since the code is executed by the embedded interpreter.

The code contained in the specified file is executed in the global environment for opcodes `pyexec` and `pyexeci` and in the private environment for the opcodes `pylexec` and `pylexeci`.

These opcodes perform no message passing. However, since the statement has access to the main namespace and the private namespace, it can interact with objects previously created in that environment.

The "local" version of the *pyexec* opcodes are useful when the code ran by different instances of an instrument should not interact.

Example of the pyexec Opcode Group

Example 415. Orchestra (pyexec.orc)

```
sr=44100
kr=4410
ksmps=10
nchnls=1

;If you're not running CsoundAC you need the following line
;to initialize the python interpreter
;pyinit

    pyruni "import random"

    pyexeci "pyexec1.py"

instr 1

    pyexec          "pyexec2.py"

    pylexeci        "pyexec3.py"
    pylexec         "pyexec4.py"

endin
```

Example 416. Score (pyexec.sco)

```
il 0 0.01
il 0 0.01
```

Example 417. The pyexec1.py Script

```
import time, os

print
print "Welcome to Csound!"

try:
    s = ', %s?' % os.getenv('USER')
except:
    s = '?'

print 'What sound do you want to hear today%s' % s
answer = raw_input()
```

Example 418. The pyexec2.py script

```
print 'your answer is "%s"' % answer
```

Example 419. The pyexec3.py script

```
message = 'a private random number: %f' % random.random()
```

Example 420. The pyexec4.py script

```
print message
```

If I run this example on my machine I get something like:

```
Using ../../csound.xmg
Csound Version 4.19 (Mar 23 2002)
Embedded Python interpreter version 2.2
orchname: pyexec.orc
scorename: pyexec.sco
sorting score ...
... done
orch compiler:
11 lines read
instr 1
Csound Version 4.19 (Mar 23 2002)
displays suppressed

Welcome to Csound!
What sound do you want to hear today, maurizio?
```

then I answer

a sound

then Csound continues with the normal performance

```
your answer is "a sound"
a private random number: 0.884006
new alloc for instr 1:
your answer is "a sound"
a private random number: 0.884006
your answer is "a sound"
a private random number: 0.889868
your answer is "a sound"
a private random number: 0.884006
your answer is "a sound"
a private random number: 0.889868
your answer is "a sound"
a private random number: 0.884006
your answer is "a sound"
...
```

In the same instrument a message is created in the private namespace and printed, appearing different for each instance.

Credits

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pyinit Opcodes

pyinit — Initialize the Python interpreter.

Syntax

```
pyinit
```

Description

In the command-line version of Csound, you must first invoke the *pyinit* opcode in the orchestra header to initialize the Python interpreter, before using any of the other Python opcodes.

But if you use the Python opcodes in the CsoundAC version of Csound, you need not invoke *pyinit*, because CsoundAC automatically initializes the Python interpreter for you. In addition, CsoundAC automatically creates a Python interface to the Csound API, in the form a global instance of the `CsoundAC.CppSound` class named `csound`. Therefore, Python code written in the Csound orchestra has access to the global `csound` object.

Credits

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pyrun Opcodes

pyrun — Run a Python statement or block of statements.

Syntax

```
pyrun "statement"

pyruni "statement"

pylrun "statement"

pylruni "statement"

pyrunt ktrigger, "statement"

pylrunt ktrigger, "statement"
```

Description

Execute the specified Python statement at k-time (*pyrun* and *pylrun*) or i-time (*pyruni* and *pylruni*).

The statement is executed in the global environment for *pyrun* and *pyruni* or the local environment for *pylrun* and *pylruni*.

These opcodes perform no message passing. However, since the statement have access to the main namespace and the private namespace, it can interact with objects previously created in that environment.

The "local" version of the *pyrun* opcodes are useful when the code ran by different instances of an instrument should not interact.

Example of the pyrun Opcode Group

Example 421. Orchestra

```
sr=44100
kr=4410
ksmps=10
nchnls=1

;If you're not running CsoundAC you need the following line
;to initialize the python interpreter
;pyinit

pyruni "import random"

instr 1
    ; This message is stored in the main namespace
    ; and is the same for every instance
    pyruni "message = 'a global random number: %f' % random.random()"
    pyrun "print message"

    ; This message is stored in the private namespace
    ; and is different for different instances
    pylruni "message = 'a private random number: %f' % random.random()"
    pylrun "print message"
endin
```

Example 422. Score

```
i1 0 0.1
```

Running this score you should get intermixed pairs of messages from the two instances of instrument 1.

The first message of each pair is stored into the main namespace and so the second instance overwrites the message of the first instance. The result is that first message will be the same for both instances.

The second message is different for the two instances, being stored in the private namespace.

Credits

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rand

rand — Generates a controlled random number series.

Description

Output is a controlled random number series between *-amp* and *+amp*

Syntax

```
ares rand xamp [, iseed] [, isel] [, ioffset]
```

```
kres rand xamp [, iseed] [, isel] [, ioffset]
```

Initialization

iseed (optional, default=0.5) -- a seed value for the recursive pseudo-random formula. A value between 0 and 1 will produce an initial output of *kamp * iseed*. A value greater than 1 will be seeded from the system clock. A negative value will cause seed re-initialization to be skipped. The default seed value is .5.

isel (optional, default=0) -- if zero, a 16-bit number is generated. If non-zero, a 31-bit random number is generated. Default is 0.

ioffset (optional, default=0) -- a base value added to the random result. New in Csound version 4.03.

Performance

kamp, *xamp* -- range over which random numbers are distributed.

The internal pseudo-random formula produces values which are uniformly distributed over the range *kamp* to *-kamp*. *rand* will thus generate uniform white noise with an R.M.S value of *kamp / root 2*.

Examples

Here is an example of the rand opcode. It uses the file *rand.csd* [examples/rand.csd].

Example 423. Example of the rand opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o rand.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
```

```
instr 1
; Choose a random frequency between 4,100 and 44,100.
kfreq rand 20000
kcps = kfreq + 24100

al oscil 30000, kcps, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

randh, randi

Credits

Example written by Kevin Conder.

Thanks to a note from John ffitch, I changed the names of the parameters.

randh

randh — Generates random numbers and holds them for a period of time.

Description

Generates random numbers and holds them for a period of time.

Syntax

```
ares randh xamp, xcps [, iseed] [, isize] [, ioffset]
```

```
kres randh kamp, kcps [, iseed] [, isize] [, ioffset]
```

Initialization

iseed (optional, default=0.5) -- seed value for the recursive pseudo-random formula. A value between 0 and +1 will produce an initial output of *kamp * iseed*. A negative value will cause seed re-initialization to be skipped. A value greater than 1 will seed from system time, this is the best option to generate a different random sequence for each run.

isize (optional, default=0) -- if zero, a 16 bit number is generated. If non-zero, a 31-bit random number is generated. Default is 0.

ioffset (optional, default=0) -- a base value added to the random result. New in Csound version 4.03.

Performance

kamp, *xamp* -- range over which random numbers are distributed.

kcps, *xcps* -- the frequency which new random numbers are generated.

The internal pseudo-random formula produces values which are uniformly distributed over the range *-kamp* to *+kamp*. *rand* will thus generate uniform white noise with an R.M.S value of *kamp / root 2*.

The remaining units produce band-limited noise: the *kcps* and *xcps* parameters permit the user to specify that new random numbers are to be generated at a rate less than the sampling or control frequencies. *randh* will hold each new number for the period of the specified cycle.

Examples

Here is an example of the randh opcode. It uses the file *randh.csd* [examples/randh.csd].

Example 424. Example of the randh opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o randh.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```

```
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  ; Choose a random frequency between 200 and 1000.
  ; Generate new random numbers at 4 Hz.
  kamp = 400
  kcps = 4
  iseed = 0.5
  isize = 0
  ioffset = 600

  kcps randh 400, 4, 0.5, 0, 600
  printk2 kcps

  al oscil 30000, kcps, 1
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

See Also

rand, *randi*

Credits

Example written by Kevin Conder.

randi

rand — Generates a controlled random number series with interpolation between each new number.

Description

Generates a controlled random number series with interpolation between each new number.

Syntax

```
ares randi kamp, xcps [, iseed] [, isize] [, ioffset]
```

```
kres randi kamp, kcps [, iseed] [, isize] [, ioffset]
```

Initialization

iseed (optional, default=0.5) -- seed value for the recursive pseudo-random formula. A value between 0 and +1 will produce an initial output of *kamp * iseed*. A negative value will cause seed re-initialization to be skipped. A value greater than 1 will seed from system time, this is the best option to generate a different random sequence for each run.

isize (optional, default=0) -- if zero, a 16 bit number is generated. If non-zero, a 31-bit random number is generated. Default is 0.

ioffset (optional, default=0) -- a base value added to the random result. New in Csound version 4.03.

Performance

kamp, *xamp* -- range over which random numbers are distributed.

kcps, *xcps* -- the frequency which new random numbers are generated.

The internal pseudo-random formula produces values which are uniformly distributed over the range *kamp* to *-kamp*. *rand* will thus generate uniform white noise with an R.M.S value of *kamp / root 2*.

The remaining units produce band-limited noise: the *kcps* and *xcps* parameters permit the user to specify that new random numbers are to be generated at a rate less than the sampling or control frequencies. *randi* will produce straight-line interpolation between each new number and the next.

Examples

Here is an example of the randi opcode. It uses the file *randi.csd* [examples/rand_i.csd].

Example 425. Example of the randi opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o randi.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```



```
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  ; Choose a random frequency between 4,100 and 44,100.
  ; Generate new random numbers at 10 Hz.
  kamp = 40000
  kcps = 10
  iseed = 0.5
  isize = 0
  ioffset = 4100

  kcps randi 40000, 10, 0.5, 0, 4100

  al oscil 30000, kcps, 1
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

rand, *randh*

Credits

Example written by Kevin Conder.

random

random — Generates a controlled pseudo-random number series between min and max values.

Description

Generates is a controlled pseudo-random number series between min and max values.

Syntax

```
ares random kmin, kmax
```

```
ires random imin, imax
```

```
kres random kmin, kmax
```

Initialization

imin -- minimum range limit

imax -- maximum range limit

Performance

kmin -- minimum range limit

kmax -- maximum range limit

The *random* opcode is similar to *linrand* and *trirand* but sometimes I [Gabriel Maldonado] find it more convenient because allows the user to set arbitrary minimum and maximum values.

Examples

Here is an example of the random opcode. It uses the file *random.csd* [examples/random.csd].

Example 426. Example of the random opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o random.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number between 220 and 440.
kmin init 220
```

```
kmax init 440
k1 random kmin, kmax

printks "k1 = %f\\n", 0.1, k1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
k1 = 414.232056
k1 = 419.393402
k1 = 275.376373
```

See Also

linrand, randomh, randomi, trirand

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

randomh

randomh — Generates random numbers with a user-defined limit and holds them for a period of time.

Description

Generates random numbers with a user-defined limit and holds them for a period of time.

Syntax

```
ares randomh kmin, kmax, acps
```

```
kres randomh kmin, kmax, kcps
```

Performance

kmin -- minimum range limit

kmax -- maximum range limit

kcps, acps -- rate of random break-point generation

The *randomh* opcode is similar to *randh* but allows the user to set arbitrary minimum and maximum values.

Examples

Here is an example of the randomh opcode. It uses the file *randomh.csd* [examples/randomh.csd].

Example 427. Example of the randomh opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o randomh.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Choose a random frequency between 220 and 440 Hz.
; Generate new random numbers at 10 Hz.
kmin = 220
kmax = 440
kcps = 10

k1 randomh kmin, kmax, kcps

printks "k1 = %f\\n", 0.1, k1
endin
```

```
</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
k1 = 220.000000
k1 = 414.232056
k1 = 284.095184
```

See Also

randh, random, randomi

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

randomi

randomi — Generates a user-controlled random number series with interpolation between each new number.

Description

Generates a user-controlled random number series with interpolation between each new number.

Syntax

```
ares randomi kmin, kmax, acps
```

```
kres randomi kmin, kmax, kcps
```

Performance

kmin -- minimum range limit

kmax -- maximum range limit

kcps, acps -- rate of random break-point generation

The *randomi* opcode is similar to *randi* but allows the user to set arbitrary minimum and maximum values.

Examples

Here is an example of the randomi opcode. It uses the file *randomi.csd* [examples/randomi.csd].

Example 428. Example of the randomi opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o randomi.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Choose a random frequency between 220 and 440.
; Generate new random numbers at 10 Hz.
kmin init 220
kmax init 440
kcps init 10

k1 randomi kmin, kmax, kcps

printks "k1 = %f\\n", 0.1, k1
endin
```

```
</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
k1 = 220.000000
k1 = 414.226196
k1 = 284.101074
```

See Also

randi, random, randomh

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

rbjeq

rbjeq — Parametric equalizer and filter opcode with 7 filter types, based on algorithm by Robert Bristow-Johnson.

Description

Parametric equalizer and filter opcode with 7 filter types, based on algorithm by Robert Bristow-Johnson.

Syntax

```
ar rbjeq asig, kfco, klvl, kQ, kS[, imode]
```

Initialization

imode (optional, defaults to zero) - sum of:

- 1: skip initialization (should be used in tied, or re-initialized notes only)

and exactly one of the following values to select filter type:

- 0: resonant lowpass filter. *kQ* controls the resonance: at the cutoff frequency (*kfco*), the amplitude gain is *kQ* (e.g. 20 dB for *kQ* = 10), and higher *kQ* values result in a narrower resonance peak. If *kQ* is set to $\sqrt{0.5}$ (about 0.7071), there is no resonance, and the filter has a response that is very similar to that of `butterlp`. If *kQ* is less than $\sqrt{0.5}$, there is no resonance, and the filter has a -6 dB / octave response from about $kfco * kQ$ to *kfco*. Above *kfco*, there is always a -12 dB / octave cutoff.



NOTE

The `rbjeq` lowpass filter is basically the same as `ar pareq asig, kfco, 0, kQ, 2` but is faster to calculate.

- 2: resonant highpass filter. The parameters are the same as for the lowpass filter, but the equivalent filter is `butterhp` if *kQ* is 0.7071, and "ar `pareq asig, kfco, 0, kQ, 1`" in other cases.
- 4: bandpass filter. *kQ* controls the bandwidth, which is $kfco / kQ$, and must be always less than $sr / 2$. The bandwidth is measured between -3 dB points (i.e. amplitude gain = 0.7071), beyond which there is a +/- 6 dB / octave slope. This filter type is very similar to `ar butterbp asig, kfco, kfco / kQ`.
- 6: band-reject filter, with the same parameters as the bandpass filter, and a response similar to that of `butterbr`.
- 8: peaking EQ. It has an amplitude gain of 1 (0 dB) at 0 Hz and $sr / 2$, and *klvl* at the center frequency (*kfco*). Thus, *klvl* controls the amount of boost (if it is greater than 1), or cut (if it is less than 1). Setting *klvl* to 1 results in a flat response. Similarly to the bandpass and band-reject filters, the bandwidth is determined by $kfco / kQ$ (which must be less than $sr / 2$ again); however, this time it is between \sqrt{klvl} points (or, in other words, half the boost or cut in decibels). NOTE: excessively low or high values of *klvl* should be avoided (especially with 32-bit floats), though the opcode was tested with *klvl* = 0.01 and *klvl* = 100. *klvl* = 0 is always an error, unlike in the case of `pareq`, which does allow a zero level.
- 10: low shelf EQ, controlled by *klvl* and *kS* (*kQ* is ignored by this filter type). There is an amp-

litude gain of *klvl* at zero frequency, while the level of high frequencies (around $sr / 2$) is not changed. At the corner frequency (*kfco*), the gain is $\sqrt{\text{klvl}}$ (half the boost or cut in decibels). The *kS* parameter controls the steepness of the slope of the frequency response (see below).

- 12: high shelf EQ. Very similar to the low shelf EQ, but affects the high frequency range.

The default value for *imode* is zero (lowpass filter, initialization not skipped).

Performance

ar -- the output signal.

asig -- the input signal



NOTE

If the input contains silent sections, on Intel CPUs a significant slowdown can occur due to denormals. In such cases, it is recommended to process the input signal with "denorm" opcode before filtering it with *rbjeq* (and actually many other filters).

kfco -- cutoff, corner, or center frequency, depending on filter type, in Hz. It must be greater than zero, and less than $sr / 2$ (the range of about $sr * 0.0002$ to $sr * 0.49$ should be safe).

klvl -- level (amount of boost or cut), as amplitude gain (e.g. 1: flat response, 4: 12 dB boost, 0.1: 20 dB cut); zero or negative values are not allowed. It is recognized by the peaking and shelving EQ types (8, 10, 12) only, and is ignored by other filters.

kQ -- resonance (also *kfco* / bandwidth in many filter types). Not used by the shelving EQs (*imode* = 10 and 12). The exact meaning of this parameter depends on the filter type (see above), but it should be always greater than zero, and usually (*kfco* / *kQ*) less than $sr / 2$.

kS -- shelf slope parameter for shelving filters. Must be greater than zero; a higher value means a steeper slope, with resonance if $kS > 1$ (however, a too high *kS* value may make the filter unstable). If *kS* is set to exactly 1, the shelf slope is as steep as possible without a resonance. Note that the effect of *kS* - especially if it is greater than 1 - also depends on *klvl*, and it does not have any well defined unit.

Examples

Here is an example of the *rbjeq* opcode. It uses the file *rbjeq.csd* [examples/rbjeq.csd].

Example 429. An example of the *rbjeq* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o rbjeq.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr      = 44100
ksmps   = 10
nchnls  = 1

instr 1

a1      vco2    10000, 155.6      ; sawtooth wave
```

```
kfco      expon      8000, p3, 200          ; filter frequency
a1        rbjeq      a1, kfco, 1, kfco * 0.005, 1, 0 ; resonant lowpass
          out a1

          endin

</CsInstruments>
<CsScore>

i 1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Original algorithm by Robert Bristow-Johnson
Csound orchestra version by Josep M Comajuncosas, Aug 1999
Converted to C (with optimizations and bug fixes) by Istvan Varga, Dec 2002

readclock

readclock — Reads the value of an internal clock.

Description

Reads the value of an internal clock.

Syntax

```
ir readclock inum
```

Initialization

inum -- the number of a clock. There are 32 clocks numbered 0 through 31. All other values are mapped to clock number 32.

ir -- value at i-time, of the clock specified by *inum*

Performance

Between a *clockon* and a *clockoff* opcode, the CPU time used is accumulated in the clock. The precision is machine dependent but is the millisecond range on UNIX and Windows systems. The *readclock* opcode reads the current value of a clock at initialization time.

Examples

Here is an example of the readclock opcode. It uses the file *readclock.csd* [examples/readclock.csd].

Example 430. Example of the readclock opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o readclock.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1.
instr 1
; Start clock #1.
clockon 1
; Do something that keeps Csound busy.
a1 oscili 10000, 440, 1
out a1
; Stop clock #1.
clockoff 1
; Print the time accumulated in clock #1.
i1 readclock 1
print i1
```

```
endin

</CsInstruments>
<CsScore>

; Initialize the function tables.
; Table 1: an ordinary sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for one second starting at 0:00.
i 1 0 1
; Play Instrument #1 for one second starting at 0:01.
i 1 1 1
; Play Instrument #1 for one second starting at 0:02.
i 1 2 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1: i1 = 0.000
instr 1: i1 = 90.000
instr 1: i1 = 180.000
```

See Also

clockoff, clockon

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
July, 1999

Example written by Kevin Conder.

New in Csound version 3.56

readk

readk — Periodically reads an orchestra control-signal value from an external file.

Description

Periodically reads an orchestra control-signal value from a named external file in a specific format.

Syntax

```
kres readk ifilename, iformat, iprd
```

Initialization

ifilename -- an integer N denoting a file named "readk.N" or a character string (in double quotes, spaces permitted) denoting the external file name. For a string, it may either be a full path name with directory specified or a simple filename. In the later case, the file is sought first in the current directory, then in SSDIR, and finally in SFDIR.

iformat -- specifies the input data format:

- 1 = 8-bit signed integers (char)
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = ASCII long integers (plain text)
- 8 = ASCII floats (plain text)

Note that A-law and U-law formats are not available, and that all formats except the last two are binary. The input file should be a "raw", headerless data file.

iprd -- the rate (period) in seconds, rounded to the nearest orchestra control period, at which the signal is read from the input file. A value of 0 implies one control period (the enforced minimum), which will read new values at the orchestra control rate. Longer periods will cause the same values to repeat for more than one control period unless interpolation is used.

Performance

kres -- output of the signal read from ifilename.

This opcode allows a generated control signal value to be read from a named external file. The file should contain no header information but it should contain a regularly sampled time series of control values. For ASCII text formats, the values are assumed to be separated by at least one whitespace character. There may be any number of *readk* opcodes in an instrument or orchestra and they may read from the same or different files.

Examples

Here is an example of the *readk* opcode. It uses the file *dumpk.csd* [examples/readk.csd].

Example 431. Example of the readk opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o readk.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

0dbfs = 1
; By Andres Cabrera 2008

instr 1
; Read a number from the file every 0.5 seconds
kfibo readk "fibonacci.txt", 7, 0.5
kpitchclass = 8 + ((kfibo % 12)/100)
printk2 kpitchclass
kcps = cpspch( kpitchclass )
printk2 kcps
a1 oscil 0.5, kcps, 1
out a1
endin

</CsInstruments>
<CsScore>
f 1 0 1024 10 1
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

See Also

dumpk, dumpk2, dumpk3, dumpk4, readk2, readk3, readk4

Credits

By: John ffitch and Barry Vercoe

1999 or earlier

readk2

readk2 — Periodically reads two orchestra control-signal values from an external file.

Description

Periodically reads two orchestra control-signal values from an external file.

Syntax

```
kr1, kr2 readk2 ifilename, iformat, iprd
```

Initialization

ifilename -- an integer N denoting a file named "readk.N" or a character string (in double quotes, spaces permitted) denoting the external file name. For a string, it may either be a full path name with directory specified or a simple filename. In the later case, the file is sought first in the current directory, then in SSDIR, and finally in SFDIR.

iformat -- specifies the input data format:

- 1 = 8-bit signed integers (char)
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = ASCII long integers (plain text)
- 8 = ASCII floats (plain text)

Note that A-law and U-law formats are not available, and that all formats except the last two are binary. The input file should be a "raw", headerless data file.

iprd -- the rate (period) in seconds, rounded to the nearest orchestra control period, at which the signals are read from the input file. A value of 0 implies one control period (the enforced minimum), which will read new values at the orchestra control rate. Longer periods will cause the same values to repeat for more than one control period unless interpolation is used.

Performance

kr1, *kr2* -- output of the signals read from ifilename.

This opcode allows two generated control signal values to be read from a named external file. The file should contain no header information but it should contain a regularly sampled time series of control values. For binary formats, the individual samples of each signal are interleaved. For ASCII text formats, the values are assumed to be separated by at least one whitespace character. The two "channels" in a sample frame may be on the same line or separated by newline characters, it does not matter. There may be any number of *readk2* opcodes in an instrument or orchestra and they may read from the same or different files.

Examples

See the example for *readk*. The only difference between *readk* and *readk2* is that *readk2* can read two values at a time from the file.

See Also

dumpk, *dumpk2*, *dumpk3*, *dumpk4*, *readk*, *readk3*, *readk4*

Credits

By: John ffitch and Barry Vercoe

1999 or earlier

readk3

readk3 — Periodically reads three orchestra control-signal values from an external file.

Description

Periodically reads three orchestra control-signal values from an external file.

Syntax

```
kr1, kr2, kr3 readk3 ifilename, iformat, iprd
```

Initialization

ifilename -- an integer N denoting a file named "readk.N" or a character string (in double quotes, spaces permitted) denoting the external file name. For a string, it may either be a full path name with directory specified or a simple filename. In the later case, the file is sought first in the current directory, then in SSDIR, and finally in SFDIR.

iformat -- specifies the input data format:

- 1 = 8-bit signed integers (char)
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = ASCII long integers (plain text)
- 8 = ASCII floats (plain text)

Note that A-law and U-law formats are not available, and that all formats except the last two are binary. The input file should be a "raw", headerless data file.

iprd -- the rate (period) in seconds, rounded to the nearest orchestra control period, at which the signals are read from the input file. A value of 0 implies one control period (the enforced minimum), which will read new values at the orchestra control rate. Longer periods will cause the same values to repeat for more than one control period unless interpolation is used.

Performance

kr1, *kr2*, *kr3* -- output of the signals read from ifilename.

This opcode allows three generated control signal values to be read from a named external file. The file should contain no header information but it should contain a regularly sampled time series of control values. For binary formats, the individual samples of each signal are interleaved. For ASCII text formats, the values are assumed to be separated by at least one whitespace character. The three "channels" in a sample frame may be on the same line or separated by newline characters, it does not matter. There may be any number of *readk3* opcodes in an instrument or orchestra and they may read from the same or different files.

Examples

See the example for *readk*. The only difference between *readk* and *readk3* is that *readk3* can read three values at a time from the file.

See Also

dumpk, *dumpk2*, *dumpk3*, *dumpk4*, *readk*, *readk2*, *readk4*

Credits

By: John ffitch and Barry Vercoe

1999 or earlier

readk4

readk4 — Periodically reads four orchestra control-signal values from an external file.

Description

Periodically reads four orchestra control-signal values from an external file.

Syntax

```
kr1, kr2, kr3, kr4 readk4 ifilename, iformat, iprd
```

Initialization

ifilename -- an integer N denoting a file named "readk.N" or a character string (in double quotes, spaces permitted) denoting the external file name. For a string, it may either be a full path name with directory specified or a simple filename. In the later case, the file is sought first in the current directory, then in SSDIR, and finally in SFDIR.

iformat -- specifies the input data format:

- 1 = 8-bit signed integers (char)
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = ASCII long integers (plain text)
- 8 = ASCII floats (plain text)

Note that A-law and U-law formats are not available, and that all formats except the last two are binary. The input file should be a "raw", headerless data file.

iprd -- the rate (period) in seconds, rounded to the nearest orchestra control period, at which the signals are read from the input file. A value of 0 implies one control period (the enforced minimum), which will read new values at the orchestra control rate. Longer periods will cause the same values to repeat for more than one control period unless interpolation is used.

Performance

kr1, kr2, kr3, kr4 -- output of the signals read from ifilename.

This opcode allows four generated control signal values to be read from a named external file. The file should contain no header information but it should contain a regularly sampled time series of control values. For binary formats, the individual samples of each signal are interleaved. For ASCII text formats, the values are assumed to be separated by at least one whitespace character. The four "channels" in a sample frame may be on the same line or separated by newline characters, it does not matter. There may be any number of *readk4* opcodes in an instrument or orchestra and they may read from the same or different files.

Examples

See the example for *readk*. The only difference between *readk* and *readk4* is that *readk4* can read four values at a time from the file.

See Also

dumpk, *dumpk2*, *dumpk3*, *dumpk4*, *readk*, *readk2*, *readk3*

Credits

By: John ffitch and Barry Vercoe

1999 or earlier

reinit

reinit — Suspends a performance while a special initialization pass is executed.

Description

Suspends a performance while a special initialization pass is executed.

Whenever this statement is encountered during a p-time pass, performance is temporarily suspended while a special Initialization pass, beginning at *label* and continuing to *rireturn* or *endin*, is executed. Performance will then be resumed from where it left off.

Syntax

```
reinit label
```

Examples

The following statements will generate an exponential control signal whose value moves from 440 to 880 exactly ten times over the duration p3. They use the file *reinit.csd* [examples/reinit.csd].

Example 432. Example of the reinit opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o reinit.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

instr 1

reset:
    timeout 0, p3/10, contin
    reinit reset

contin:
    kLine expon 440, p3/10, 880
    aSig oscil 10000, kLine, 1
    out aSig
    rireturn

endin

</CsInstruments>
<CsScore>

f1 0 4096 10 1

i1 0 10

e

</CsScore>
</CsoundSynthesizer>
```

See Also

rigoto, rireturn

release

release — Indicates whether a note is in its “release” stage.

Description

Provides a way of knowing when a note off message for the current note is received. Only a noteoff message with the same MIDI note number as the one which triggered the note will be reported by *release*.

Syntax

kflag **release**

Performance

kflag -- indicates whether the note is in its “release” stage. (1 if a note off is received, otherwise 0)

release outputs current note state. If current note is in the “release” stage (i.e. if its duration has been extended with *xtratim* opcode and if it has only just deactivated), then the *kflag* output argument is set to 1. Otherwise (in sustain stage of current note), *kflag* is set to 0.

This opcode is useful for implementing complex release-oriented envelopes. When used in conjunction with *xtratim* it can provide an alternative to the hard-coded behaviour of the “r” opcodes (*linsegr*, *expsegr* et al), where release time is set to the longest time specified in the active instrument.

Examples

See the examples for *xtratim*.

See Also

xtratim

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.47

remoteport

remoteport — Defines the port for use with the remote system.

Description

Defines the port for use with the *insremot*, *midremot*, *insglobal* and *midglobal* opcodes.

Syntax

```
remoteport iportnum
```

Initialization

iportnum -- number of the port to be used. If zero or negative the default port 40002 is selected.

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
Novemer, 2006

New in Csound version 5.05

remove

`remove` — Removes the definition of an instrument.

Description

Removes the definition of an instrument as long as it is not in use.

Syntax

```
remove insnum
```

Initialization

insnum -- number or name of the instrument to be deleted

Performance

As long as the indicated instrument is not active, *remove* deletes the instrument and memory associated with it. It should be treated with care as it is possible that in some cases its use may lead to a crash.

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
June, 2006

New in Csound version 5.04

repluck

repluck — Physical model of the plucked string.

Description

repluck is an implementation of the physical model of the plucked string. A user can control the pluck point, the pickup point, the filter, and an additional audio signal, *axcite*. *axcite* is used to excite the 'string'. Based on the Karplus-Strong algorithm.

Syntax

```
ares repluck iplk, kamp, icps, kpick, kreftl, axcite
```

Initialization

iplk -- The point of pluck is *iplk*, which is a fraction of the way up the string (0 to 1). A pluck point of zero means no initial pluck.

icps -- The string plays at *icps* pitch.

Performance

kamp -- Amplitude of note.

kpick -- Proportion of the way along the string to sample the output.

kreftl -- the coefficient of reflection, indicating the lossiness and the rate of decay. It must be strictly between 0 and 1 (it will complain about both 0 and 1).

Performance

axcite -- A signal which excites the string.

Examples

Here is an example of the repluck opcode. It uses the file *repluck.csd* [examples/repluck.csd].

Example 433. Example of the repluck opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o repluck.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

```
; Instrument #1.
instr 1
  iplk = 0.75
  kamp = 30000
  icps = 220
  kpick = 0.75
  krefl = 0.5
  axcite oscil 1, 1, 1

  apluck repluck iplk, kamp, icps, kpick, krefl, axcite

  out apluck
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

wgpluck2

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
1997

New in version 3.47

reson

reson — A second-order resonant filter.

Description

A second-order resonant filter.

Syntax

```
ares reson asig, kcf, kbw [, iscl] [, iskip]
```

Initialization

iscl (optional, default=0) -- coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than *kcf* are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. (This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise.) A zero value signifies no scaling of the signal, leaving that to some later adjustment (see *balance*). The default value is 0.

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

ares -- the output signal at audio rate.

asig -- the input signal at audio rate.

kcf -- the center frequency of the filter, or frequency position of the peak response.

kbw -- bandwidth of the filter (the Hz difference between the upper and lower half-power points).

reson is a second-order filter in which *kcf* controls the center frequency, or frequency position of the peak response, and *kbw* controls its bandwidth (the frequency difference between the upper and lower half-power points).

Examples

Here is an example of the reson opcode. It uses the file *reson.csd* [examples/reson.csd].

Example 434. Example of the reson opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o reson.wav -W ;; for file output any platform
</CsOptions>
```

```
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a sine waveform.
asine buzz 15000, 440, 3, 1

; Vary the cut-off frequency from 220 to 1280.
kcf line 220, p3, 1320
kbw init 20

; Run the sine through a resonant filter.
ares reson asine, kcf, kbw

; Give the filtered signal the same amplitude
; as the original signal.
al balance ares, asine
out al
endin

</CsInstruments>
<CsScore>

; Table #1, an ordinary sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 4 seconds.
i 1 0 4
e

</CsScore>
</CsoundSynthesizer>
```

See Also

areson, aresonk, atone, atonek, port, portk, resonk, tone, tonek

Credits

Example written by Kevin Conder.

resonk

resonk — A second-order resonant filter.

Description

A second-order resonant filter.

Syntax

```
kres resonk ksig, kcf, kbw [, iscl] [, iskip]
```

Initialization

iscl (optional, default=0) -- coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than *kcf* are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. (This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise.) A zero value signifies no scaling of the signal, leaving that to some later adjustment (see *balance*). The default value is 0.

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

kres -- the output signal at control-rate.

ksig -- the input signal at control-rate.

kcf -- the center frequency of the filter, or frequency position of the peak response.

kbw -- bandwidth of the filter (the Hz difference between the upper and lower half-power points).

resonk is like *reson* except its output is at control-rate rather than audio rate.

See Also

areson, *aresonk*, *atone*, *atonek*, *port*, *portk*, *reson*, *tone*, *tonek*

Credits

Author: Robin Whittle
Australia
May 1997

resonr

resonr — A bandpass filter with variable frequency response.

Description

Implementations of a second-order, two-pole two-zero bandpass filter with variable frequency response.

Syntax

```
ares resonr asig, kcf, kbw [, iscl] [, iskip]
```

Initialization

The optional initialization variables for *resonr* are identical to the i-time variables for *reson*.

iscl (optional, default=0) -- coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than *kcf* are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise. A zero value signifies no scaling of the signal, leaving that to some later adjustment (see *balance*). The default value is 0.

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- input signal to be filtered

kcf -- cutoff or resonant frequency of the filter, measured in Hz

kbw -- bandwidth of the filter (the Hz difference between the upper and lower half-power points)

resonr and *resonz* are variations of the classic two-pole bandpass resonator (*reson*). Both filters have two zeroes in their transfer functions, in addition to the two poles. *resonz* has its zeroes located at $z = 1$ and $z = -1$. *resonr* has its zeroes located at $+\sqrt{R}$ and $-\sqrt{R}$, where R is the radius of the poles in the complex z -plane. The addition of zeroes to *resonr* and *resonz* results in the improved selectivity of the magnitude response of these filters at cutoff frequencies close to 0, at the expense of less selectivity of frequencies above the cutoff peak.

resonr and *resonz* are very close to constant-gain as the center frequency is swept, resulting in a more efficient control of the magnitude response than with traditional two-pole resonators such as *reson*.

resonr and *resonz* produce a sound that is considerably different from *reson*, especially for lower center frequencies; trial and error is the best way of determining which resonator is best suited for a particular application.

Examples

Here is an example of the *resonr* and *resonz* opcodes. It uses the file *resonr.csd* [examples/resonr.csd].

Example 435. Example of the resonr and resonz opcodes.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac        -iadc       -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o resonr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Sean Costello */
; Orchestra file for resonant filter sweep of a sawtooth-like waveform.
; The outputs of reson, resonr, and resonz are scaled by coefficients
; specified in the score, so that each filter can be heard on its own
; from the same instrument.

sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 1

    idur      =      p3
    ibegfreq  =      p4                      ; beginning of sweep frequency
    iendfreq  =      p5                      ; ending of sweep frequency
    ibw       =      p6                      ; bandwidth of filters in Hz
    ifreq     =      p7                      ; frequency of gbuzz that is to be filtered
    iamp      =      p8                      ; amplitude to scale output by
    ires      =      p9                      ; coefficient to scale amount of reson in output
    iresr     =      p10                     ; coefficient to scale amount of resonr in output
    iresz     =      p11                     ; coefficient to scale amount of resonz in output

; Frequency envelope for reson cutoff
    kfreq     linseg ibegfreq, idur * .5, iendfreq, idur * .5, ibegfreq

; Amplitude envelope to prevent clicking
    kenv      linseg 0, .1, iamp, idur - .2, iamp, .1, 0

; Number of harmonics for gbuzz scaled to avoid aliasing
    iharms    =      (sr*.4)/ifreq

    asig      gbuzz 1, ifreq, iharms, 1, .9, 1      ; "Sawtooth" waveform
    ain       =      kenv * asig                  ; output scaled by amp envelope
    ares      reson ain, kfreq, ibw, 1
    aresr     resonr ain, kfreq, ibw, 1
    aresz     resonz ain, kfreq, ibw, 1

    out       ares * ires + aresr * iresr + aresz * iresz

endin

</CsInstruments>
<CsScore>

/* Written by Sean Costello */
f1 0 8192 9 1 1 .25                      ; cosine table for gbuzz generator

i1 0 10 1 3000 200 100 4000 1 0 0        ; reson output with bw = 200
i1 10 10 1 3000 200 100 4000 0 1 0      ; resonr output with bw = 200
i1 20 10 1 3000 200 100 4000 0 0 1      ; resonz output with bw = 200
i1 30 10 1 3000 50 200 8000 1 0 0       ; reson output with bw = 50
i1 40 10 1 3000 50 200 8000 0 1 0       ; resonr output with bw = 50
i1 50 10 1 3000 50 200 8000 0 0 1       ; resonz output with bw = 50
e

</CsScore>
</CsoundSynthesizer>
```

Technical History

resonr and *resonz* were originally described in an article by Julius O. Smith and James B. Angell.¹ Smith and Angell recommended the *resonz* form (zeros at +1 and -1) when computational efficiency was the main concern, as it has one less multiply per sample, while *resonr* (zeroes at + and - the square root of the pole radius R) was recommended for situations when a perfectly constant-gain center peak was required.

Ken Steiglitz, in a later article², demonstrated that *resonz* had constant gain at the true peak of the filter, as opposed to *resonr*, which displayed constant gain at the pole angle. Steiglitz also recommended *resonz* for its sharper notches in the gain curve at zero and Nyquist frequency. Steiglitz's recent book³ features a thorough technical discussion of *reson* and *resonz*, while Dodge and Jerse's textbook⁴ illustrates the differences in the response curves of *reson* and *resonz*.

References

1. Smith, Julius O. and Angell, James B., "A Constant-Gain Resonator Tuned by a Single Coefficient," *Computer Music Journal*, vol. 6, no. 4, pp. 36-39, Winter 1982.
2. Steiglitz, Ken, "A Note on Constant-Gain Digital Resonators," *Computer Music Journal*, vol. 18, no. 4, pp. 8-10, Winter 1994.
3. Ken Steiglitz, *A Digital Signal Processing Primer, with Applications to Digital Audio and Computer Music*. Addison-Wesley Publishing Company, Menlo Park, CA, 1996.
4. Dodge, Charles and Jerse, Thomas A., *Computer Music: Synthesis, Composition, and Performance*. New York: Schirmer Books, 1997, 2nd edition, pp. 211-214.

See Also

resonz

Credits

Author: Sean Costello
Seattle, Washington
1999

New in Csound version 3.55

resonx

resonx — Emulates a stack of filters using the reson opcode.

Description

resonx is equivalent to a filters consisting of more layers of *reson* with the same arguments, serially connected. Using a stack of a larger number of filters allows a sharper cutoff. They are faster than using a larger number instances in a Csound orchestra of the old opcodes, because only one initialization and k- cycle are needed at time and the audio loop falls entirely inside the cache memory of processor.

Syntax

```
ares resonx asig, kcf, kbw [, inumlayer] [, iscl] [, iskip]
```

Initialization

inumlayer (optional) -- number of elements in the filter stack. Default value is 4.

iscl (optional, default=0) -- coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than kcf are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. (This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise.) A zero value signifies no scaling of the signal, leaving that to some later adjustment (see *balance*). The default value is 0.

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- input signal

kcf -- the center frequency of the filter, or frequency position of the peak response.

kbw -- bandwidth of the filter (the Hz difference between the upper and lower half-power points)

See Also

atonex, *tonex*

Credits

Author: Gabriel Maldonado (adapted by John ffitich)
Italy

New in Csound version 3.49

resonxk

resonxk — Control signal resonant filter stack.

Description

resonxk is equivalent to a group of *resonk* filters, with the same arguments, serially connected. Using a stack of a larger number of filters allows a sharper cutoff.

Syntax

```
kres resonxk ksig, kcf, kbw[, inumlayer, iscl, istor]
```

Initialization

inumlayer - number of elements of filter stack. Default value is 4. Maximum value is 10

iscl (optional, default=0) - coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than *kcf* are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. (This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise.) A zero value signifies no scaling of the signal, leaving that to some later adjustment (see *balance*). The default value is 0.

istor (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

kres - output signal

ksig - input signal

kcf - the center frequency of the filter, or frequency position of the peak response.

kbw - bandwidth of the filter (the Hz difference between the upper and lower half-power points)

resonxk is a lot faster than using individual instances in Csound orchestra of the old opcodes, because only one initialization and 'k' cycle are needed at a time, and the audio loop falls entirely inside the cache memory of processor.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

resony

resony — A bank of second-order bandpass filters, connected in parallel.

Description

A bank of second-order bandpass filters, connected in parallel.

Syntax

```
ares resony asig, kbf, kbw, inum, ksep [, isepmode] [, iscl] [, iskip]
```

Initialization

inum -- number of filters

isepmode (optional, default=0) -- if *isepmode* = 0, the separation of center frequencies of each filter is generated logarithmically (using octave as unit of measure). If *isepmode* not equal to 0, the separation of center frequencies of each filter is generated linearly (using Hertz). Default value is 0.

iscl (optional, default=0) -- coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than *kcf* are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. (This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise.) A zero value signifies no scaling of the signal, leaving that to some later adjustment (e.g. *balance*). The default value is 0.

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- audio input signal

kbf -- base frequency, i.e. center frequency of lowest filter in Hz

kbw -- bandwidth in Hz

ksep -- separation of the center frequency of filters in octaves

resony is a bank of second-order bandpass filters, with k-rate variant frequency separation, base frequency and bandwidth, connected in parallel (i.e. the resulting signal is a mix of the output of each filter). The center frequency of each filter depends of *kbf* and *ksep* variables. The maximum number of filters is set to 100.

Examples

Here is an example of the *resony* opcode. It uses the file *resony.csd* [examples/resony.csd], and *beats.wav* [examples/beats.wav].

Example 436. Example of the *resony* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using com-

mand line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o resony.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a nice sawtooth waveform.
asig vco 32000, 220, 1

; Vary the base frequency from 60 to 600 Hz.
kbf line 60, p3, 600
kbw = 50
inum = 2
ksep = 1
isepmode = 0
iscl = 1

al resony asig, kbf, kbw, inum, ksep, isepmode, iscl

out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave for the vco opcode.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Gabriel Maldonado
Italy
1999

Example written by Kevin Conder.

New in Csound version 3.56

resonz

resonz — A bandpass filter with variable frequency response.

Description

Implementations of a second-order, two-pole two-zero bandpass filter with variable frequency response.

Syntax

```
ares resonz asig, kcf, kbw [, iscl] [, iskip]
```

Initialization

The optional initialization variables for *resonr* and *resonz* are identical to the i-time variables for *reson*.

iskip -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

iscl -- coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than *kcf* are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise. A zero value signifies no scaling of the signal, leaving that to some later adjustment (see *balance*). The default value is 0.

Performance

resonr and *resonz* are variations of the classic two-pole bandpass resonator (*reson*). Both filters have two zeroes in their transfer functions, in addition to the two poles. *resonz* has its zeroes located at $z = 1$ and $z = -1$. *resonr* has its zeroes located at $+\sqrt{R}$ and $-\sqrt{R}$, where R is the radius of the poles in the complex z -plane. The addition of zeroes to *resonr* and *resonz* results in the improved selectivity of the magnitude response of these filters at cutoff frequencies close to 0, at the expense of less selectivity of frequencies above the cutoff peak.

resonr and *resonz* are very close to constant-gain as the center frequency is swept, resulting in a more efficient control of the magnitude response than with traditional two-pole resonators such as *reson*.

resonr and *resonz* produce a sound that is considerably different from *reson*, especially for lower center frequencies; trial and error is the best way of determining which resonator is best suited for a particular application.

asig -- input signal to be filtered

kcf -- cutoff or resonant frequency of the filter, measured in Hz

kbw -- bandwidth of the filter (the Hz difference between the upper and lower half-power points)

Technical History

resonr and *resonz* were originally described in an article by Julius O. Smith and James B. Angell.¹ Smith and Angell recommended the *resonz* form (zeros at +1 and -1) when computational efficiency was the main concern, as it has one less multiply per sample, while *resonr* (zeroes at + and - the

square root of the pole radius R) was recommended for situations when a perfectly constant-gain center peak was required.

Ken Steiglitz, in a later article ², demonstrated that *resonz* had constant gain at the true peak of the filter, as opposed to *resonr*, which displayed constant gain at the pole angle. Steiglitz also recommended *resonz* for its sharper notches in the gain curve at zero and Nyquist frequency. Steiglitz's recent book ³ features a thorough technical discussion of *reson* and *resonz*, while Dodge and Jerse's textbook ⁴ illustrates the differences in the response curves of *reson* and *resonz*.

References

1. Smith, Julius O. and Angell, James B., "A Constant-Gain Resonator Tuned by a Single Coefficient," *Computer Music Journal*, vol. 6, no. 4, pp. 36-39, Winter 1982.
2. Steiglitz, Ken, "A Note on Constant-Gain Digital Resonators," *Computer Music Journal*, vol. 18, no. 4, pp. 8-10, Winter 1994.
3. Ken Steiglitz, *A Digital Signal Processing Primer, with Applications to Digital Audio and Computer Music*. Addison-Wesley Publishing Company, Menlo Park, CA, 1996.
4. Dodge, Charles and Jerse, Thomas A., *Computer Music: Synthesis, Composition, and Performance*. New York: Schirmer Books, 1997, 2nd edition, pp. 211-214.

See Also

resonr

Credits

Author: Sean Costello
Seattle, Washington
1999

New in Csound version 3.55

resyn

resyn — Streaming partial track additive synthesis with cubic phase interpolation with pitch control and support for timescale-modified input

Description

The resyn opcode takes an input containing a TRACKS pv streaming signal (as generated, for instance by partials). It resynthesises the signal using linear amplitude and cubic phase interpolation to drive a bank of interpolating oscillators with amplitude and pitch scaling controls. Resyn is a modified version of sinsyn, allowing for the resynthesis of data with pitch and timescale changes.

Syntax

```
asig resyn fin, kscal, kpitch, kmaxtracks, ifn
```

Performance

asig -- output audio rate signal

fin -- input pv stream in TRACKS format

kscal -- amplitude scaling

kpitch -- pitch scaling

kmaxtracks -- max number of tracks in resynthesis. Limiting this will cause a non-linear filtering effect, by discarding newer and higher-frequency tracks (tracks are ordered by start time and ascending frequency, respectively)

ifn -- function table containing one cycle of a sinusoid (sine or cosine)

Examples

Example 437. Example

```
ain inch 1 ; input signal
fsl,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fsl,fsi2,.003,1,3,500 ; partial tracking
aout resyn fst, 1, 1.5, 500, 1 ; resynthesis (up a 5th)
out aout
```

The example above shows partial tracking of an ifd-analysis signal and cubic-phase additive resynthesis with pitch shifting.

Credits

Author: Victor Lazzarini;
June 2005

New plugin in version 5

November 2004.

reverb

reverb — Reverberates an input signal with a “natural room” frequency response.

Description

Reverberates an input signal with a “natural room” frequency response.

Syntax

```
ares reverb asig, krvt [, iskip]
```

Initialization

iskip (optional, default=0) -- initial disposition of delay-loop data space (cf. *reson*). The default value is 0.

Performance

krvt -- the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude).

A standard *reverb* unit is composed of four *comb* filters in parallel followed by two *alpass* units in series. Loop times are set for optimal “natural room response.” Core storage requirements for this unit are proportional only to the sampling rate, each unit requiring approximately 3K words for every 10KC. The *comb*, *alpass*, *delay*, *tone* and other Csound units provide the means for experimenting with alternate reverberator designs.

Since output from the standard *reverb* will begin to appear only after 1/20 second or so of delay, and often with less than three-fourths of the original power, it is normal to output both the source and the reverberated signal. If *krvt* is inadvertently set to a non-positive number, *krvt* will be reset automatically to 0.01. (New in Csound version 4.07.) Also, since the reverberated sound will persist long after the cessation of source events, it is normal to put *reverb* in a separate instrument to which sound is passed via a *global variable*, and to leave that instrument running throughout the performance.

Examples

Here is an example of the reverb opcode. It uses the file *reverb.csd* [examples/reverb.csd].

Example 438. Example of the reverb opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o reverb.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
```

```
ksmps = 10
nchnls = 1

; init an audio receiver/mixer
gal init 0

; Instrument #1. (there may be many copies)
instr 1
  ; generate a source signal
  a1 oscili 7000, cpspch(p4), 1
  ; output the direct sound
  out a1
  ; and add to audio receiver
  gal = gal + a1
endin

; (highest instr number executed last)
instr 99
  ; reverberate whatever is in gal
  a3 reverb gal, 1.5
  ; and output the result
  out a3
  ; empty the receiver for the next pass
  gal = 0
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 128 10 1

; p4 = frequency (in a pitch-class)
; Play Instrument #1 for a tenth of a second, p4=6.00
i 1 0 0.1 6.00
; Play Instrument #1 for a tenth of a second, p4=6.02
i 1 1 0.1 6.02
; Play Instrument #1 for a tenth of a second, p4=6.04
i 1 2 0.1 6.04
; Play Instrument #1 for a tenth of a second, p4=6.06
i 1 3 0.1 6.06

; Make sure the reverb remains active.
i 99 0 6
e

</CsScore>
</CsoundSynthesizer>
```

See Also

alpass, comb, valpass, vcomb

Credits

Author: William “Pete” Moss
University of Texas at Austin
Austin, Texas USA
January 2002

reverb2

reverb2 — Same as the nreverb opcode.

Description

Same as the *nreverb* opcode.

Syntax

```
ares reverb2 asig, ktime, khdif [, iskip] [,inumCombs] \  
      [, ifnCombs] [, inumAlpas] [, ifnAlpas]
```

reverbsc

reverbsc — 8 delay line stereo FDN reverb, based on work by Sean Costello

Description

8 delay line stereo FDN reverb, with feedback matrix based upon physical modeling scattering junction of 8 lossless waveguides of equal characteristic impedance. Based on Csound orchestra version by Sean Costello.

Syntax

```
aoutL, aoutR reverbsc ainL, ainR, kfblvl, kfco[, israte[, ipitchm[, iskip]]]
```

Initialization

israte (optional, defaults to the orchestra sample rate) -- assume a sample rate of *israte*. This is normally set to *sr*, but a different setting can be useful for special effects.

ipitchm (optional, defaults to 1) -- depth of random variation added to delay times, in the range 0 to 10. The default is 1, but this may be too high and may need to be reduced for held pitches such as piano tones.

iskip (optional, defaults to zero) -- if non-zero, initialization of the opcode is skipped, whenever possible.

Performance

aoutL, *aoutR* -- output signals for left and right channel

ainL, *ainR* -- left and right channel input. Note that having an input signal on either the left or right channel only will still result in having reverb output on both channels, making this unit more suitable for reverberating stereo input than the *freeverb* opcode.

kfblvl -- feedback level, in the range 0 to 1. 0.6 gives a good small "live" room sound, 0.8 a small hall, and 0.9 a large hall. A setting of exactly 1 means infinite length, while higher values will make the opcode unstable.

kfco -- cutoff frequency of simple first order lowpass filters in the feedback loop of delay lines, in Hz. Should be in the range 0 to $\text{israte}/2$ (not $\text{sr}/2$). A lower value means faster decay in the high frequency range.

Examples

Here is an example of the *reverbsc* opcode. It uses the file *reverbsc.csd* [examples/reverbsc.csd].

Example 439. An example of the reverbsc opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
```

```
; For Non-realtime ouput leave only the line below:
; -o reverbsc.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr      = 48000
ksmps   = 32
nchnls  = 2
0dbfs   = 1

instr 1
vco2 0.85, 440, 10
kfrq  port 100, 0.004, 20000
a1    butterlp a1, kfrq
a2    linseg 0, 0.003, 1, 0.01, 0.7, 0.005, 0, 1, 0
a1    = a1 * a2
a2    = a1 * p5
a1    = a1 * p4
denorm a1, a2
aL, aR reverbbsc a1, a2, 0.85, 12000, sr, 0.5, 1
outs a1 + aL, a2 + aR
endin

</CsInstruments>
<CsScore>
i 1 0 1 0.71 0.71
i 1 1 1 0 1
i 1 2 1 -0.71 0.71
i 1 3 1 1 0
i 1 4 4 0.71 0.71
e
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Istvan Varga
2005

rewindscore

rewindscore — Rewinds the playback position of the current score performance.

Description

Rewinds the playback position of the current score performance..

Syntax

```
rewindscore
```

Examples

Here is an example of the rewindscore opcode.

Example 440. Example of the rewindscore opcode.

```
instr 1
  rewindscore
endin
```

See Also

setscorepos

Credits

Author: Victor Lazzarini
2008

New in Csound version 5.09

rezzy

rezzy — A resonant low-pass filter.

Description

A resonant low-pass filter.

Syntax

```
ares rezzy asig, xfco, xres [, imode, iskip]
```

Initialization

imode (optional, default=0) -- high-pass or low-pass mode. If zero, *rezzy* is low-pass. If not zero, *rezzy* is high-pass. Default value is 0. (New in Csound version 3.50) *iskip* (optional, default=0) -- if non zero skip the initialisation of the filter. (New in Csound version 4.23f13 and 5.0)

Performance

asig -- input signal

xfco -- filter cut-off frequency in Hz. As of version 3.50, may i-,k-, or a-rate.

xres -- amount of resonance. Values of 1 to 100 are typical. Resonance should be one or greater. As of version 3.50, may a-rate, i-rate, or k-rate.

rezzy is a resonant low-pass filter created empirically by Hans Mikelson.

Examples

Here is an example of the *rezzy* opcode. It uses the file *rezzy.csd* [examples/rezzy.csd].

Example 441. Example of the *rezzy* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o rezzy.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a nice sawtooth waveform.
asig vco 32000, 220, 1

; Vary the filter-cutoff frequency from .2 to 2 KHz.
kfco line 200, p3, 2000
```



```
; Set the resonance amount.
kres init 25

al rezzy asig, kfco, kres

out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave for the vco opcode.
f 1 0 16384 10 1

; Play Instrument #1 for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

biquad, moogvcf

Credits

Author: Hans Mikelson
October 1998

Example written by Kevin Conder.

New in Csound version 3.49

rigoto

rigoto — Transfers control during a reinit pass.

Description

Similar to *igoto*, but effective only during a *reinit* pass (i.e., no-op at standard i-time). This statement is useful for bypassing units that are not to be reinitialized.

Syntax

```
rigoto label
```

See Also

cigoto, igoto, reinit, rireturn

rireturn

rireturn — Terminates a reinit pass.

Description

Terminates a *reinit* pass (i.e., no-op at standard i-time). This statement, or an *endin*, will cause normal performance to be resumed.

Syntax

rireturn

Examples

The following statements will generate an exponential control signal whose value moves from 440 to 880 exactly ten times over the duration p3. They use the file *reinit.csd* [examples/reinit.csd].

Example 442. Example of the reireturn opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o reinit.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

instr 1

reset:
  timeout 0, p3/10, contin
  reinit reset

contin:
  kLine expon 440, p3/10, 880
  aSig oscil 10000, kLine, 1
  out aSig
  reireturn

endin

</CsInstruments>
<CsScore>

f1 0 4096 10 1

i1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

See Also

reinit, rigoto

rms

rms — Determines the root-mean-square amplitude of an audio signal.

Description

Determines the root-mean-square amplitude of an audio signal. It low-pass filters the actual value, to average in the manner of a VU meter.

Syntax

```
kres rms asig [, ihp] [, iskip]
```

Initialization

ihp (optional, default=10) -- half-power point (in Hz) of a special internal low-pass filter. The default value is 10.

iskip (optional, default=0) -- initial disposition of internal data space (see *reson*). The default value is 0.

Performance

asig -- input audio signal

kres -- low-pass filtered rms value of the input signal

rms output values *kres* will trace the root-mean-square value of the audio input *asig*. This unit is not a signal modifier, but functions rather as a signal power-gauge. It uses an internal low-pass filter to make the response smoother. *ihp* can be used to control this smoothing. The higher the value, the "snappier" the measurement.

This opcode can also be used as an envelope follower.

The *kres* output from this opcode is given in amplitude and depends on *Odbfs*. If you want the output in decibels, you can use *dbamp*

Examples

```
arms rms      asig ; get rms value of signal asig
```

Here is an example of the rms opcode. It uses the file *rms.csd* [examples/rms.csd].

Example 443. Example of the rms opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
```

```
; Audio out   Audio in   No messages
-odac         -iadc      -d  -m0    ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o rms.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
ksmps = 128
nchnls = 1

;Example by Andres Cabrera 2007

Odbfs = 1

FLpanel "rms", 400, 100, 50, 50
  gkrms text, gihrms text FLtext "Rms", -100, 0, 0.1, 3, 110, 30, 60, 50
  gkihp, gihandle FLtext "ihp", 0, 10, 0.05, 1, 100, 30, 220, 50
  gkrms slider, gihrms slider FLslider "", -60, -0.5, -1, 5, -1, 380, 20, 10, 10

FLpanelEnd
FLrun

FLsetVal_i 5, gihandle
; Instrument #1.
instr 1
  al inch 1

label:
  kval rms al, i(gkihp) ;measures rms of input channel 1
  rreturn

  kval = dbamp(kval) ; convert to db full scale
  printk 0.5, kval
  FLsetVal 1, kval, gihrms slider ;update the slider and text values
  FLsetVal 1, kval, gihrms text
  knewihp changed gkihp ; reinit when ihp text has changed
  if (knewihp == 1) then
    reinit label ;needed because ihp is an i-rate parameter
  endif
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one minute
i 1 0 60
e

</CsScore>
</CsoundSynthesizer>
```

See Also

balance, gain

rnd

rnd — Returns a random number in a unipolar range at the rate given by the input argument.

Description

Returns a random number in a unipolar range at the rate given by the input argument.

Syntax

rnd(x) (init- or control-rate only)

Where the argument within the parentheses may be an expression. These value converters sample a global random sequence, but do not reference *seed*. The result can be a term in a further expression.

Performance

Returns a random number in the unipolar range 0 to *x*.

Examples

Here is an example of the rnd opcode. It uses the file *rnd.csd* [examples/rnd.csd].

Example 444. Example of the rnd opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o rnd.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number from 0 to 1.
i1 = rnd(1)
print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #1 for one second.
i 1 1 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should be:

```
rnd at i-rate: 0.973500   rnd at k-rate: 0.139405
rnd at i-rate: 0.973500   rnd at k-rate: 0.040065
rnd at i-rate: 0.973500   rnd at k-rate: 0.412845
rnd at i-rate: 0.973500   rnd at k-rate: 0.440650
rnd at i-rate: 0.973500   rnd at k-rate: 0.663581
rnd at i-rate: 0.973500   rnd at k-rate: 0.876723
rnd at i-rate: 0.973500   rnd at k-rate: 0.302459
rnd at i-rate: 0.973500   rnd at k-rate: 0.398580
rnd at i-rate: 0.973500   rnd at k-rate: 0.448875
rnd at i-rate: 0.973500   rnd at k-rate: 0.907728
```

See Also

birnd

Credits

Author: Barry L. Vercoe
MIT
Cambridge, Massachusetts
1997

Original Example written by Kevin Conder. Modified by John Harrison.

rnd31

rnd31 — 31-bit bipolar random opcodes with controllable distribution.

Description

31-bit bipolar random opcodes with controllable distribution. These units are portable, i.e. using the same seed value will generate the same random sequence on all systems. The distribution of generated random numbers can be varied at k-rate.

Syntax

```
ax rnd31 kscl, krpow [, iseed]
```

```
ix rnd31 iscl, irpow [, iseed]
```

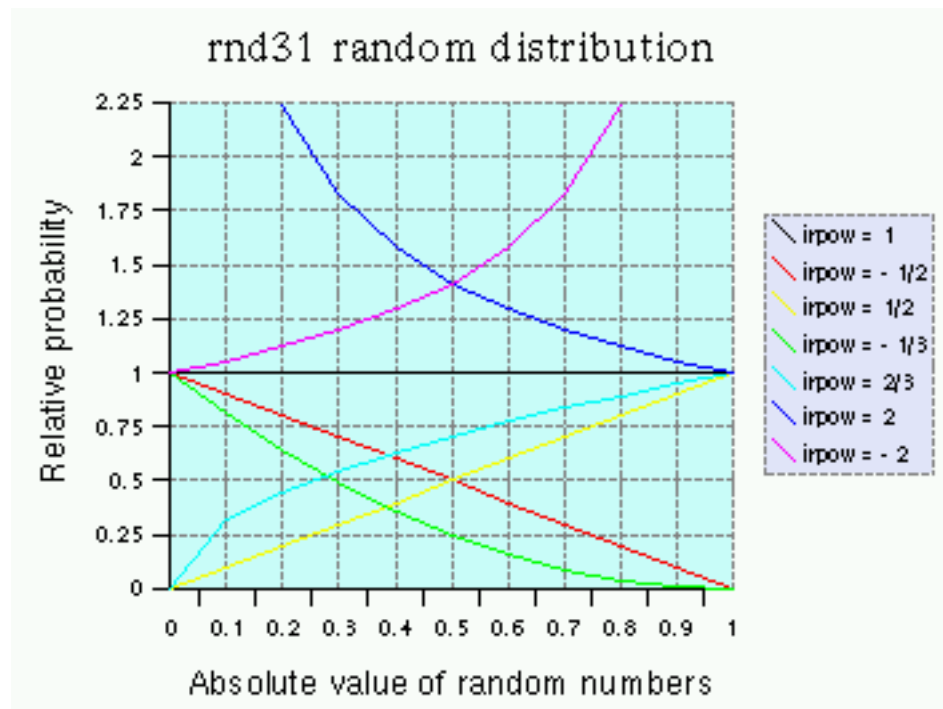
```
kx rnd31 kscl, krpow [, iseed]
```

Initialization

ix -- i-rate output value.

iscl -- output scale. The generated random numbers are in the range -iscl to iscl.

irpow -- controls the distribution of random numbers. If *irpow* is positive, the random distribution (*x* is in the range -1 to 1) is $\text{abs}(x)^{(1/\text{irpow}) - 1}$; for negative *irpow* values, it is $(1 - \text{abs}(x))^{((-1/\text{irpow}) - 1)}$. Setting *irpow* to -1, 0, or 1 will result in uniform distribution (this is also faster to calculate).



A graph of distributions for different values of *irpow*.

iseed (optional, default=0) -- seed value for random number generator (positive integer in the range

1 to 2147483646 ($2^{31} - 2$). Zero or negative value seeds from current time (this is also the default). Seeding from current time is guaranteed to generate different random sequences, even if multiple random opcodes are called in a very short time.

In the a- and k-rate version the seed is set at opcode initialization. With i-rate output, if iseed is zero or negative, it will seed from current time in the first call, and return the next value from the random sequence in successive calls; positive seed values are set at all i-rate calls. The seed is local for a- and k-rate, and global for i-rate units.



Notes

- although seed values up to 2147483646 are allowed, it is recommended to use smaller numbers (< 1000000) for portability, as large integers may be rounded to a different value if 32-bit floats are used.
- i-rate *rnd31* with a positive seed will always produce the same output value (this is not a bug). To get different values, set seed to 0 in successive calls, which will return the next value from the random sequence.

Performance

ax -- a-rate output value.

kx -- k-rate output value.

kscl -- output scale. The generated random numbers are in the range -kscl to kscl. It is the same as *iscl*, but can be varied at k-rate.

kpow -- controls the distribution of random numbers. It is the same as *ipow*, but can be varied at k-rate.

Examples

Here is an example of the *rnd31* opcode at a-rate. It uses the file *rnd31.csd* [examples/rnd31.csd].

Example 445. An example of the *rnd31* opcode at a-rate.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o rnd31.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create random numbers at a-rate in the range -2 to 2 with
; a triangular distribution, seed from the current time.
a31 rnd31 2, -0.5

; Use the random numbers to choose a frequency.
```

```
afreq = a31 * 500 + 100

al oscil 30000, afreq, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the `rnd31` opcode at k-rate. It uses the file `rnd31_krate.csd` [examples/rnd31_krate.csd].

Example 446. An example of the `rnd31` opcode at k-rate.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o rnd31_krate.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create random numbers at k-rate in the range -1 to 1
; with a uniform distribution, seed=10.
k1 rnd31 1, 0, 10

printks "k1=%f\\n", 0.1, k1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1=0.112106
k1=-0.274665
k1=0.403933
```

Here is an example of the `rnd31` opcode that uses the number 7 as a seed value. It uses the file `rnd31_seed7.csd` [examples/rnd31_seed7.csd].

Example 447. An example of the `rnd31` opcode that uses the number 7 as a seed value.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o rnd31_seed7.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; i-rate random numbers with linear distribution, seed=7.
; (Note that the seed was used only in the first call.)
i1 rnd31 1, 0.5, 7
i2 rnd31 1, 0.5
i3 rnd31 1, 0.5

print i1
print i2
print i3
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1: i1 = -0.649
instr 1: i2 = -0.761
instr 1: i3 = 0.677
```

Here is an example of the `rnd31` opcode that uses the current time as a seed value. It uses the file `rnd31_time.csd` [examples/rnd31_time.csd].

Example 448. An example of the `rnd31` opcode that uses the current time as a seed value.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o rnd31_time.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
```

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  ; i-rate random numbers with linear distribution,
  ; seeding from the current time. (Note that the seed
  ; was used only in the first call.)
  i1 rnd31 1, 0.5, 0
  i2 rnd31 1, 0.5
  i3 rnd31 1, 0.5

  print i1
  print i2
  print i3
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1: i1 = -0.691
instr 1: i2 = -0.686
instr 1: i3 = -0.358
```

Credits

Author: Istvan Varga

New in version 4.16

round

round — Returns the integer value nearest to x ; if the fractional part of x is exactly 0.5, the direction of rounding is undefined.

Description

The integer value nearest to x ; if the fractional part of x is exactly 0.5, the direction of rounding is undefined.

Syntax

round(x) (init-, control-, or audio-rate arg allowed)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

See Also

abs, exp, int, log, log10, i, sqrt

Credits

Author: Istvan Varga
New in Csound 5
2005

rspline

rspline — Generate random spline curves.

Description

Generate random spline curves.

Syntax

```
ares rspline xrangeMin, xrangeMax, kcpsMin, kcpsMax
```

```
kres rspline krangeMin, krangeMax, kcpsMin, kcpsMax
```

Performance

kres, ares -- Output signal

xrangeMin, xrangeMax -- Range of values of random-generated points

kcpsMin, kcpsMax -- Range of point-generation rate. Min and max limits are expressed in cps.

rspline (random-spline-curve generator) is similar to *jspline* but output range is defined by means of two limit values. Also in this case, real output range could be a bit greater of range values, because of interpolating curves between each pair of random-points.

At present time generated curves are quite smooth when *cpsMin* is not too different from *cpsMax*. When *cpsMin-cpsMax* interval is big, some little discontinuity could occur, but it should not be a problem, in most cases. Maybe the algorithm will be improved in next versions.

These opcodes are often better than *jitter* when user wants to “naturalize” or “analogize” digital sounds. They could be used also in algorithmic composition, to generate smooth random melodic lines when used together with *samphold* opcode.

Note that the result is quite different from the one obtained by filtering white noise, and they allow the user to obtain a much more precise control.

Credits

Author: Gabriel Maldonado

New in version 4.15

rtclock

rtclock — Read the real time clock from the operating system.

Description

Read the real-time clock from the operating system.

Syntax

```
ires rtclock
```

```
kres rtclock
```

Performance

Read the real-time clock from operating system. Under Windows, this changes only once per second. Under GNU/Linux, it ticks every microsecond. Performance under other systems varies.

Examples

Here is an example of the rtclock opcode. It uses the file *rtclock.csd* [examples/rtclock.csd].

Example 449. Example of the rtclock opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o rtclock.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1
instr 1
; Get the system time.
k1 rtclock
; Print it once per second.
printk 1, k1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```


Its output should include lines like this:

```
i  1 time      0.00002: 1018236096.00000
i  1 time      1.00002: 1018236224.00000
```

Credits

Author: John ffitch

Example written by Kevin Conder.

New in version 4.10

s16b14

s16b14 — Creates a bank of 16 different 14-bit MIDI control message numbers.

Description

Creates a bank of 16 different 14-bit MIDI control message numbers.

Syntax

```
i1,...,i16 s16b14 ichan, ictlno_msb1, ictlno_lsb1, imin1, imax1, \  
    initvalue1, ifn1,..., ictlno_msb16, ictlno_lsb16, imin16, imax16, initvalue16, ifn16  
  
k1,...,k16 s16b14 ichan, ictlno_msb1, ictlno_lsb1, imin1, imax1, \  
    initvalue1, ifn1,..., ictlno_msb16, ictlno_lsb16, imin16, imax16, initvalue16, ifn16
```

Initialization

i1 ... i64 -- output values

ichan -- MIDI channel (1-16)

ictlno_msb1 ictlno_msb32 -- MIDI control number, most significant byte (0-127)

ictlno_lsb1 ictlno_lsb32 -- MIDI control number, least significant byte (0-127)

imin1 ... imin64 -- minimum values for each controller

imax1 ... imax64 -- maximum values for each controller

init1 ... init64 -- initial value for each controller

ifn1 ... ifn64 -- function table for conversion for each controller

icutoff1 ... icutoff64 -- low-pass filter cutoff frequency for each controller

Performance

k1 ... k64 -- output values

s16b14 is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

s16b14 allows a bank of 16 different MIDI control message numbers. It uses 14-bit values instead of MIDI's normal 7-bit values.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

In the i-rate version of *s16b14*, there is not an initial value input argument. The output is taken directly from the current status of internal controller array of Csound.

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

s32b14

s32b14 — Creates a bank of 32 different 14-bit MIDI control message numbers.

Description

Creates a bank of 32 different 14-bit MIDI control message numbers.

Syntax

```
i1,...,i32 s32b14 ichan, ictlno_msb1, ictlno_lsb1, imin1, imax1, \  
    initvalue1, ifn1,..., ictlno_msb32, ictlno_lsb32, imin32, imax32, initvalue32, ifn32  
  
k1,...,k32 s32b14 ichan, ictlno_msb1, ictlno_lsb1, imin1, imax1, \  
    initvalue1, ifn1,..., ictlno_msb32, ictlno_lsb32, imin32, imax32, initvalue32, ifn32
```

Initialization

i1 ... i64 -- output values

ichan -- MIDI channel (1-16)

ictlno_msb1 ictlno_msb32 -- MIDI control number, most significant byte (0-127)

ictlno_lsb1 ictlno_lsb32 -- MIDI control number, least significant byte (0-127)

imin1 ... imin64 -- minimum values for each controller

imax1 ... imax64 -- maximum values for each controller

init1 ... init64 -- initial value for each controller

ifn1 ... ifn64 -- function table for conversion for each controller

icutoff1 ... icutoff64 -- low-pass filter cutoff frequency for each controller

Performance

k1 ... k64 -- output values

s32b14 is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

s32b14 allows a bank of 32 different MIDI control message numbers. It uses 14-bit values instead of MIDI's normal 7-bit values.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

In the i-rate version of *s32b14*, there is not an initial value input argument. The output is taken directly from the current status of internal controller array of Csound.

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

scale

scale — Arbitrary signal scaling.

Description

Scales incoming value to user-definable range. Similar to scale object found in popular dataflow languages.

Syntax

```
kscl scale kinput, kmax, kmin
```

Performance

kin -- Input value. Can originate from any k-rate source as long as that source's output is in range 0-1.

kmin -- Minimum value of the resultant scale operation.

kmax -- Maximum value of the resultant scale operation.

Examples

Here is an example of the scale opcode. It uses the file *scale.csd* [examples/scale.csd].

Example 450. Example of the scale opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  Silent
-odac          -iadc     -d      ;;realtime output
</CsOptions>
<CsInstruments>

sr = 22050
ksmps = 10
nchnls = 2

/*--- */

instr 1 ; scale test

kmod ctrl1 1, 1, 0, 1
printk2 kmod

kout scale kmod, 0, -127
printk2 kout

endin

/*--- */
</CsInstruments>
<CsScore>

i1 0 8888
e
</CsScore>
</CsoundSynthesizer>
```

See Also

gainslider, logcurve, expcurve

Credits

Author: David Akbari
October
2006

samphold

samphold — Performs a sample-and-hold operation on its input.

Description

Performs a sample-and-hold operation on its input.

Syntax

```
ares samphold asig, agate [, ival] [, ivstor]
```

```
kres samphold ksig, kgate [, ival] [, ivstor]
```

Initialization

ival, *ivstor* (optional) -- controls initial disposition of internal save space. If *ivstor* is zero the internal “hold” value is set to *ival* ; else it retains its previous value. Defaults are 0,0 (i.e. init to zero)

Performance

kgate, *xgate* -- controls whether to hold the signal.

samphold performs a sample-and-hold operation on its input according to the value of *gate*. If *gate* != 0, the input samples are passed to the output; If *gate* = 0, the last output value is repeated. The controlling *gate* can be a constant, a control signal, or an audio signal.

Examples

```
asrc  buzz      10000,440,20, 1    ; band-limited pulse train
adif  diff      asrc              ; emphasize the highs
anew  balance   adif, asrc         ; but retain the power
agate  reson     asrc,0,440        ; use a lowpass of the original
asamp  samphold  anew, agate       ; to gate the new audiosig
aout  tone      asamp,100         ; smooth out the rough edges
```

See Also

diff, *downsamp*, *integ*, *interp*, *upsamp*

sandpaper

sandpaper — Semi-physical model of a sandpaper sound.

Description

sandpaper is a semi-physical model of a sandpaper sound. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares sandpaper iamp, idettack [, inum] [, idamp] [, imaxshake]
```

Initialization

iamp -- Amplitude of output. Note: As these instruments are stochastic, this is only a approximation.

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 128.

idamp (optional) -- the damping factor, as part of this equation:

$$\text{damping_amount} = 0.998 + (\text{idamp} * 0.002)$$

The default *damping_amount* is 0.999 which means that the default value of *idamp* is 0.5. The maximum *damping_amount* is 1.0 (no damping). This means the maximum value for *idamp* is 1.0.

The recommended range for *idamp* is usually below 75% of the maximum value.

imaxshake (optional) -- amount of energy to add back into the system. The value should be in range 0 to 1.

Examples

Here is an example of the sandpaper opcode. It uses the file *sandpaper.csd* [examples/sandpaper.csd].

Example 451. Example of the sandpaper opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o sandpaper.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

;orchestra -----

sr =          44100
kr =          4410
```

```
ksmps = 10
nchnls = 1

instr 01 ;an example of sandpaper blocks
a1 line 2, p3, 2 ;preset amplitude increase
a2 sandpaper p4, 0.01 ;sandpaper needs a little amp help at these settings
a3 product a1, a2 ;increase amplitude
    out a3
endin

</CsInstruments>
<CsScore>

;score -----

i1 0 1 26000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

cabasa, crunch, sekere, stix

Credits

Author: Perry Cook, part of the PhOLIES (Physically-Oriented Library of Imitated Environmental Sounds)

Adapted by John ffitch

University of Bath, Codemist Ltd.

Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

scanhammer

scanhammer — Copies from one table to another with a gain control.

Description

This is a variant of *tablecopy*, copying from one table to another, starting at *ipos*, and with a gain control. The number of points copied is determined by the length of the source. Other points are not changed. This opcode can be used to “hit” a string in the scanned synthesis code.

Syntax

```
scanhammer isrc, idst, ipos, imult
```

Initialization

isrc -- source function table.

idst -- destination function table.

ipos -- starting position (in points).

imult -- gain multiplier. A value of 0 will leave values unchanged.

See Also

scantable

Credits

Author: Matt Gilliard
April 2002

New in version 4.20

scans

scans — Generate audio output using scanned synthesis.

Description

Generate audio output using scanned synthesis.

Syntax

```
ares scans kamp, kfreq, ifn, id [, iorder]
```

Initialization

ifn -- ftable containing the scanning trajectory. This is a series of numbers that contains addresses of masses. The order of these addresses is used as the scan path. It should not contain values greater than the number of masses, or negative numbers. See the *introduction to the scanned synthesis section*.

id -- ID number of the *scanu* opcode's waveform to use

iorder (optional, default=0) -- order of interpolation used internally. It can take any value in the range 1 to 4, and defaults to 4, which is quartic interpolation. The setting of 2 is quadratic and 1 is linear. The higher numbers are slower, but not necessarily better.

Performance

kamp -- output amplitude. Note that the resulting amplitude is also dependent on instantaneous value in the wavetable. This number is effectively the scaling factor of the wavetable.

kfreq -- frequency of the scan rate

Examples

Here is an example of the scanned synthesis. It uses the file *scans.csd* [examples/scans.csd], and *string-128.matrix* [examples/string-128.matrix].

Example 452. Example of the scans opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o scans.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

    sr = 44100
    ksmpr = 128
    nchnls = 1

    instr 1
a0 = 0
; scanu init, irate, ifnvel, ifnmass, ifnstif, ifncentr, ifndamp, kmass, kstif, kcentr, kdamp, il
```

```
scanu 1, .01, 6, 2, 3, 4, 5, 2, .1, .1, -.01, .
;ar scans kamp, kfreq, ifntraj, id
a1 scans ampdb(p4), cpspch(p5), 7, 2
out a1
endin

</CsInstruments>
<CsScore>

; Initial condition
f1 0 128 7 0 64 1 64 0

; Masses
f2 0 128 -7 1 128 1

; Spring matrices
f3 0 16384 -23 "string-128.matrix"

; Centering force
f4 0 128 -7 0 128 2

; Damping
f5 0 128 -7 1 128 1

; Initial velocity
f6 0 128 -7 0 128 0

; Trajectories
f7 0 128 -5 .001 128 128

; Note list
i1 0 10 86 6.00
i1 11 14 86 7.00
i1 15 20 86 5.00
e

</CsScore>
</CsoundSynthesizer>
```

The matrix file “string-128.matrix”, as well as several other matrices, is also available in a *zipped file* [<http://www.csounds.com/scanned/zip/scanmatrices.zip>] from the *Scanned Synthesis* page [<http://www.csounds.com/scanned/>] at cSounds.com.

Credits

Author: Paris Smaragdis
MIT Media Lab
Boston, Massachusetts USA

New in Csound version 4.05

scantable

scantable — A simpler scanned synthesis implementation.

Description

A simpler scanned synthesis implementation. This is an implementation of a circular string scanned using external tables. This opcode will allow direct modification and reading of values with the table opcodes.

Syntax

```
aout scantable kamp, kpch, ipos, imass, istiff, idamp, ivel
```

Initialization

ipos -- table containing position array.

imass -- table containing the mass of the string.

istiff -- table containing the stiffness of the string.

idamp -- table containing the damping factors of the string.

ivel -- table containing the velocities.

Performance

kamp -- amplitude (gain) of the string.

kpch -- the string's scanned frequency.

Examples

Here is an example of the scantable opcode. It uses the file *scantable.csd* [examples/scantable.csd].

Example 453. Example of the scantable opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o scantable.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Table #1 - initial position
gitl ftgen 1, 0, 128, 7, 0, 64, 1, 64, 0
; Table #2 - masses
```

```
git2 ftgen 2, 0, 128, -7, 1, 128, 1
; Table #3 - stiffness
git3 ftgen 3, 0, 128, -7, 0, 64, 100, 64, 0
; Table #4 - damping
git4 ftgen 4, 0, 128, -7, 1, 128, 1
; Table #5 - initial velocity
git5 ftgen 5, 0, 128, -7, 0, 128, 0

; Instrument #1.
instr 1
  kamp init 20000
  kpch init 220
  ipos = 1
  imass = 2
  istiff = 3
  idamp = 4
  ivel = 5

  a1 scantable kamp, kpch, ipos, imass, istiff, idamp, ivel
  a2 dcblock a1

  out a2
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for ten seconds.
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

See Also

scanhammer

Credits

Author: Matt Gilliard
April 2002

Example written by Kevin Conder.

New in version 4.20

scanu

scanu — Compute the waveform and the wavetable for use in scanned synthesis.

Description

Compute the waveform and the wavetable for use in scanned synthesis.

Syntax

```
scanu init, irate, ifnvel, ifnmass, ifnstif, ifncentr, ifndamp, kmass, \  
      kstif, kcentr, kdamp, ileft, iright, kpos, kstrngth, ain, idisp, id
```

Initialization

init -- the initial position of the masses. If this is a negative number, then the absolute of *init* signifies the table to use as a hammer shape. If *init* > 0, the length of it should be the same as the intended mass number, otherwise it can be anything.

ifnvel -- the ftable that contains the initial velocity for each mass. It should have the same size as the intended mass number.

ifnmass -- ftable that contains the mass of each mass. It should have the same size as the intended mass number.

ifnstif -- ftable that contains the spring stiffness of each connection. It should have the same size as the square of the intended mass number. The data ordering is a row after row dump of the connection matrix of the system.

ifncentr -- ftable that contains the centering force of each mass. It should have the same size as the intended mass number.

ifndamp -- the ftable that contains the damping factor of each mass. It should have the same size as the intended mass number.

ileft -- If *init* < 0, the position of the left hammer (*ileft* = 0 is hit at leftmost, *ileft* = 1 is hit at rightmost).

iright -- If *init* < 0, the position of the right hammer (*iright* = 0 is hit at leftmost, *iright* = 1 is hit at rightmost).

idisp -- If 0, no display of the masses is provided.

id -- If positive, the ID of the opcode. This will be used to point the scanning opcode to the proper waveform maker. If this value is negative, the absolute of this value is the wavetable on which to write the waveshape. That wavetable can be used later from an other opcode to generate sound. The initial contents of this table will be destroyed.

Performance

kmass -- scales the masses

kstif -- scales the spring stiffness

kcentr -- scales the centering force

kdamp -- scales the damping

kpos -- position of an active hammer along the string (*kpos* = 0 is leftmost, *kpos* = 1 is rightmost). The shape of the hammer is determined by *init* and the power it pushes with is *kstrngth*.

kstrngth -- power that the active hammer uses

ain -- audio input that adds to the velocity of the masses. Amplitude should not be too great.

Examples

For an example, see the documentation on *scans*.

Credits

Author: Paris Smaragdis
MIT Media Lab
Boston, Massachusetts USA
March 2000

New in Csound version 4.05

scoreline

scoreline — Issues one or more score line events from an instrument.

Description

Scoreline will issue one or more score events, if ktrig is 1 every k-period. It can handle strings in the same conditions as the standard score. Multi-line strings are accepted, using {{ }} to enclose the string.

Syntax

```
scoreline Sin, ktrig
```

Initialization

“Sin” -- a string (in double-quotes or enclosed by {{ }}) containing one or more score events.

Performance

“ktrig” -- event trigger, 1 issues the score event, 0 bypasses it.

Examples

Here is an example of the scoreline opcode.

Example 454. Example

```
instr 1
  ktrig init 1
  scoreline {{
    i 2 0 3 "flutec3.wav"
    i 2 1 3 "clarac3.wav"
  }}, ktrig
  ktrig = 0
endin

instr 2
  aout soundin p4
  out aout
endin
```

You can use string opcodes like *sprintfk* to produce strings to be passed to *scoreline* like this:

```
Sfil = "/Volumes/Bla/file.aif"
String sprintfk {{i 2 0 %f "%s" %f %f %f %f}}, idur, Sfil, p5, p6, knorm, iskip
scoreline String, ktrig
```

Credits

Author: Victor Lazzarini, 2007

scoreline_i

scoreline_i — Issues one or more score line events from an instrument at i-time.

Description

scoreline_i will issue score events at i-time. It can handle strings in the same conditions as the standard score. Multi-line strings are accepted, using `{{ }}` to enclose the string.

Syntax

```
scoreline_i Sin
```

Initialization

“Sin” -- a string (in double-quotes or enclosed by `{{ }}`) containing one or more score events.

Examples

Here is an example of the scoreline_i opcode.

Example 455. Example

```
instr 1
  scoreline_i {{
    i 2 0 3 "flutec3.wav"
    i 2 1 3 "clar3.wav"
  }}
endin

instr 2
  aout soundin p4
  out aout
endin
```

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

Credits

Author: Victor Lazzarini, 2007

schedkwhen

schedkwhen — Adds a new score event generated by a k-rate trigger.

Description

Adds a new score event generated by a k-rate trigger.

Syntax

```
schedkwhen ktrigger, kmintim, kmaxnum, kinsnum, kwhen, kdur \  
[, ip4] [, ip5] [...]
```

```
schedkwhen ktrigger, kmintim, kmaxnum, "insname", kwhen, kdur \  
[, ip4] [, ip5] [...]
```

Initialization

“insname” -- A string (in double-quotes) representing a named instrument.

ip4, ip5, ... -- Equivalent to p4, p5, etc., in a score *i statement*

Performance

ktrigger -- triggers a new score event. If *ktrigger* = 0, no new event is triggered.

kmintim -- minimum time between generated events, in seconds. If *kmintim* <= 0, no time limit exists. If the *kinsnum* is negative (to turn off an instrument), this test is bypassed.

kmaxnum -- maximum number of simultaneous instances of instrument *kinsnum* allowed. If the number of extant instances of *kinsnum* is >= *kmaxnum*, no new event is generated. If *kmaxnum* is <= 0, it is not used to limit event generation. If the *kinsnum* is negative (to turn off an instrument), this test is bypassed.

kinsnum -- instrument number. Equivalent to p1 in a score *i statement*.

kwhen -- start time of the new event. Equivalent to p2 in a score *i statement*. Measured from the time of the triggering event. *kwhen* must be >= 0. If *kwhen* > 0, the instrument will not be initialized until the actual time when it should start performing.

kdur -- duration of event. Equivalent to p3 in a score *i statement*. If *kdur* = 0, the instrument will only do an initialization pass, with no performance. If *kdur* is negative, a held note is initiated. (See *ihold* and *i statement*.)

Note: While waiting for events to be triggered by *schedkwhen*, the performance must be kept going, or Csound may quit if no score events are expected. To guarantee continued performance, an *f0 statement* may be used in the score.

Examples

Here is an example of the schedkwhen opcode. It uses the file *schedkwhen.csd* [examples/schedkwhen.csd].

Example 456. Example of the schedkwhen opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o schedkwhen.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1 - oscillator with a high note.
instr 1
; Use the fourth p-field as the trigger.
ktrigger = p4
kmintim = 0
kmaxnum = 2
kinsnum = 2
kwhen = 0
kdur = 0.5

; Play Instrument #2 at the same time, if the trigger is set.
schedkwhen ktrigger, kmintim, kmaxnum, kinsnum, kwhen, kdur

; Play a high note.
al oscils 10000, 880, 1
out al
endin

; Instrument #2 - oscillator with a low note.
instr 2
; Play a low note.
al oscils 10000, 220, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; p4 = trigger for Instrument #2 (when p4 > 0).
; Play Instrument #1 for half a second, no trigger.
i 1 0 0.5 0
; Play Instrument #1 for half a second, trigger Instrument #2.
i 1 1 0.5 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Rasmus Ekman
EMS, Stockholm, Sweden

Example written by Kevin Conder.

New in Csound version 3.59

schedkwhennamed

`schedkwhennamed` — Similar to `schedkwhen` but uses a named instrument at init-time.

Description

Similar to *schedkwhen* but uses a named instrument at init-time.

Syntax

```
schedkwhennamed ktrigger, kmintim, kmaxnum, "name", kwhen, kdur \  
[, ip4] [, ip5] [...]
```

Initialization

ip4, ip5, ... -- Equivalent to *p4, p5, etc.*, in a score *i statement*

Performance

ktrigger -- triggers a new score event. If *ktrigger* is 0, no new event is triggered.

kmintim -- minimum time between generated events, in seconds. If *kmintim* is less than or equal to 0, no time limit exists.

kmaxnum -- maximum number of simultaneous instances of named instrument allowed. If the number of extant instances of the named instrument is greater than or equal to *kmaxnum*, no new event is generated. If *kmaxnum* is less than or equal to 0, it is not used to limit event generation.

"name" -- the named instrument's name.

kwhen -- start time of the new event. Equivalent to *p2* in a score *i statement*. Measured from the time of the triggering event. *kwhen* must be greater than or equal to 0. If *kwhen* greater than 0, the instrument will not be initialized until the actual time when it should start performing.

kdur -- duration of event. Equivalent to *p3* in a score *i statement*. If *kdur* is 0, the instrument will only do an initialization pass, with no performance. If *kdur* is negative, a held note is initiated. (See *ihold* and *i statement*.)

Note: While waiting for events to be triggered by *schedkwhennamed*, the performance must be kept going, or Csound may quit if no score events are expected. To guarantee continued performance, an *f0 statement* may be used in the score.

Examples

Here is an example of the `schedkwhennamed` opcode. It uses the file *schedkwhennamed.csd* [examples/schedkwhennamed.csd].

Example 457. Example of the `schedkwhennamed` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  No messages
```

```
-odac          -iadc      -d
; For Non-realtime ouput leave only the line below:
; -o schedkwhennamed.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

    sr          = 48000
    ksmps       = 16
    nchnls      = 2
    0dbfs       = 1

; Example by Jonathan Murphy 2007

gSinstr2 = "printer"

    instr 1

    ktrig      metro      1
    if (ktrig == 1) then
        ;Call instrument "printer" once per second
        schedkwhennamed ktrig, 0, 1, gSinstr2, 0, 1
    endif

    endin

    instr printer

    ktime      timeinsts
               printk2    ktime

    endin

</CsInstruments>
<CsScore>
i1 0 10
e
</CsScore>
</CsoundSynthesizer>
```

See Also

schedkwhen

Credits

Author: Rasmus Ekman
EMS, Stockholm, Sweden

New in Csound version 4.23

schedule

schedule — Adds a new score event.

Description

Adds a new score event.

Syntax

```
schedule insnum, iwhen, idur [, ip4] [, ip5] [...]
```

```
schedule "insname", iwhen, idur [, ip4] [, ip5] [...]
```

Initialization

insnum -- instrument number. Equivalent to p1 in a score *i statement*. *insnum* must be a number greater than the number of the calling instrument.

"insname" -- A string (in double-quotes) representing a named instrument.

iwhen -- start time of the new event. Equivalent to p2 in a score *i statement*. *iwhen* must be nonnegative. If *iwhen* is zero, *insnum* must be greater than or equal to the p1 of the current instrument.

idur -- duration of event. Equivalent to p3 in a score *i statement*.

ip4, ip5, ... -- Equivalent to p4, p5, etc., in a score *i statement*.

Performance

ktrigger -- trigger value for new event

schedule adds a new score event. The arguments, including options, are the same as in a score. The *iwhen* time (p2) is measured from the time of this event.

If the duration is zero or negative the new event is of MIDI type, and inherits the release sub-event from the scheduling instruction.

Examples

Here is an example of the schedule opcode. It uses the file *schedule.csd* [examples/schedule.csd].

Example 458. Example of the schedule opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o schedule.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```



```
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - oscillator with a high note.
instr 1
; Play Instrument #2 at the same time.
schedule 2, 0, p3

; Play a high note.
a1 oscils 10000, 880, 1
out a1
endin

; Instrument #2 - oscillator with a low note.
instr 2
; Play a low note.
a1 oscils 10000, 220, 1
out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for half a second.
i 1 0 0.5
; Play Instrument #1 for half a second.
i 1 1 0.5
e

</CsScore>
</CsoundSynthesizer>
```

See Also

schedwhen

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
November 1998

Example written by Kevin Conder.

New in Csound version 3.491

Based on work by Gabriel Maldonado

Thanks goes to David Gladstein, for clarifying the *iwhen* parameter.

schedwhen

schedwhen — Adds a new score event.

Description

Adds a new score event.

Syntax

```
schedwhen ktrigger, kinsnum, kwhen, kdur [, ip4] [, ip5] [...]
```

```
schedwhen ktrigger, "insname", kwhen, kdur [, ip4] [, ip5] [...]
```

Initialization

ip4, ip5, ... -- Equivalent to *p4, p5, etc.*, in a score *i statement*.

Performance

kinsnum -- instrument number. Equivalent to *p1* in a score *i statement*.

"insname" -- A string (in double-quotes) representing a named instrument.

ktrigger -- trigger value for new event

kwhen -- start time of the new event. Equivalent to *p2* in a score *i statement*.

kdur -- duration of event. Equivalent to *p3* in a score *i statement*.

schedwhen adds a new score event. The event is only scheduled when the k-rate value *ktrigger* is first non-zero. The arguments, including options, are the same as in a score. The *iwhen* time (*p2*) is measured from the time of this event.

If the duration is zero or negative the new event is of MIDI type, and inherits the release sub-event from the scheduling instruction.



Warning

Support for named instruments is broken in version 4.23

Examples

Here is an example of the schedwhen opcode. It uses the file *schedwhen.csd* [examples/schedwhen.csd].

Example 459. Example of the schedwhen opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform
```

```
; Audio out      Audio in
-odac            -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o schedwhen.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1 - oscillator with a high note.
instr 1
; Use the fourth p-field as the trigger.
ktrigger = p4
kinsnum = 2
kwhen = 0
kdur = p3

; Play Instrument #2 at the same time, if the trigger is set.
schedwhen ktrigger, kinsnum, kwhen, kdur

; Play a high note.
al oscils 10000, 880, 1
out al
endin

; Instrument #2 - oscillator with a low note.
instr 2
; Play a low note.
al oscils 10000, 220, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; p4 = trigger for Instrument #2 (when p4 > 0).
; Play Instrument #1 for half a second, trigger Instrument #2.
i 1 0 0.5 1
; Play Instrument #1 for half a second, no trigger.
i 1 1 0.5 0
e

</CsScore>
</CsoundSynthesizer>
```

See Also

schedule

Credits

Author: John fitch
University of Bath/Codemist Ltd.
Bath, UK
November 1998

Example written by Kevin Conder.

New in Csound version 3.491

Based on work by Gabriel Maldonado

seed

seed — Sets the global seed value.

Description

Sets the global seed value for all *x-class noise generators*, as well as other opcodes that use a random call, such as *grain*.



Please Note

rand, *randh*, *randi*, *rnd(x)* and *birnd(x)* are not affected by seed.

Syntax

```
seed ival
```

Performance

Use of *seed* will provide predictable results from an orchestra using with random generators, when required from multiple performances.

When specifying a seed value, *ival* should be an integer between 0 and 2^{32} . If *ival* = 0, the value of *ival* will be derived from the system clock.

sekere

sekere — Semi-physical model of a sekere sound.

Description

sekere is a semi-physical model of a sekere sound. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares sekere iamp, idettack [, inum] [, idamp] [, imaxshake]
```

Initialization

iamp -- Amplitude of output. Note: As these instruments are stochastic, this is only a approximation.

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 64.

idamp (optional) -- the damping factor, as part of this equation:

$$\text{damping_amount} = 0.998 + (\text{idamp} * 0.002)$$

The default *damping_amount* is 0.999 which means that the default value of *idamp* is 0.5. The maximum *damping_amount* is 1.0 (no damping). This means the maximum value for *idamp* is 1.0.

The recommended range for *idamp* is usually below 75% of the maximum value.

imaxshake (optional) -- amount of energy to add back into the system. The value should be in range 0 to 1.

Examples

Here is an example of the sekere opcode. It uses the file *sekere.csd* [examples/sekere.csd].

Example 460. Example of the sekere opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o sekere.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

;orchestra -----

sr =          44100
kr =          4410
ksmps =       10
nchnls =      1
```

```
instr 01
al      sekere p4, 0.01      ;an example of a sekere
        out al
        endin

</CsInstruments>
<CsScore>

;score -----

i1 0 1 26000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

cabasa, crunch, sandpaper, stix

Credits

Author: Perry Cook, part of the PhISEM (Physically Informed Stochastic Event Modeling)
Adapted by John ffitch
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

semitone

semitone — Calculates a factor to raise/lower a frequency by a given amount of semitones.

Description

Calculates a factor to raise/lower a frequency by a given amount of semitones.

Syntax

`semitone(x)`

This function works at a-rate, i-rate, and k-rate.

Initialization

x -- a value expressed in semitones.

Performance

The value returned by the *semitone* function is a factor. You can multiply a frequency by this factor to raise/lower it by the given amount of semitones.

Examples

Here is an example of the semitone opcode. It uses the file *semitone.csd* [examples/semitone.csd].

Example 461. Example of the semitone opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o semitone.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; The root note is A above middle-C (440 Hz)
iroot = 440

; Raise the root note by three semitones to C.
isemitone = 3

; Calculate the new note.
ifactor = semitone(isemitone)
inew = iroot * ifactor

; Print out all of the values.
print iroot
print ifactor
```

```
    print inew
  endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like:

```
instr 1:  iroot = 440.000
instr 1:  ifactor = 1.189
instr 1:  inew = 523.229
```

See Also

cent, db, octave

Credits

Example written by Kevin Conder.

New in version 4.16

sense

sense — Same as the sensekey opcode.

Description

Same as the *sensekey* opcode.

sensekey

sensekey — Returns the ASCII code of a key that has been pressed.

Description

Returns the ASCII code of a key that has been pressed, or -1 if no key has been pressed.

Syntax

```
kres[ , kkeydown] sensekey
```

Performance

kres - returns the ASCII value of a key which is pressed or released.

kkeydown - returns 1 if the key was pressed, 0 if it was released or if there is no key event.

kres can be used to read keyboard events from stdin and returns the ASCII value of any key that is pressed or released, or it returns -1 when there is no keyboard activity. The value of *kkeydown* is 1 when a key was pressed, or 0 otherwise. This behavior is emulated by default, so a key release is generated immediately after every key press. To have full functionality, FLTK can be used to capture keyboard events. *FLpanel* can be used to capture keyboard events and send them to the sensekey opcode, by adding an additional optional argument. See *FLpanel* for more information.



Note

This opcode can also be written as *sense*.

Examples

Here is an example of the sensekey opcode. It uses the file *sensekey.csd* [examples/sensekey.csd].

Example 462. Example of the sensekey opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o sensekey.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  k1 sensekey
  printk2 k1
endin
```

```
</CsInstruments>
<CsScore>

; Play Instrument #1 for thirty seconds.
i 1 0 30
e

</CsScore>
</CsoundSynthesizer>
```

Here is what the output should look like when the "q" button is pressed...

```
q i1 113.00000
```

Here is an example of the sensekey opcode in conjunction with *FLpanel*. It uses the file *FLpanel-sensekey.csd* [examples/FLpanel-sensekey.csd].

Example 463. Example of the sensekey opcode using keyboard capture from an FLpanel.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o FLpanel-sensekey.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
; Example by Johnathan Murphy

sr          = 44100
ksmps       = 128
nchnls      = 2

; ikbdcapture flag set to 1
ikey        init      1

gkasc, giasc FLpanel "sensekey", 740, 340, 100, 250, 2, ikey
               FLbutBank 2, 16, 8, 700, 300, 20, 20, -1
               FLpanelEnd
               FLrun

instr 1

kkey        sensekey
kprint      changed   kkey
               FLsetVal kprint, kkey, giasc

endin

</CsInstruments>
<CsScore>
i1 0 60
e
</CsScore>
</CsoundSynthesizer>
```

The lit button in the FLpanel window shows the last key pressed.

Here is a more complex example of the sensekey opcode in conjunction with *FLpanel*. It uses the file *FLpanel-sensekey2.csd* [examples/FLpanel-sensekey2.csd].

Example 464. Example of the sensekey opcode using keyboard capture from an FLpanel.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          ; -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o FLpanel-sensekey2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr = 48000
ksmps = 32
nchnls = 1
; Example by Istvan Varga
; if the FLTK opcodes are commented out, sensekey will read keyboard
; events from stdin
    FLpanel  "", 150, 50, 100, 100, 0, 1
    FLlabel 18, 10, 1, 0, 0, 0
    FLgroup  "Keyboard Input", 150, 50, 0, 0, 0
    FLgroupEnd
    FLpanelEnd

    FLrun

    instr 1

ktrig1 init 1
ktrig2 init 1
nxtKey1:
k1, k2 sensekey
    if (k1 != -1 || k2 != 0) then
        printf "Key code = %02X, state = %d\n", ktrig1, k1, k2
    ktrig1 = 3 - ktrig1
        kgoto nxtKey1
    endif
nxtKey2:
k3 sensekey
    if (k3 != -1) then
        printf "Character = '%c'\n", ktrig2, k3
    ktrig2 = 3 - ktrig2
        kgoto nxtKey2
    endif

    endin

</CsInstruments>
<CsScore>
i 1 0 3600
e
</CsScore>
</CsoundSynthesizer>
```

The console output will look something like:

```
new alloc for instr 1:
Key code = 65, state = 1
Character = 'e'
Key code = 65, state = 0
Key code = 72, state = 1
Character = 'r'
Key code = 72, state = 0
Key code = 61, state = 1
Character = 'a'
Key code = 61, state = 0
```

See also

FLpanel, FLkeyIn

Credits

Author: John ffitch
University of Bath, Codemist. Ltd.

Bath, UK
October 2000

Examples written by Kevin Conder, Johnathan Murphy and Istvan Varga.

New in Csound version 4.09. Renamed in Csound version 4.10.

seqtime

seqtime — Generates a trigger signal according to the values stored in a table.

Description

Generates a trigger signal according to the values stored in a table.

Syntax

```
ktrig_out seqtime ktime_unit, kstart, kloop, kinitndx, kfn_times
```

Performance

ktrig_out -- output trigger signal

ktime_unit -- unit of measure of time, related to seconds.

kstart -- start index of looped section

kloop -- end index of looped section

kinitndx -- initial index



Note

Although *kinitndx* is listed as k-rate, it is in fact accessed only at init-time. So if you are using a k-rate argument, it must be assigned with *init*.

kfn_times -- number of table containing a sequence of times

This opcode handles timed-sequences of groups of values stored into a table.

seqtime generates a trigger signal (a sequence of impulses, see also *trigger* opcode), according to the values stored in the *kfn_times* table. This table should contain a series of delta-times (i.e. times between to adjacent events). The time units stored into table are expressed in seconds, but can be rescaled by means of *ktime_unit* argument. The table can be filled with *GEN02* or by means of an external text-file containing numbers, with *GEN23*.



Note

Note that the *kloop* index marks the loop boundary and is NOT included in the looped elements. If you want to loop the first four elements, you would set *kstart* to 0 and *kloop* to 4.

It is possible to start the sequence from a value different than the first, by assigning to *kinitndx* an index different than zero (which corresponds to the first value of the table). Normally the sequence is looped, and the start and end of loop can be adjusted by modifying *kstart* and *kloop* arguments. User must be sure that values of these arguments (as well as *kinitndx*) correspond to valid table numbers, otherwise Csound will crash (because no range-checking is implemented).

It is possible to disable loop (one-shot mode) by assigning the same value both to *kstart* and *kloop* arguments. In this case, the last read element will be the one corresponding to the value of such arguments. Table can be read backward by assigning a negative *kloop* value. It is possible to trigger two events almost at the same time (actually separated by a k-cycle) by giving a zero value to the corresponding delta-time. First element contained in the table should be zero, if the user intends to

send a trigger impulse, it should come immediately after the orchestra instrument containing *seqtime* opcode.

Examples

Here is an example of the *seqtime* opcode. It uses the file *seqtime.csd* [examples/seqtime.csd].

Example 465. Example of the *seqtime* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o seqtime.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 64
nchnls = 1

; By Tim Mortimer and Andres Cabrera 2007

0dbfs = 1

gisine      ftgen      0, 0, 8192, 10,      1
;;; table defining an integer pitch set
gipset      ftgen      0, 0, 4, -2, 8.00, 8.04, 8.07, 8.10
;;;DELTA times for seqtime
gidelta     ftgen      0, 0, 4, -2, .5, 1, .25, 1.25

instr 1
kndx init 0
ktrigger init 0

ktime_unit init 1
kstart init p4
kloop init p5
kinitndx init 0
kfn_times init gidelta

ktrigger seqtime ktime_unit, kstart, kloop, kinitndx, kfn_times

printk2 ktrigger

if (ktrigger > 0) then
  kpitch table kndx, gipset
  event "i", 2, 0, 1, kpitch
  kndx = kndx + 1
  kndx = kndx % kloop
endif

endin

instr 2
icps = cpspch (p4)
a1 buzz 1, icps, 7, gisine
aamp expseg 0.00003, .02, 1, p3-.02, 0.00003

a1 = a1 * aamp * 0.5

out a1
endin

</CsInstruments>
<CsScore>
; start dur kstart kloop
i 1 0 7 0 4
i 1 8 10 0 3
i 1 19 10 4 4
```

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

GEN02, GEN23, trigseq seqtime2

Credits

Author: Gabriel Maldonado

November 2002. Added a note about the *kinitndx* parameter, thanks to Rasmus Ekman.

New in version 4.06

Example by: Tim Mortimer and Andres Cabrera 2007

seqtime2

seqtime2 — Generates a trigger signal according to the values stored in a table.

Description

Generates a trigger signal according to the values stored in a table.

Syntax

```
ktrig_out seqtime2 ktrig_in, ktime_unit, kstart, kloop, kinitndx, kfn_times
```

Performance

ktrig_out -- output trigger signal

ktime_unit -- unit of measure of time, related to seconds.

ktime_in -- input trigger signal.

kstart -- start index of looped section

kloop -- end index of looped section

kinitndx -- initial index



Note

Although *kinitndx* is listed as k-rate, it is in fact accessed only at init-time. So if you are using a k-rate argument, it must be assigned with *init*.

kfn_times -- number of table containing a sequence of times

This opcode handles timed-sequences of groups of values stored into a table.

seqtime2 generates a trigger signal (a sequence of impulses, see also *trigger* opcode), according to the values stored in the *kfn_times* table. This table should contain a series of delta-times (i.e. times between to adjacent events). The time units stored into table are expressed in seconds, but can be rescaled by means of *ktime_unit* argument. The table can be filled with *GEN02* or by means of an external text-file containing numbers, with *GEN23*.

It is possible to start the sequence from a value different than the first, by assigning to *initndx* an index different than zero (which corresponds to the first value of the table). Normally the sequence is looped, and the start and end of loop can be adjusted by modifying *kstart* and *kloop* arguments. User must be sure that values of these arguments (as well as *initndx*) correspond to valid table numbers, otherwise Csound will crash (because no range-checking is implemented).

It is possible to disable loop (one-shot mode) by assigning the same value both to *kstart* and *kloop* arguments. In this case, the last read element will be the one corresponding to the value of such arguments. Table can be read backward by assigning a negative *kloop* value. It is possible to trigger two events almost at the same time (actually separated by a k-cycle) by giving a zero value to the corresponding delta-time. First element contained in the table should be zero, if the user intends to send a trigger impulse, it should come immediately after the orchestra instrument containing *seqtime2* opcode.

seqtime2 is similar to *seqtime*, the difference is that when *ktrig_in* contains a non-zero value, current index is reset to *kinitndx* value. *kinitndx* can be varied at performance time.

See Also

GEN02, GEN23, seqtime, trigseq, timedseq

Credits

Author: Gabriel Maldonado

setctrl

setctrl — Configurable slider controls for realtime user input.

Description

Configurable slider controls for realtime user input. Requires Winsound or TCL/TK. *setctrl* sets a slider to a specific value, or sets a minimum or maximum range.

Syntax

```
setctrl inum, ival, itype
```

Initialization

inum -- number of the slider to set

ival -- value to be sent to the slider

itype -- type of value sent to the slider as follows:

- 1 -- set the current value. Initial value is 0.
- 2 -- set the minimum value. Default is 0.
- 3 -- set the maximum value. Default is 127.
- 4 -- set the label. (New in Csound version 4.09)

Performance

Calling *setctrl* will create a new slider on the screen. There is no theoretical limit to the number of sliders. Windows and TCL/TK use only integers for slider values, so the values may need rescaling. GUIs usually pass values at a fairly slow rate, so it may be advisable to pass the output of control through *port*.

Examples

Here is an example of the setctrl opcode. It uses the file *setctrl.csd* [examples/setctrl.csd].

Example 466. Example of the setctrl opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o setctrl.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
```

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Display the label "Volume" on Slider #1.
setctrl 1, "Volume", 4
; Set Slider #1's initial value to 20.
setctrl 1, 20, 1

; Capture and display the values for Slider #1.
k1 control 1
printk2 k1

; Play a simple oscillator.
; Use the values from Slider #1 for amplitude.
kamp = k1 * 128
a1 oscil kamp, 440, 1
out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for thirty seconds.
i 1 0 30
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
i1      38.00000
i1      40.00000
i1      43.00000
```

See Also

control

Credits

Author: John ffitch
University of Bath, Codemist. Ltd.
Bath, UK
May 2000

Example written by Kevin Conder.

New in Csound version 4.06

setksmps

setksmps — Sets the local ksmps value in a user-defined opcode block.

Description

Sets the local ksmps value in a user-defined opcode block.

The *setksmps* statement can be used to set the local *ksmps* value of the user-defined opcode block. It has one i-time parameter specifying the new *ksmps* value (which is left unchanged if zero is used). *setksmps* should be used before any other opcodes (but allowed after *xin*), otherwise unpredictable results may occur.

Syntax

```
setksmps iksmps
```

Initialization

iksmps -- sets the local ksmps value.

If *iksmps* is set to zero, the *ksmps* of the caller instrument or opcode is used (this is the default behavior).



Note

The local *ksmps* is implemented by splitting up a control period into smaller sub-periods and temporarily modifying internal Csound global variables. This also requires converting the rate of k-rate input and output arguments (input variables receive the same value in all sub-kperiods, while outputs are written only in the last one). It also means that you cannot use a local *ksmps* that is higher than the global *ksmps*.



Warning about local ksmps

When the local *ksmps* is not the same as the orchestra level *ksmps* value (as specified in the orchestra header). Global a-rate operations must not be used in the user-defined opcode block.

These include:

- any access to “ga” variables
- a-rate zak opcodes (*zar*, *zaw*, etc.)
- *tablera* and *tablewa* (these two opcodes may in fact work, but caution is needed)
- The *in* and *out* opcode family (these read from, and write to global a-rate buffers)

In general, the local *ksmps* should be used with care as it is an experimental feature. Though it works correctly in most cases.

The *setksmps* statement can be used to set the local *ksmps* value of the user-defined opcode block. It has one i-time parameter specifying the new *ksmps* value (which is left unchanged if zero is used). *setksmps* should be used before any other opcodes (but allowed after *xin*), otherwise unpredictable

results may occur.

Performance

The syntax of a user-defined opcode block is as follows:

```
opcode name, outtypes, intypes
xinarg1 [, xinarg2] [, xinarg3] ... [xinargN] xin
[setksmps iksmps]
... the rest of the instrument's code.
xout xoutarg1 [, xoutarg2] [, xoutarg3] ... [xoutargN]
endop
```

The new opcode can then be used with the usual syntax:

```
[xinarg1] [, xinarg2] ... [xinargN] name [xoutarg1] [, xoutarg2] ... [xoutargN] [, iksmps]
```

Examples

See the example for the *opcode* opcode.

See Also

endop, *opcode*, *xin*, *xout*

Credits

Author: Istvan Varga, 2002; based on code by Matt J. Ingalls

New in version 4.22

setscorepos

setscorepos — Sets the playback position of the current score performance to a given position.

Description

Sets the playback position of the current score performance to a given position.

Syntax

```
setscorepos ipos
```

Initialization

ipos -- playback position in seconds.

Examples

Here is an example of the setscorepos opcode.

Example 467. Example of the setscorepos opcode.

```
instr 1
  setscorepos 10
endin
```

See Also

setscorepos,

Credits

Author: Victor Lazzarini;
2008

New in Csound version 5.09

sfilist

`sfilist` — Prints a list of all instruments of a previously loaded SoundFont2 (SF2) file.

Description

Prints a list of all instruments of a previously loaded SoundFont2 (SF2) sample file. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
sfilist ifilhandle
```

Initialization

ifilhandle -- unique number generated by *sfload* opcode to be used as an identifier for a SF2 file. Several SF2 files can be loaded and activated at the same time.

Performance

sfilist prints a list of all instruments of a previously loaded SF2 file to the console.

These opcodes only support the sample structure of SF2 files. The modulator structure of the SoundFont2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfinstr, *sfinstrm*, *sfload*, *sfpassign*, *sfplay*, *sfplaym*, *sfplist*, *sfpreset*

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

sfinstr

sfinstr — Plays a SoundFont2 (SF2) sample instrument, generating a stereo sound.

Description

Plays a SoundFont2 (SF2) sample instrument, generating a stereo sound. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
ar1, ar2 sfinstr ivel, inotenum, xamp, xfreq, instrnum, ifilhandle \  
[, iflag] [, ioffset]
```

Initialization

ivel -- velocity value

inotenum -- MIDI note number value

instrnum -- number of an instrument of a SF2 file.

ifilhandle -- unique number generated by *sfload* opcode to be used as an identifier for a SF2 file. Several SF2 files can be loaded and activated at the same time.

iflag (optional) -- flag regarding the behavior of *xfreq* and *inotenum*

ioffset (optional) -- start playing at offset, in samples.

Performance

xamp -- amplitude correction factor

xfreq -- frequency value or frequency multiplier, depending by *iflag*. When *iflag* = 0, *xfreq* is a multiplier of a the default frequency, assigned by SF2 preset to the *inotenum* value. When *iflag* = 1, *xfreq* is the absolute frequency of the output sound, in Hz. Default is 0.

When *iflag* = 0, *inotenum* sets the frequency of the output according to the MIDI note number used, and *xfreq* is used as a multiplier. When *iflag* = 1, the frequency of the output, is set directly by *xfreq*. This allows the user to use any kind of micro-tuning based scales. However, this method is designed to work correctly only with presets tuned to the default equal temperament. Attempts to use this method with a preset already having non-standard tunings, or with drum-kit-based presets, could give unexpected results.

Adjustment of the amplitude can be done by varying the *xamp* argument, which acts as a multiplier.

The *ioffset* parameter allows the sound to start from a sample different than the first one. The user should make sure that its value is within the length of the specific sound. Otherwise, Csound will probably crash.

sfinstr plays an SF2 instrument instead of a preset (an SF2 instrument is the base of a preset layer). *instrnum* specifies the instrument number, and the user must be sure that the specified number belongs to an existing instrument of a determinate soundfont bank. Notice that both *xamp* and *xfreq* can operate at k-rate as well as a-rate, but both arguments must work at the same rate.

These opcodes only support the sample structure of SF2 files. The modulator structure of the Sound-

Font2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, sfinstrm, sfload, sfpassign, sfplay, sfplaym, sfplist, sfpreset

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

sfinstr3

sfinstr3 — Plays a SoundFont2 (SF2) sample instrument, generating a stereo sound with cubic interpolation.

Description

Plays a SoundFont2 (SF2) sample instrument, generating a stereo sound with cubic interpolation. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
ar1, ar2 sfinstr3 ivel, inotenum, xamp, xfreq, instrnum, ifilhandle \  
[, iflag] [, ioffset]
```

Initialization

ivel -- velocity value

inotenum -- MIDI note number value

instrnum -- number of an instrument of a SF2 file.

ifilhandle -- unique number generated by *sfload* opcode to be used as an identifier for a SF2 file. Several SF2 files can be loaded and activated at the same time.

iflag (optional) -- flag regarding the behavior of *xfreq* and *inotenum*

ioffset (optional) -- start playing at offset, in samples.

Performance

xamp -- amplitude correction factor

xfreq -- frequency value or frequency multiplier, depending by *iflag*. When *iflag* = 0, *xfreq* is a multiplier of a the default frequency, assigned by SF2 preset to the *inotenum* value. When *iflag* = 1, *xfreq* is the absolute frequency of the output sound, in Hz. Default is 0.

When *iflag* = 0, *inotenum* sets the frequency of the output according to the MIDI note number used, and *xfreq* is used as a multiplier. When *iflag* = 1, the frequency of the output, is set directly by *xfreq*. This allows the user to use any kind of micro-tuning based scales. However, this method is designed to work correctly only with presets tuned to the default equal temperament. Attempts to use this method with a preset already having non-standard tunings, or with drum-kit-based presets, could give unexpected results.

Adjustment of the amplitude can be done by varying the *xamp* argument, which acts as a multiplier.

The *ioffset* parameter allows the sound to start from a sample different than the first one. The user should make sure that its value is within the length of the specific sound. Otherwise, Csound will probably crash.

sfinstr3 is a cubic-interpolation version of *sfinstr*. Difference of sound-quality is noticeable specially in bass-frequency-transposed samples. In high-freq-transposed samples the difference is less noticeable, and I suggest to use linear-interpolation versions, because they are faster.

These opcodes only support the sample structure of SF2 files. The modulator structure of the Sound-

Font2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, sfinstr3m, sfinstrm, sfinstr, sfload, sfpassign, sfplay3, sfplay3m, sfplay, sfplaym, sfplist, sfpre-set

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

sfinstr3m

sfinstr3m — Plays a SoundFont2 (SF2) sample instrument, generating a mono sound with cubic interpolation.

Description

Plays a SoundFont2 (SF2) sample instrument, generating a mono sound with cubic interpolation. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
ares sfinstr3m ivel, inotenum, xamp, xfreq, instrnum, ifilhandle \  
    [, iflag] [, ioffset]
```

Initialization

ivel -- velocity value

inotenum -- MIDI note number value

instrnum -- number of an instrument of a SF2 file.

ifilhandle -- unique number generated by *sfload* opcode to be used as an identifier for a SF2 file. Several SF2 files can be loaded and activated at the same time.

iflag (optional) -- flag regarding the behavior of *xfreq* and *inotenum*

ioffset (optional) -- start playing at offset, in samples.

Performance

xamp -- amplitude correction factor

xfreq -- frequency value or frequency multiplier, depending by *iflag*. When *iflag* = 0, *xfreq* is a multiplier of a the default frequency, assigned by SF2 preset to the *inotenum* value. When *iflag* = 1, *xfreq* is the absolute frequency of the output sound, in Hz. Default is 0.

When *iflag* = 0, *inotenum* sets the frequency of the output according to the MIDI note number used, and *xfreq* is used as a multiplier. When *iflag* = 1, the frequency of the output, is set directly by *xfreq*. This allows the user to use any kind of micro-tuning based scales. However, this method is designed to work correctly only with presets tuned to the default equal temperament. Attempts to use this method with a preset already having non-standard tunings, or with drum-kit-based presets, could give unexpected results.

Adjustment of the amplitude can be done by varying the *xamp* argument, which acts as a multiplier.

The *ioffset* parameter allows the sound to start from a sample different than the first one. The user should make sure that its value is within the length of the specific sound. Otherwise, Csound will probably crash.

sfinstr3m is a cubic-interpolation version of *sfinstrm*. Difference of sound-quality is noticeable specially in bass-frequency-transposed samples. In high-freq-transposed samples the difference is less noticeable, and I suggest to use linear-interpolation versions, because they are faster.

These opcodes only support the sample structure of SF2 files. The modulator structure of the Sound-

Font2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, sfinstr3, sfinstr, sfinstrm, sfload, sfpassign, sfplay3, sfplay3m, sfplay, sfplaym, sfplist, sfpreset

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

sfinstrm

sfinstrm — Plays a SoundFont2 (SF2) sample instrument, generating a mono sound.

Description

Plays a SoundFont2 (SF2) sample instrument, generating a mono sound. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
ares sfinstrm ivel, inotenum, xamp, xfreq, instrnum, ifilhandle \  
    [, iflag] [, ioffset]
```

Initialization

ivel -- velocity value

inotenum -- MIDI note number value

instrnum -- number of an instrument of a SF2 file.

ifilhandle -- unique number generated by *sfload* opcode to be used as an identifier for a SF2 file. Several SF2 files can be loaded and activated at the same time.

iflag (optional) -- flag regarding the behavior of *xfreq* and *inotenum*

ioffset (optional) -- start playing at offset, in samples.

Performance

xamp -- amplitude correction factor

xfreq -- frequency value or frequency multiplier, depending by *iflag*. When *iflag* = 0, *xfreq* is a multiplier of a the default frequency, assigned by SF2 preset to the *inotenum* value. When *iflag* = 1, *xfreq* is the absolute frequency of the output sound, in Hz. Default is 0.

When *iflag* = 0, *inotenum* sets the frequency of the output according to the MIDI note number used, and *xfreq* is used as a multiplier. When *iflag* = 1, the frequency of the output, is set directly by *xfreq*. This allows the user to use any kind of micro-tuning based scales. However, this method is designed to work correctly only with presets tuned to the default equal temperament. Attempts to use this method with a preset already having non-standard tunings, or with drum-kit-based presets, could give unexpected results.

Adjustment of the amplitude can be done by varying the *xamp* argument, which acts as a multiplier.

The *ioffset* parameter allows the sound to start from a sample different than the first one. The user should make sure that its value is within the length of the specific sound. Otherwise, Csound will probably crash.

sfinstrm plays is a mono version of *sfinstr*. This is the fastest opcode of the SF2 family.

These opcodes only support the sample structure of SF2 files. The modulator structure of the SoundFont2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, sfinstr, sfload, sfpassign, sfplay, sfplaym, sfplist, sfpreset

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

sfload

sfload — Loads an entire SoundFont2 (SF2) sample file into memory.

Description

Loads an entire SoundFont2 (SF2) sample file into memory. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *Sound-Font2 File Format Appendix*.

sfload should be placed in the header section of a Csound orchestra.

Syntax

```
ir sfload "filename"
```

Initialization

ir -- output to be used by other SF2 opcodes. For *sfload*, *ir* is *ifilhandle*.

"filename" -- name of the SF2 file, with its complete path. It must be a string typed within double-quotes with "/" to separate directories (this applies to DOS and Windows as well, where using a backslash will generate an error), or an integer that has been the subject of a *strset* operation

Performance

sfload loads an entire SF2 file into memory. It returns a file handle to be used by other opcodes. Several instances of *sfload* can be placed in the header section of an orchestra, allowing use of more than one SF2 file in a single orchestra.

These opcodes only support the sample structure of SF2 files. The modulator structure of the Sound-Font2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, *sfinstr*, *sfinstrm*, *sfpassign*, *sfplay*, *sfplaym*, *sfplist*, *sfpreset*

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

sflooper

sflooper — Plays a SoundFont2 (SF2) sample preset, generating a stereo sound, with user-defined time-varying crossfade looping.

Description

Plays a SoundFont2 (SF2) sample preset, generating a stereo sound, similarly to `sfplay`. Unlike that opcode, though, it ignores the looping points set in the SF2 file and substitutes them for a user-defined crossfade loop. It is a cross between `sfplay` and `flooper2`.

Syntax

```
ar1, ar2 sflooper ivel, inotenum, kamp, kpitch, ipreindex, kloopstart, kloopend, kcrossfade, ifn \
[, istart, imode, ifenv, iskip]
```

Initialization

ivel -- velocity value

inotenum -- MIDI note number value

ipreindex -- preset index

istart -- playback start pos in seconds

imode -- loop modes: 0 forward, 1 backward, 2 back-and-forth [def: 0]

ifenv -- if non-zero, crossfade envelope shape table number. The default, 0, sets the crossfade to linear.

iskip -- if 1, the opcode initialisation is skipped, for tied notes, performance continues from the position in the loop where the previous note stopped. The default, 0, does not skip initialisation

Performance

kamp -- amplitude scaling

kpitch -- pitch control (transposition ratio); negative values are not allowed.

kloopstart -- loop start point (secs). Note that although k-rate, loop parameters such as this are only updated once per loop cycle. If loop start is set beyond the end of the sample, no looping will result.

kloopend -- loop end point (secs), updated once per loop cycle.

kcrossfade -- crossfade length (secs), updated once per loop cycle and limited to loop length.

sflooper plays a preset, generating a stereo sound.

These opcodes only support the sample structure of SF2 files. The modulator structure of the SoundFont2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, *sfinstr*, *sfinstrm*, *sfload*, *sfpassign*, *sfplaym*, *sfplist*, *sfpreset*

Credits

Author: Victor Lazzarini;
August 2007

New in Csound Version 5.07

sfpassign

sfpassign — Assigns all presets of a SoundFont2 (SF2) sample file to a sequence of progressive index numbers.

Description

Assigns all presets of a previously loaded SoundFont2 (SF2) sample file to a sequence of progressive index numbers. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

sfpassign should be placed in the header section of a Csound orchestra.

Syntax

```
sfpassign istartindex, ifilhandle[, imsgs]
```

Initialization

istartindex -- starting index preset by the user in bulk preset assignments.

ifilhandle -- unique number generated by *sfload* opcode to be used as an identifier for a SF2 file. Several SF2 files can be loaded and activated at the same time.

imsgs -- if non-zero messages are suppressed.

Performance

sfpassign assigns all presets of a previously loaded SF2 file to a sequence of progressive index numbers, to be used later with the opcodes *sfplay* and *sfplaym*. *istartindex* specifies the starting index number. Any number of *sfpassign* instances can be placed in the header section of an orchestra, each one assigning presets belonging to different SF2 files. The user must take care that preset index numbers of different SF2 files do not overlap.

These opcodes only support the sample structure of SF2 files. The modulator structure of the SoundFont2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, *sfinstr*, *sfinstrm*, *sfload*, *sfplay*, *sfplaym*, *sfplist*, *sfpreset*

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

sfplay

sfplay — Plays a SoundFont2 (SF2) sample preset, generating a stereo sound.

Description

Plays a SoundFont2 (SF2) sample preset, generating a stereo sound. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
ar1, ar2 sfplay ivel, inotenum, xamp, xfreq, ipreindex [, iflag] [, ioffset] [, ienv]
```

Initialization

ivel -- velocity value

inotenum -- MIDI note number value

ipreindex -- preset index

iflag (optional) -- flag regarding the behavior of *xfreq* and *inotenum*

ioffset (optional) -- start playing at offset, in samples.

ienv (optional) -- enables and determines amplitude envelope amplitude envelope. 0 = no envelope, 1 = linear attack and decay, 2 = linear attack, exponential decay (see below). Default = 0.

Performance

xamp -- amplitude correction factor

xfreq -- frequency value or frequency multiplier, depending by *iflag*. When *iflag* = 0, *xfreq* is a multiplier of a the default frequency, assigned by SF2 preset to the *inotenum* value. When *iflag* = 1, *xfreq* is the absolute frequency of the output sound, in Hz. Default is 0.

When *iflag* = 0, *inotenum* sets the frequency of the output according to the MIDI note number used, and *xfreq* is used as a multiplier. When *iflag* = 1, the frequency of the output, is set directly by *xfreq*. This allows the user to use any kind of micro-tuning based scales. However, this method is designed to work correctly only with presets tuned to the default equal temperament. Attempts to use this method with a preset already having non-standard tunings, or with drum-kit-based presets, could give unexpected results.

Adjustment of the amplitude can be done by varying the *xamp* argument, which acts as a multiplier.

Notice that both *xamp* and *xfreq* can use k-rate as well as a-rate signals. Both arguments must use variables of the same rate, or *sfplay* will not work correctly. *ipreindex* must contain the number of a previously assigned preset, or Csound will crash.

The *ioffset* parameter allows the sound to start from a sample different than the first one. The user should make sure that its value is within the length of the specific sound. Otherwise, Csound will probably crash.

The *ienv* parameter enables and determines the type of amplitude envelope used. The default value is 0, or no envelope. If *ienv* is set to 1, the attack and decay portions are linear. If set to 2, the attack is linear and the decay is exponential. The release portion of the envelope has not yet been imple-

mented.

sfplay plays a preset, generating a stereo sound. *ivel* does not directly affect the amplitude of the output, but informs *sfplay* about which sample should be chosen in multi-sample, velocity-split presets.

These opcodes only support the sample structure of SF2 files. The modulator structure of the SoundFont2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, *sfinstr*, *sfinstrm*, *sfload*, *sfpassign*, *sfplaym*, *sfplaym*, *sfplaym*, *sfplist*, *sfpreset*

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

New optional parameter *ienv* in version 5.09

sfplay3

sfplay3 — Plays a SoundFont2 (SF2) sample preset, generating a stereo sound with cubic interpolation.

Description

Plays a SoundFont2 (SF2) sample preset, generating a stereo sound with cubic interpolation. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
ar1, ar2 sfplay3 ivel, inotenum, xamp, xfreq, ipreindex [, iflag] [, ioffset] [, ienv]
```

Initialization

ivel -- velocity value

inotenum -- MIDI note number value

ipreindex -- preset index

iflag -- flag regarding the behavior of *xfreq* and *inotenum*

ioffset (optional) -- start playing at offset, in samples.

ienv (optional) -- enables and determines amplitude envelope. 0 = no envelope, 1 = linear attack and decay, 2 = linear attack, exponential decay (see below). Default = 0.

Performance

xamp -- amplitude correction factor

xfreq -- frequency value or frequency multiplier, depending by *iflag*. When *iflag* = 0, *xfreq* is a multiplier of a the default frequency, assigned by SF2 preset to the *inotenum* value. When *iflag* = 1, *xfreq* is the absolute frequency of the output sound, in Hz. Default is 0.

When *iflag* = 0, *inotenum* sets the frequency of the output according to the MIDI note number used, and *xfreq* is used as a multiplier. When *iflag* = 1, the frequency of the output, is set directly by *xfreq*. This allows the user to use any kind of micro-tuning based scales. However, this method is designed to work correctly only with presets tuned to the default equal temperament. Attempts to use this method with a preset already having non-standard tunings, or with drum-kit-based presets, could give unexpected results.

Adjustment of the amplitude can be done by varying the *xamp* argument, which acts as a multiplier.

Notice that both *xamp* and *xfreq* can use k-rate as well as a-rate signals. Both arguments must use variables of the same rate, or *sfplay3* will not work correctly. *ipreindex* must contain the number of a previously assigned preset, or Csound will crash.

The *ioffset* parameter allows the sound to start from a sample different than the first one. The user should make sure that its value is within the length of the specific sound. Otherwise, Csound will probably crash.

The *ienv* parameter enables and determines the type of amplitude envelope used. The default value is 0, or no envelope. If *ienv* is set to 1, the attack and decay portions are linear. If set to 2, the attack

is linear and the decay is exponential. The release portion of the envelope has not yet been implemented.

sfplay3 plays a preset, generating a stereo sound with cubic interpolation. *ivel* does not directly affect the amplitude of the output, but informs *sfplay3* about which sample should be chosen in multi-sample, velocity-split presets.

sfplay3 is a cubic-interpolation version of *sfplay*. Difference of sound-quality is noticeable specially in bass-frequency-transposed samples. In high-freq-transposed samples the difference is less noticeable, and I suggest to use linear-interpolation versions, because they are faster.

These opcodes only support the sample structure of SF2 files. The modulator structure of the SoundFont2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, *sfinstr3*, *sfinstr3m*, *sfinstr*, *sfinstrm*, *sfload*, *sfpassign*, *sfplay3m*, *sfplaym*, *sfplay*, *sfplist*, *sfpreset*

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

New optional parameter *ienv* in version 5.09

sfplay3m

sfplay3m — Plays a SoundFont2 (SF2) sample preset, generating a mono sound with cubic interpolation.

Description

Plays a SoundFont2 (SF2) sample preset, generating a mono sound with cubic interpolation. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
ares sfplay3m ivel, inotenum, xamp, xfreq, ipreindex [, iflag] [, ioffset] [, ienv]
```

Initialization

ivel -- velocity value

inotenum -- MIDI note number value

ipreindex -- preset index

iflag (optional) -- flag regarding the behavior of *xfreq* and *inotenum*

ioffset (optional) -- start playing at offset, in samples.

ienv (optional) -- enables and determines amplitude envelope amplitude envelope. 0 = no envelope, 1 = linear attack and decay, 2 = linear attack, exponential decay (see below). Default = 0.

Performance

xamp -- amplitude correction factor

xfreq -- frequency value or frequency multiplier, depending by *iflag*. When *iflag* = 0, *xfreq* is a multiplier of a the default frequency, assigned by SF2 preset to the *inotenum* value. When *iflag* = 1, *xfreq* is the absolute frequency of the output sound, in Hz. Default is 0.

When *iflag* = 0, *inotenum* sets the frequency of the output according to the MIDI note number used, and *xfreq* is used as a multiplier. When *iflag* = 1, the frequency of the output, is set directly by *xfreq*. This allows the user to use any kind of micro-tuning based scales. However, this method is designed to work correctly only with presets tuned to the default equal temperament. Attempts to use this method with a preset already having non-standard tunings, or with drum-kit-based presets, could give unexpected results.

Adjustment of the amplitude can be done by varying the *xamp* argument, which acts as a multiplier.

Notice that both *xamp* and *xfreq* can use k-rate as well as a-rate signals. Both arguments must use variables of the same rate, or *sfplay3m* will not work correctly. *ipreindex* must contain the number of a previously assigned preset, or Csound will crash.

The *ioffset* parameter allows the sound to start from a sample different than the first one. The user should make sure that its value is within the length of the specific sound. Otherwise, Csound will probably crash.

The *ienv* parameter enables and determines the type of amplitude envelope used. The default value is 0, or no envelope. If *ienv* is set to 1, the attack and decay portions are linear. If set to 2, the attack

is linear and the decay is exponential. The release portion of the envelope has not yet been implemented.

sfplay3m is a mono version of *sfplay3*. It should be used with mono preset, or with the stereo presets in which stereo output is not required. It is faster than *sfplay3*.

sfplay3m is also a cubic-interpolation version of *sfplaym*. Difference of sound-quality is noticeable specially in bass-frequency-transposed samples. In high-freq-transposed samples the difference is less noticeable, and I suggest to use linear-interpolation versions, because they are faster.

These opcodes only support the sample structure of SF2 files. The modulator structure of the Sound-Font2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, *sfinstr3*, *sfinstr3m*, *sfinstr*, *sfinstrm*, *sfload*, *sfpassign*, *sfplay3*, *sfplaym*, *sfplay*, *sfplist*, *sfpreset*

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

New optional parameter *ienv* in version 5.09

sfplaym

sfplaym — Plays a SoundFont2 (SF2) sample preset, generating a mono sound.

Description

Plays a SoundFont2 (SF2) sample preset, generating a mono sound. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
ares sfplaym ivel, inotenum, xamp, xfreq, ipreindex [, iflag] [, ioffset] [, ienv]
```

Initialization

ivel -- velocity value

inotenum -- MIDI note number value

ipreindex -- preset index

iflag (optional) -- flag regarding the behavior of *xfreq* and *inotenum*

ioffset (optional) -- start playing at offset, in samples.

ienv (optional) -- enables and determines amplitude envelope amplitude envelope. 0 = no envelope, 1 = linear attack and decay, 2 = linear attack, exponential decay (see below). Default = 0.

Performance

xamp -- amplitude correction factor

xfreq -- frequency value or frequency multiplier, depending by *iflag*. When *iflag* = 0, *xfreq* is a multiplier of a the default frequency, assigned by SF2 preset to the *inotenum* value. When *iflag* = 1, *xfreq* is the absolute frequency of the output sound, in Hz. Default is 0.

When *iflag* = 0, *inotenum* sets the frequency of the output according to the MIDI note number used, and *xfreq* is used as a multiplier. When *iflag* = 1, the frequency of the output, is set directly by *xfreq*. This allows the user to use any kind of micro-tuning based scales. However, this method is designed to work correctly only with presets tuned to the default equal temperament. Attempts to use this method with a preset already having non-standard tunings, or with drum-kit-based presets, could give unexpected results.

Adjustment of the amplitude can be done by varying the *xamp* argument, which acts as a multiplier.

Notice that both *xamp* and *xfreq* can use k-rate as well as a-rate signals. Both arguments must use variables of the same rate, or *sfplaym* will not work correctly. *ipreindex* must contain the number of a previously assigned preset, or Csound will crash.

The *ioffset* parameter allows the sound to start from a sample different than the first one. The user should make sure that its value is within the length of the specific sound. Otherwise, Csound will probably crash.

The *ienv* parameter enables and determines the type of amplitude envelope used. The default value is 0, or no envelope. If *ienv* is set to 1, the attack and decay portions are linear. If set to 2, the attack is linear and the decay is exponential. The release portion of the envelope has not yet been imple-

mented.

sfplaym is a mono version of *sfplay*. It should be used with mono preset, or with the stereo presets in which stereo output is not required. It is faster than *sfplay*.

These opcodes only support the sample structure of SF2 files. The modulator structure of the Sound-Font2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, *sfinstr*, *sfinstrm*, *sfload*, *sfpassign*, *sfplay*, *sfplay*, *sfplay*, *sfplist*, *sfpreset*

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

New optional parameter *ienv* in version 5.09

sfplist

sfplist — Prints a list of all presets of a SoundFont2 (SF2) sample file.

Description

Prints a list of all presets of a previously loaded SoundFont2 (SF2) sample file. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

Syntax

```
sfplist ifilhandle
```

Initialization

ifilhandle -- unique number generated by *sfload* opcode to be used as an identifier for a SF2 file. Several SF2 files can be loaded and activated at the same time.

Performance

sfplist prints a list of all presets of a previously loaded SF2 file to the console.

These opcodes only support the sample structure of SF2 files. The modulator structure of the SoundFont2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, *sfinstr*, *sfinstrm*, *sfload*, *sfpassign*, *sfplay*, *sfplaym*, *sfpreset*

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

sfpreset

sfpreset — Assigns an existing preset of a SoundFont2 (SF2) sample file to an index number.

Description

Assigns an existing preset of a previously loaded SoundFont2 (SF2) sample file to an index number. These opcodes allow management the sample-structure of SF2 files. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format can be found in the *SoundFont2 File Format Appendix*.

sfpreset should be placed in the header section of a Csound orchestra.

Syntax

```
ir sfpreset iprog, ibank, ifilhandle, ipreindex
```

Initialization

ir -- output to be used by other SF2 opcodes. For *sfpreset*, *ir* is *ipreindex*.

iprog -- program number of a bank of presets in a SF2 file

ibank -- number of a specific bank of a SF2 file

ifilhandle -- unique number generated by *sfload* opcode to be used as an identifier for a SF2 file. Several SF2 files can be loaded and activated at the same time.

ipreindex -- preset index

Performance

sfpreset assigns an existing preset of a previously loaded SF2 file to an index number, to be used later with the opcodes *sfplay* and *sfplaym*. The user must previously know the program and the bank numbers of the preset in order to fill the corresponding arguments. Any number of *sfpreset* instances can be placed in the header section of an orchestra, each one assigning a different preset belonging to the same (or different) SF2 file to different index numbers.

These opcodes only support the sample structure of SF2 files. The modulator structure of the SoundFont2 format is not supported in Csound. Any modulation or processing to the sample data is left to the Csound user, bypassing all restrictions forced by the SF2 standard.

See Also

sfilist, *sfinstr*, *sfinstrm*, *sfload*, *sfpassign*, *sfplay*, *sfplaym*, *sfplist*

Credits

Author: Gabriel Maldonado
Italy
May 2000

New in Csound Version 4.07

shaker

shaker — Sounds like the shaking of a maraca or similar gourd instrument.

Description

Audio output is a tone related to the shaking of a maraca or similar gourd instrument. The method is a physically inspired model developed from Perry Cook, but re-coded for Csound.

Syntax

```
ares shaker kamp, kfreq, kbeans, kdamp, ktimes [, idecay]
```

Initialization

idecay -- If present indicates for how long at the end of the note the shaker is to be damped. The default value is zero.

Performance

A note is played on a maraca-like instrument, with the arguments as below.

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kbeans -- The number of beans in the gourd. A value of 8 seems suitable.

kdamp -- The damping value of the shaker. Values of 0.98 to 1 seems suitable, with 0.99 a reasonable default.

ktimes -- Number of times shaken.



Note

The argument *knum* was redundant, so it was removed in version 3.49.

Examples

Here is an example of the shaker opcode. It uses the file *shaker.csd* [examples/shaker.csd].

Example 468. Example of the shaker opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o shaker.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```

```
; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1
instr 1
  a1 shaker 10000, 440, 8, 0.999, 100, 0
  out a1
endin

</CsInstruments>
<CsScore>

i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

Fixed the example thanks to a message from Istvan Varga.

sin

sin — Performs a sine function.

Description

Returns the sine of x (x in radians).

Syntax

`sin(x)` (no rate restriction)

Examples

Here is an example of the sin opcode. It uses the file *sin.csd* [examples/sin.csd].

Example 469. Example of the sin opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o sin.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  irad = 25
  i1 = sin(irad)

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = -0.132
```

See Also

cos, cosh, cosinv, sinh, sininv, tan, tanh, taninv

Credits

Example written by Kevin Conder.

sinh

sinh — Performs a hyperbolic sine function.

Description

Returns the hyperbolic sine of x (x in radians).

Syntax

`sinh(x)` (no rate restriction)

Examples

Here is an example of the sinh opcode. It uses the file *sinh.csd* [examples/sinh.csd].

Example 470. Example of the sinh opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o sinh.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  irad = 1
  i1 = sinh(irad)

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should a line like this:

```
instr 1:  i1 = 1.175
```

See Also

cos, cosh, cosinv, sin, sininv, tan, tanh, taninv

Credits

Example written by Kevin Conder.

New in version 3.47

sininv

sininv — Performs an arcsine function.

Description

Returns the arcsine of x (x in radians).

Syntax

`sininv(x)` (no rate restriction)

Examples

Here is an example of the sininv opcode. It uses the file *sininv.csd* [examples/sininv.csd].

Example 471. Example of the sininv opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o sininv.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  irad = 0.5
  i1 = sininv(irad)

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 0.524
```

See Also

cos, cosh, cosinv, sin, sinh, tan, tanh, taninv

Credits

Author: John ffitch

New in version 3.48

Example written by Kevin Conder.

sinsyn

sinsyn — Streaming partial track additive synthesis with cubic phase interpolation

Description

The sinsyn opcode takes an input containing a TRACKS pv streaming signal (as generated, for instance by the partials opcode). It sinsynthesises the signal using linear amplitude and cubic phase interpolation to drive a bank of interpolating oscillators with amplitude and pitch scaling controls. Sinsyn attempts to preserve the phase of the partials in the original signal and in so doing it does not allow for pitch or timescale modifications of the signal.

Syntax

```
asig sinsyn fin, kscal, kmaxtracks, ifn
```

Performance

asig -- output audio rate signal

fin -- input pv stream in TRACKS format

kscal -- amplitude scaling

kmaxtracks -- max number of tracks in sinsynthesis. Limiting this will cause a non-linear filtering effect, by discarding newer and higher-frequency tracks (tracks are ordered by start time and ascending frequency, respectively)

ifn -- function table containing one cycle of a sinusoid (sine or cosine)

Examples

Example 472. Example

```
ain inch 1 ; input signal
fsl,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fsl,fsi2,.003,1,3,500 ; partial tracking
aout sinsyn fst, 1, 1.5, 500, 1 ; resynthesis (up a 5th)
out aout
```

The example above shows partial tracking of an ifd-analysis signal and cubic-phase additive resynthesis.

Credits

Author: Victor Lazzarini;
June 2005

New plugin in version 5

November 2004.

sleighbells

sleighbells — Semi-physical model of a sleighbell sound.

Description

sleighbells is a semi-physical model of a sleighbell sound. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares sleighbells kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] \  
    [, ifreq1] [, ifreq2]
```

Initialization

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 32.

idamp (optional) -- the damping factor, as part of this equation:

$\text{damping_amount} = 0.9994 + (\text{idamp} * 0.002)$

The default *damping_amount* is 0.9994 which means that the default value of *idamp* is 0. The maximum *damping_amount* is 1.0 (no damping). This means the maximum value for *idamp* is 0.03.

The recommended range for *idamp* is usually below 75% of the maximum value.

imaxshake (optional, default=0) -- amount of energy to add back into the system. The value should be in range 0 to 1.

ifreq (optional) -- the main resonant frequency. The default value is 2500.

ifreq1 (optional) -- the first resonant frequency. The default value is 5300.

ifreq2 (optional) -- the second resonant frequency. The default value is 6500.

Performance

kamp -- Amplitude of output. Note: As these instruments are stochastic, this is only an approximation.

Examples

Here is an example of the sleighbells opcode. It uses the file *sleighbells.csd* [examples/sleighbells.csd].

Example 473. Example of the sleighbells opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.


```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o sleighbells.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1: An example of sleighbells.
instr 1
  al sleighbells 20000, 0.01

  out a1
endin

</CsInstruments>
<CsScore>

i 1 0.00 0.25
i 1 0.30 0.25
i 1 0.60 0.25
i 1 0.90 0.25
i 1 1.20 0.25
i 1 1.50 0.25
i 1 1.80 0.25
i 1 2.10 0.25
i 1 2.40 0.25
i 1 2.70 0.25
i 1 3.00 0.25
e

</CsScore>
</CsoundSynthesizer>
```

See Also

bamboo, dripwater, guiro, tambourine

Credits

Author: Perry Cook, part of the PhISEM (Physically Informed Stochastic Event Modeling)
Adapted by John ffitch
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

slider16

slider16 — Creates a bank of 16 different MIDI control message numbers.

Description

Creates a bank of 16 different MIDI control message numbers.

Syntax

```
i1,...,i16 slider16 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \  
            ictlnum16, imin16, imax16, init16, ifn16  
  
k1,...,k16 slider16 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \  
            ictlnum16, imin16, imax16, init16, ifn16
```

Initialization

i1 ... i16 -- output values

ichan -- MIDI channel (1-16)

ictlnum1 ... ictlnum16 -- MIDI control number (0-127)

imin1 ... imin16 -- minimum values for each controller

imax1 ... imax16 -- maximum values for each controller

init1 ... init16 -- initial value for each controller

ifn1 ... ifn16 -- function table for conversion for each controller

icutoff1 ... icutoff16 -- low-pass filter cutoff frequency for each controller

Performance

k1 ... k16 -- output values

slider16 is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider16 allows a bank of 16 different MIDI control message numbers.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

In the i-rate version of *slider16*, there is not an initial value input argument, because the output is gotten directly from current status of internal controller array of Csound.

See Also

s16b14, s32b14, slider16f, slider32, slider32f, slider64, slider64f, slider8, slider8f

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

slider16f

slider16f — Creates a bank of 16 different MIDI control message numbers, filtered before output.

Description

Creates a bank of 16 different MIDI control message numbers, filtered before output.

Syntax

```
k1,...,k16 slider16f ichan, ictlnum1, imin1, imax1, init1, ifn1, \  
            icutoff1,..., ictlnum16, imin16, imax16, init16, ifn16, icutoff16
```

Initialization

ichan -- MIDI channel (1-16)

ictlnum1 ... *ictlnum16* -- MIDI control number (0-127)

imin1 ... *imin16* -- minimum values for each controller

imax1 ... *imax16* -- maximum values for each controller

init1 ... *init16* -- initial value for each controller

ifn1 ... *ifn16* -- function table for conversion for each controller

icutoff1 ... *icutoff16* -- low-pass filter cutoff frequency for each controller

Performance

k1 ... *k16* -- output values

slider16f is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider16f allows a bank of 16 different MIDI control message numbers. It filters the signal before output. This eliminates discontinuities due to the low resolution of the MIDI (7 bit). The cutoff frequency can be set separately for each controller (suggested range: .1 to 5 Hz).

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.



Warning

slider16f does not output the required initial value immediately, but only after some k-cycles because the filter slightly delays the output.

See Also

s16b14, s32b14, slider16, slider32, slider32f, slider64, slider64f, slider8, slider8f

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

slider32

slider32 — Creates a bank of 32 different MIDI control message numbers.

Description

Creates a bank of 32 different MIDI control message numbers.

Syntax

```
i1,...,i32 slider32 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \  
            ictlnum32, imin32, imax32, init32, ifn32  
  
k1,...,k32 slider32 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \  
            ictlnum32, imin32, imax32, init32, ifn32
```

Initialization

i1 ... i32 -- output values

ichan -- MIDI channel (1-16)

ictlnum1 ... ictlnum32 -- MIDI control number (0-127)

imin1 ... imin32 -- minimum values for each controller

imax1 ... imax32 -- maximum values for each controller

init1 ... init32 -- initial value for each controller

ifn1 ... ifn32 -- function table for conversion for each controller

icutoff1 ... icutoff32 -- low-pass filter cutoff frequency for each controller

Performance

k1 ... k32 -- output values

slider32 is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider32 allows a bank of 32 different MIDI control message numbers.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

In the i-rate version of *slider32*, there is not an initial value input argument, because the output is gotten directly from current status of internal controller array of Csound.

See Also

s16b14, s32b14, slider16, slider16f, slider32f, slider64, slider64f, slider8, slider8f

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

slider32f

slider32f — Creates a bank of 32 different MIDI control message numbers, filtered before output.

Description

Creates a bank of 32 different MIDI control message numbers, filtered before output.

Syntax

```
k1,...,k32 slider32f ichan, ictlnum1, imin1, imax1, init1, ifn1, icutoff1, \  
..., ictlnum32, imin32, imax32, init32, ifn32, icutoff32
```

Initialization

ichan -- MIDI channel (1-16)

ictlnum1 ... *ictlnum32* -- MIDI control number (0-127)

imin1 ... *imin32* -- minimum values for each controller

imax1 ... *imax32* -- maximum values for each controller

init1 ... *init32* -- initial value for each controller

ifn1 ... *ifn32* -- function table for conversion for each controller

icutoff1 ... *icutoff32* -- low-pass filter cutoff frequency for each controller

Performance

k1 ... *k32* -- output values

slider32f is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider32f allows a bank of 32 different MIDI control message numbers. It filters the signal before output. This eliminates discontinuities due to the low resolution of the MIDI (7 bit). The cutoff frequency can be set separately for each controller (suggested range: .1 to 5 Hz).

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl17* and *tonek*) when more controllers are required.



Warning

slider32f opcodes do not output the required initial value immediately, but only after some k-cycles because the filter slightly delays the output.

See Also

s16b14, s32b14, slider16, slider16f, slider32, slider64, slider64f, slider8, slider8f

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

slider64

slider64 — Creates a bank of 64 different MIDI control message numbers.

Description

Creates a bank of 64 different MIDI control message numbers.

Syntax

```
i1,...,i64 slider64 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \  
            ictlnum64, imin64, imax64, init64, ifn64  
  
k1,...,k64 slider64 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \  
            ictlnum64, imin64, imax64, init64, ifn64
```

Initialization

i1 ... i64 -- output values

ichan -- MIDI channel (1-16)

ictlnum1 ... ictlnum64 -- MIDI control number (0-127)

imin1 ... imin64 -- minimum values for each controller

imax1 ... imax64 -- maximum values for each controller

init1 ... init64 -- initial value for each controller

ifn1 ... ifn64 -- function table for conversion for each controller

icutoff1 ... icutoff64 -- low-pass filter cutoff frequency for each controller

Performance

k1 ... k64 -- output values

slider64 is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider64 allows a bank of 64 different MIDI control message numbers.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

In the i-rate version of *slider64*, there is not an initial value input argument, because the output is gotten directly from current status of internal controller array of Csound.

See Also

s16b14, s32b14, slider16, slider16f, slider32, slider32f, slider64f slider8, slider8f

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

slider64f

slider64f — Creates a bank of 64 different MIDI control message numbers, filtered before output.

Description

Creates a bank of 64 different MIDI control message numbers, filtered before output.

Syntax

```
k1,...,k64 slider64f ichan, ictlnum1, imin1, imax1, init1, ifn1, \  
            icutoff1,..., ictlnum64, imin64, imax64, init64, ifn64, icutoff64
```

Initialization

ichan -- MIDI channel (1-16)

ictlnum1 ... *ictlnum64* -- MIDI control number (0-127)

imin1 ... *imin64* -- minimum values for each controller

imax1 ... *imax64* -- maximum values for each controller

init1 ... *init64* -- initial value for each controller

ifn1 ... *ifn64* -- function table for conversion for each controller

icutoff1 ... *icutoff64* -- low-pass filter cutoff frequency for each controller

Performance

k1 ... *k64* -- output values

slider64f is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider64f allows a bank of 64 different MIDI control message numbers. It filters the signal before output. This eliminates discontinuities due to the low resolution of the MIDI (7 bit). The cutoff frequency can be set separately for each controller (suggested range: .1 to 5 Hz).

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl17* and *tonek*) when more controllers are required.



Warning

slider64f opcodes do not output the required initial value immediately, but only after some k-cycles because the filter slightly delays the output.

See Also

s16b14, s32b14, slider16, slider16f, slider32, slider32f, slider64, slider8, slider8f

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

slider8

slider8 — Creates a bank of 8 different MIDI control message numbers.

Description

Creates a bank of 8 different MIDI control message numbers.

Syntax

```
i1,...,i8 slider8 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \  
    ictlnum8, imin8, imax8, init8, ifn8  
  
k1,...,k8 slider8 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \  
    ictlnum8, imin8, imax8, init8, ifn8
```

Initialization

i1 ... i64 -- output values

ichan -- MIDI channel (1-16)

ictlnum1 ... ictlnum64 -- MIDI control number (0-127)

imin1 ... imin64 -- minimum values for each controller

imax1 ... imax64 -- maximum values for each controller

init1 ... init64 -- initial value for each controller

ifn1 ... ifn64 -- function table for conversion for each controller

icutoff1 ... icutoff64 -- low-pass filter cutoff frequency for each controller

Performance

k1 ... k64 -- output values

slider8 is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider8 allows a bank of 8 different MIDI control message numbers.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

In the i-rate version of *slider8*, there is not an initial value input argument, because the output is gotten directly from current status of internal controller array of Csound.

See Also

s16b14, s32b14, slider16, slider16f, slider32, slider32f, slider64, slider64f, slider8f

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

slider8f

slider8f — Creates a bank of 8 different MIDI control message numbers, filtered before output.

Description

Creates a bank of 8 different MIDI control message numbers, filtered before output.

Syntax

```
k1,...,k8 slider8f ichan, ictlnum1, imin1, imax1, init1, ifn1, icutoff1, \  
..., ictlnum8, imin8, imax8, init8, ifn8, icutoff8
```

Initialization

ichan -- MIDI channel (1-16)

ictlnum1 ... *ictlnum64* -- MIDI control number (0-127)

imin1 ... *imin64* -- minimum values for each controller

imax1 ... *imax64* -- maximum values for each controller

init1 ... *init64* -- initial value for each controller

ifn1 ... *ifn64* -- function table for conversion for each controller

icutoff1 ... *icutoff64* -- low-pass filter cutoff frequency for each controller

Performance

k1 ... *k64* -- output values

slider8f is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider8f allows a bank of 8 different MIDI control message numbers. It filters the signal before output. This eliminates discontinuities due to the low resolution of the MIDI (7 bit). The cutoff frequency can be set separately for each controller (suggested range: .1 to 5 Hz).

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.



Warning

slider8f opcodes do not output the required initial value immediately, but only after some k-cycles because the filter slightly delays the output.

See Also

s16b14, s32b14, slider16, slider16f, slider32, slider32f, slider64, slider64f, slider8

Credits

Author: Gabriel Maldonado
Italy
December 1998

New in Csound version 3.50

Thanks goes to Rasmus Ekman for pointing out the correct MIDI channel and controller number ranges.

slider16table

slider16table — Stores a bank of 16 different MIDI control messages to a table.

Description

Stores a bank of 16 different MIDI control messages to a table.

Syntax

```
kflag slider16table ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \  
init1, ifn1, .... , ictlnum16, imin16, imax16, init16, ifn16
```

Initialization

i1 ... i16 -- output values

ichan -- MIDI channel (1-16)

ioutTable -- number of the table that will contain the output

ioffset -- output table offset. A zero means that the output of the first slider will affect the first table element. A 10 means that the output of the first slider will affect the 11th table element.

ictlnum1 ... ictlnum16 -- MIDI control number (0-127)

imin1 ... imin16 -- minimum values for each controller

imax1 ... imax16 -- maximum values for each controller

init1 ... init16 -- initial value for each controller

ifn1 ... ifn16 -- function table for conversion for each controller

Performance

kflag -- a flag that informs if any control-change message in the bank has been received. In this case *kflag* is set to 1 is set to 1. Otherwise is set to zero.

slider16table is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider16table allows a bank of 16 different MIDI control message numbers.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

slider16table is very similar to *slider16* and *sliderN* family of opcodes (see their manual for more information). The actual difference is that the output is not stored to k-rate variables, but to a table,

denoted by the *ioutTable* argument. It is possible to define a starting index in order to use the same table for more than one spider bank (or other purposes).

It is possible to use this opcode together with *FLslidBnk2Setk* and *FLslidBnk2*, so you can synchronize the position of the MIDI values to the position of the FLTK valuator widgets of *FLslidBnk2*. Notice that you have to specify the same min/max values as well the linear/exponential responses in both *sliderNtable(f)* and *FLslidBnk2*. The exception is when using table-indexed response instead of a lin/exp response. In this case, in order to achieve a useful result, the table-indexed response and actual min/max values must be set only in *FLslidBnk2*, whereas, in *sliderNtable(f)*, you have to set a linear response and a minimum of zero and a maximum of one in all sliders.

See Also

slider16tablef, *slider32table*, *slider32tablef*, *slider64table*, *slider64tablef*, *slidertable8*, *slider8tablef*

Credits

Author: Gabriel Maldonado

New in Csound version 5.06

slider16tablef

slider16tablef — Stores a bank of 16 different MIDI control messages to a table, filtered before output.

Description

Stores a bank of 16 different MIDI control messages to a table, filtered before output.

Syntax

```
kflag slider16tablef ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \  
init1, ifn1, icutoff1, .... , ictlnum16, imin16, imax16, init16, ifn16, icutoff16
```

Initialization

ichan -- MIDI channel (1-16)

ioffset -- output table offset. A zero means that the output of the first slider will affect the first table element. A 10 means that the output of the first slider will affect the 11th table element.

ioutTable -- number of the table that will contain the output

ictlnum1 ... *ictlnum16* -- MIDI control number (0-127)

imin1 ... *imin16* -- minimum values for each controller

imax1 ... *imax16* -- maximum values for each controller

init1 ... *init16* -- initial value for each controller

ifn1 ... *ifn16* -- function table for conversion for each controller

icutoff1 ... *icutoff16* -- low-pass filter cutoff frequency for each controller

Performance

kflag -- a flag that informs if any control-change message in the bank has been received. In this case *kflag* is set to 1 is set to 1. Otherwise is set to zero.

slider16tablef is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider16tablef allows a bank of 16 different MIDI control message numbers. It filters the signal before output. This eliminates discontinuities due to the low resolution of the MIDI (7 bit). The cutoff frequency can be set separately for each controller (suggested range: .1 to 5 Hz).

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

slider8table is very similar to *slider16tablef* and *sliderNf* family of opcodes (see their manual for more information). The actual difference is that the output is not stored to k-rate variables, but to a table, denoted by the *ioutTable* argument. It is possible to define a starting index in order to use the same table for more than one spider bank (or other purposes).

It is possible to use this opcode together with *FLslidBnk2Setk* and *FLslidBnk2*, so you can synchronize the position of the MIDI values to the position of the FLTK valuator widgets of *FLslidBnk2*. Notice that you have to specify the same min/max values as well the linear/exponential responses in both *sliderNtable(f)* and *FLslidBnk2*. The exception is when using table-indexed response instead of a lin/exp response. In this case, in order to achieve a useful result, the table-indexed response and actual min/max values must be set only in *FLslidBnk2*, whereas, in *sliderNtable(f)*, you have to set a linear response and a minimum of zero and a maximum of one in all sliders.



Warning

slider16tablef does not output the required initial value immediately, but only after some k-cycles because the filter slightly delays the output.

See Also

slider16table, *slider32table*, *slider32tablef*, *slider64table*, *slider64tablef*, *slider8table*, *slider8tablef*

Credits

Author: Gabriel Maldonado

New in Csound version 5.06

slider32table

slider32table — Stores a bank of 32 different MIDI control messages to a table.

Description

Creates a bank of 32 different MIDI control messages to a table.

Syntax

```
kflag slider32table ichan, ioutTable, ioffset, ictlnum1, imin1, \  
imax1, init1, ifn1, .... , ictlnum32, imin32, imax32, init32, ifn32
```

Initialization

i1 ... i32 -- output values

ichan -- MIDI channel (1-16)

ioutTable -- number of the table that will contain the output

ioffset -- output table offset. A zero means that the output of the first slider will affect the first table element. A 10 means that the output of the first slider will affect the 11th table element.

ictlnum1 ... ictlnum32 -- MIDI control number (0-127)

imin1 ... imin32 -- minimum values for each controller

imax1 ... imax32 -- maximum values for each controller

init1 ... init32 -- initial value for each controller

ifn1 ... ifn32 -- function table for conversion for each controller

Performance

kflag -- a flag that informs if any control-change message in the bank has been received. In this case *kflag* is set to 1 is set to 1. Otherwise is set to zero.

slider32table is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider32table allows a bank of 32 different MIDI control message numbers.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

slider32table is very similar to *slider32* and *sliderN* family of opcodes (see their manual for more information). The actual difference is that the output is not stored to k-rate variables, but to a table,

denoted by the *ioutTable* argument. It is possible to define a starting index in order to use the same table for more than one spider bank (or other purposes).

It is possible to use this opcode together with *FLslidBnk2Setk* and *FLslidBnk2*, so you can synchronize the position of the MIDI values to the position of the FLTK valuator widgets of *FLslidBnk2*. Notice that you have to specify the same min/max values as well the linear/exponential responses in both *sliderNtable(f)* and *FLslidBnk2*. The exception is when using table-indexed response instead of a lin/exp response. In this case, in order to achieve a useful result, the table-indexed response and actual min/max values must be set only in *FLslidBnk2*, whereas, in *sliderNtable(f)*, you have to set a linear response and a minimum of zero and a maximum of one in all sliders.

See Also

slider16table, *slider16tablef*, *slider32tablef*, *slider64table*, *slider64tablef*, *slider8table*, *slider8tablef*

Credits

Author: Gabriel Maldonado

New in Csound version 5.06

slider32tablef

slider32tablef — Creates a bank of 32 different MIDI control message numbers, filtered before output.

Description

Creates a bank of 32 different MIDI control message numbers, filtered before output.

Syntax

```
kflag slider32tablef ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \  
init1, ifn1, icutoff1, .... , ictlnum32, imin32, imax32, init32, ifn32, icutoff32
```

Initialization

ichan -- MIDI channel (1-16)

ioutTable -- number of the table that will contain the output

ioffset -- output table offset. A zero means that the output of the first slider will affect the first table element. A 10 means that the output of the first slider will affect the 11th table element.

ictlnum1 ... *ictlnum32* -- MIDI control number (0-127)

imin1 ... *imin32* -- minimum values for each controller

imax1 ... *imax32* -- maximum values for each controller

init1 ... *init32* -- initial value for each controller

ifn1 ... *ifn32* -- function table for conversion for each controller

icutoff1 ... *icutoff32* -- low-pass filter cutoff frequency for each controller

Performance

kflag -- a flag that informs if any control-change message in the bank has been received. In this case *kflag* is set to 1 is set to 1. Otherwise is set to zero.

slider32tablef is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider32tablef allows a bank of 32 different MIDI control message numbers. It filters the signal before output. This eliminates discontinuities due to the low resolution of the MIDI (7 bit). The cutoff frequency can be set separately for each controller (suggested range: .1 to 5 Hz).

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

slider32tablef is very similar to *slider32tablef* and *sliderNf* family of opcodes (see their manual for more information). The actual difference is that the output is not stored to k-rate variables, but to a table, denoted by the *ioutTable* argument. It is possible to define a starting index in order to use the same table for more than one spider bank (or other purposes).

It is possible to use this opcode together with *FLslidBnk2Setk* and *FLslidBnk2*, so you can synchronize the position of the MIDI values to the position of the FLTK valuator widgets of *FLslidBnk2*. Notice that you have to specify the same min/max values as well the linear/exponential responses in both *sliderNtable(f)* and *FLslidBnk2*. The exception is when using table-indexed response instead of a lin/exp response. In this case, in order to achieve a useful result, the table-indexed response and actual min/max values must be set only in *FLslidBnk2*, whereas, in *sliderNtable(f)*, you have to set a linear response and a minimum of zero and a maximum of one in all sliders.



Warning

slider32tablef opcodes do not output the required initial value immediately, but only after some k-cycles because the filter slightly delays the output.

See Also

slider16, *slider16f*, *slider32*, *slider64*, *slider64f*, *slider8*, *slider8f*

Credits

Author: Gabriel Maldonado

New in Csound version 5.06

slider64table

slider64table — Stores a bank of 64 different MIDI control messages to a table.

Description

Creates a bank of 64 different MIDI control messages to a table.

Syntax

```
kflag slider64table ichan, ioutTable, ioffset, ictlnum1, imin1, \  
imax1, init1, ifn1, .... , ictlnum64, imin64, imax64, init64, ifn64
```

Initialization

i1 ... i64 -- output values

ichan -- MIDI channel (1-16)

ioutTable -- number of the table that will contain the output

ioffset -- output table offset. A zero means that the output of the first slider will affect the first table element. A 10 means that the output of the first slider will affect the 11th table element.

ictlnum1 ... ictlnum64 -- MIDI control number (0-127)

imin1 ... imin64 -- minimum values for each controller

imax1 ... imax64 -- maximum values for each controller

init1 ... init64 -- initial value for each controller

ifn1 ... ifn64 -- function table for conversion for each controller

Performance

kflag -- a flag that informs if any control-change message in the bank has been received. In this case *kflag* is set to 1 is set to 1. Otherwise is set to zero.

slider64table is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider64table allows a bank of 64 different MIDI control message numbers.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

slider64table is very similar to *slider64* and *sliderN* family of opcodes (see their manual for more information). The actual difference is that the output is not stored to k-rate variables, but to a table,

denoted by the *ioutTable* argument. It is possible to define a starting index in order to use the same table for more than one spider bank (or other purposes).

It is possible to use this opcode together with *FLslidBnk2Setk* and *FLslidBnk2*, so you can synchronize the position of the MIDI values to the position of the FLTK valuator widgets of *FLslidBnk2*. Notice that you have to specify the same min/max values as well the linear/exponential responses in both *sliderNtable(f)* and *FLslidBnk2*. The exception is when using table-indexed response instead of a lin/exp response. In this case, in order to achieve a useful result, the table-indexed response and actual min/max values must be set only in *FLslidBnk2*, whereas, in *sliderNtable(f)*, you have to set a linear response and a minimum of zero and a maximum of one in all sliders.

See Also

slider16table, *slider16tablef*, *slider32table*, *slider32tablef*, *slider64tablef* *slider8table*, *slider8tablef*

Credits

Author: Gabriel Maldonado

New in Csound version 5.06

slider64tablef

slider64tablef — Stores a bank of 64 different MIDI control messages to a table, filtered before output.

Description

Stores a bank of 64 different MIDI MIDI control messages to a table, filtered before output.

Syntax

```
kflag slider64tablef ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \  
init1, ifn1, icutoff1, .... , ictlnum64, imin64, imax64, init64, ifn64, icutoff64
```

Initialization

ichan -- MIDI channel (1-16)

ioutTable -- number of the table that will contain the output

ioffset -- output table offset. A zero means that the output of the first slider will affect the first table element. A 10 means that the output of the first slider will affect the 11th table element.

ictlnum1 ... *ictlnum64* -- MIDI control number (0-127)

imin1 ... *imin64* -- minimum values for each controller

imax1 ... *imax64* -- maximum values for each controller

init1 ... *init64* -- initial value for each controller

ifn1 ... *ifn64* -- function table for conversion for each controller

icutoff1 ... *icutoff64* -- low-pass filter cutoff frequency for each controller

Performance

kflag -- a flag that informs if any control-change message in the bank has been received. In this case *kflag* is set to 1 is set to 1. Otherwise is set to zero.

slider64tablef is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider64tablef allows a bank of 64 different MIDI control message numbers. It filters the signal before output. This eliminates discontinuities due to the low resolution of the MIDI (7 bit). The cutoff frequency can be set separately for each controller (suggested range: .1 to 5 Hz).

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

slider64tablef is very similar to *slider64tablef* and *sliderN* family of opcodes (see their manual for more information). The actual difference is that the output is not stored to k-rate variables, but to a table, denoted by the *ioutTable* argument. It is possible to define a starting index in order to use the same table for more than one spider bank (or other purposes).

It is possible to use this opcode together with *FLslidBnk2Setk* and *FLslidBnk2*, so you can synchronize the position of the MIDI values to the position of the FLTK valuator widgets of *FLslidBnk2*. Notice that you have to specify the same min/max values as well the linear/exponential responses in both *sliderNtable(f)* and *FLslidBnk2*. The exception is when using table-indexed response instead of a lin/exp response. In this case, in order to achieve a useful result, the table-indexed response and actual min/max values must be set only in *FLslidBnk2*, whereas, in *sliderNtable(f)*, you have to set a linear response and a minimum of zero and a maximum of one in all sliders.



Warning

slider64tablef opcodes do not output the required initial value immediately, but only after some k-cycles because the filter slightly delays the output.

See Also

slider16table, *slider16tablef*, *slider32table*, *slider32tablef*, *slider64table*, *slider8table*, *slider8tablef*

Credits

Author: Gabriel Maldonado

New in Csound version 5.06

slider8table

slider8table — Stores a bank of 8 different MIDI control messages to a table.

Description

Stores a bank of 8 different MIDI control messages to a table.

Syntax

```
kflag slider8table ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \  
init1, ifn1,..., ictlnum8, imin8, imax8, init8, ifn8
```

Initialization

i1 ... i8 -- output values

ichan -- MIDI channel (1-16)

ioutTable -- number of the table that will contain the output

ioffset -- output table offset. A zero means that the output of the first slider will affect the first table element. A 10 means that the output of the first slider will affect the 11th table element.

ictlnum1 ... ictlnum8 -- MIDI control number (0-127)

imin1 ... imin8 -- minimum values for each controller

imax1 ... imax8 -- maximum values for each controller

init1 ... init8 -- initial value for each controller

ifn1 ... ifn8 -- function table for conversion for each controller

Performance

kflag -- a flag that informs if any control-change message in the bank has been received. In this case *kflag* is set to 1 is set to 1. Otherwise is set to zero.

slider8table handles a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider8table allows a bank of 8 different MIDI control message numbers.

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

slider8table is very similar to *slider8* and *sliderN* family of opcodes (see their manual for more information). The actual difference is that the output is not stored to k-rate variables, but to a table, de-

noted by the *ioutTable* argument. It is possible to define a starting index in order to use the same table for more than one spider bank (or other purposes).

It is possible to use this opcode together with *FLslidBnk2Setk* and *FLslidBnk2*, so you can synchronize the position of the MIDI values to the position of the FLTK valuator widgets of *FLslidBnk2*. Notice that you have to specify the same min/max values as well the linear/exponential responses in both *sliderNtable(f)* and *FLslidBnk2*. The exception is when using table-indexed response instead of a lin/exp response. In this case, in order to achieve a useful result, the table-indexed response and actual min/max values must be set only in *FLslidBnk2*, whereas, in *sliderNtable(f)*, you have to set a linear response and a minimum of zero and a maximum of one in all sliders.

See Also

slider16table, *slider16tablef*, *slider32table*, *slider32tablef*, *slider64table*, *slider64tablef*, *slider8tabletablef*

Credits

Author: Gabriel Maldonado

New in Csound version 5.06

slider8tablef

slider8tablef — Stores a bank of 8 different MIDI control messages to a table, filtered before output.

Description

Stores a bank of 8 different MIDI control messages to a table, filtered before output.

Syntax

```
kflag slider8tablef ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \  
init1, ifn1, icutoff1, ...., ictlnum8, imin8, imax8, init8, ifn8, icutoff8
```

Initialization

ichan -- MIDI channel (1-16)

ioutTable -- number of the table that will contain the output

ioffset -- output table offset. A zero means that the output of the first slider will affect the first table element. A 10 means that the output of the first slider will affect the 11th table element.

ictlnum1 ... *ictlnum8* -- MIDI control number (0-127)

imin1 ... *imin8* -- minimum values for each controller

imax1 ... *imax8* -- maximum values for each controller

init1 ... *init8* -- initial value for each controller

ifn1 ... *ifn8* -- function table for conversion for each controller

icutoff1 ... *icutoff8* -- low-pass filter cutoff frequency for each controller

Performance

kflag -- a flag that informs if any control-change message in the bank has been received. In this case *kflag* is set to 1 is set to 1. Otherwise is set to zero.

slider8tablef is a bank of MIDI controllers, useful when using MIDI mixer such as Kawai MM-16 or others for changing whatever sound parameter in real-time. The raw MIDI control messages at the input port are converted to agree with *iminN* and *imaxN*, and an initial value can be set. Also, an optional non-interpolated function table with a custom translation curve is allowed, useful for enabling exponential response curves.

When no function table translation is required, set the *ifnN* value to 0, else set *ifnN* to a valid function table number. When table translation is enabled (i.e. setting *ifnN* value to a non-zero number referring to an already allocated function table), *initN* value should be set equal to *iminN* or *imaxN* value, else the initial output value will not be the same as specified in *initN* argument.

slider8tablef allows a bank of 8 different MIDI control message numbers. It filters the signal before output. This eliminates discontinuities due to the low resolution of the MIDI (7 bit). The cutoff frequency can be set separately for each controller (suggested range: .1 to 5 Hz).

As the input and output arguments are many, you can split the line using `\` (backslash) character (new in 3.47 version) to improve the readability. Using these opcodes is considerably more efficient than using the separate ones (*ctrl7* and *tonek*) when more controllers are required.

slider8tablef is very similar to *slider8f* and *sliderNf* family of opcodes (see their manual for more information). The actual difference is that the output is not stored to k-rate variables, but to a table, denoted by the *ioutTable* argument. It is possible to define a starting index in order to use the same table for more than one spider bank (or other purposes).

It is possible to use this opcode together with *FLslidBnk2Setk* and *FLslidBnk2*, so you can synchronize the position of the MIDI values to the position of the FLTK valuator widgets of *FLslidBnk2*. Notice that you have to specify the same min/max values as well the linear/exponential responses in both *sliderNtable(f)* and *FLslidBnk2*. The exception is when using table-indexed response instead of a lin/exp response. In this case, in order to achieve a useful result, the table-indexed response and actual min/max values must be set only in *FLslidBnk2*, whereas, in *sliderNtable(f)*, you have to set a linear response and a minimum of zero and a maximum of one in all sliders.



Warning

slider8tablef opcodes do not output the required initial value immediately, but only after some k-cycles because the filter slightly delays the output.

See Also

slider16table, *slider16tablef*, *slider32table*, *slider32tablef*, *slider64table*, *slider64tablef*, *slider8table*

Credits

Author: Gabriel Maldonado

New in Csound version 5.06

sliderKawai

sliderKawai — Creates a bank of 16 different MIDI control message numbers from a KAWAI MM-16 midi mixer.

Description

Creates a bank of 16 different MIDI control message numbers from a KAWAI MM-16 midi mixer.

Syntax

```
k1, k2, ..., k16 sliderKawai imin1, imax1, init1, ifn1, \  
imin2, imax2, init2, ifn2, ..., imin16, imax16, init16, ifn16
```

Initialization

ictnum1 ... *ictnum32* -- MIDI control number (0-127)

imin1 ... *imin16* -- minimum values for each controller

imax1 ... *imax16* -- maximum values for each controller

init1 ... *init16* -- initial value for each controller

ifn1 ... *ifn16* -- function table for conversion for each controller

Performance

k1 ... *k16* -- output values

The opcode *sliderKawai* is equivalent to *slider16*, but it has the controller and channel numbers (*ichan* and *ictnum*) hard-coded to make for quick compatibility with the KAWAI MM-16 midi mixer. This device doesn't allow changing the midi message associated to each slider. It can only output on control 7 for each fader on a separate midi channel. This opcode is a quick way of assigning the mixer's 16 faders to k-rate variables in csound.

See Also

slider16, *slider16f*, *slider32*, *slider32f*, *slider64*, *slider64f*, *slider8*, *slider8f*

Credits

Author: Gabriel Maldonado

New in Csound version 5.06

sndload

sndload — Loads a sound file into memory for use by *loscilx*

Description

sndload loads a sound file into memory for use by *loscilx*.

Syntax

```
sndload Sfname[, ifmt[, ichns[, isr[, ibas[, iamp[, istrtrt \
[, ilpmod[, ilps[, ilpe]]]]]]]]]
```

Initialization

Sfname - file name as a string constant or variable, string p-field, or a number that is used either as an index to strings set with *strset*, or, if that is not available, a file name in the format *soundin.n* is used. If the file name does not include a full path, the file is searched in the current directory first, then those specified by *SSDIR* (if defined), and finally *SFDIR*. If the same file was already loaded previously, it will not be read again, but the parameters *ibas*, *iamp*, *istrtrt*, *ilpmod*, *ilps*, and *ilpe* are still updated.

ifmt (optional, defaults to zero) - default sample format for raw (headerless) sound files; if the file has a header, this is ignored. Can be one of the following:

- 1: do not allow headerless files (fail with an init error)
- 0: use the same format as the one specified on the command line
- 1: 8 bit signed integers
- 2: a-law
- 3: u-law
- 4: 16 bit signed integers
- 5: 32 bit signed integers
- 6: 32 bit floats
- 7: 8 bit unsigned integers
- 8: 24 bit signed integers
- 9: 64 bit floats

ichns (optional, defaults to zero) - default number of channels for raw (headerless) sound files; if the file has a header, this is ignored. Zero or negative values are interpreted as 1 channel.

isr (optional, defaults to zero) - default sample rate for raw (headerless) sound files; if the file has a header, this is ignored. Zero or negative values are interpreted as the orchestra sample rate (*sr*).

ibas (optional, defaults to zero) - base frequency in Hz. If positive, overrides the value specified in the sound file header; otherwise, the value from the header is used if present, and 1.0 if the file does not include such information.

iamp (optional, defaults to zero) - amplitude scale. If non-zero, overrides the value specified in the sound file header (note: negative values are allowed, and will invert the sound output); otherwise, the value from the header is used if present, and 1.0 if the file does not include such information.

istrtrt (optional, defaults to -1) - starting position in sample frames, can be fractional. If non-negative, overrides the value specified in the sound file header; otherwise, the value from the header is used if present, and 0 if the file does not include such information. Note: even if this parameter is specified, the whole file is still read into memory.

ilpmod (optional, defaults to -1) - loop mode, can be one of the following:

any negative value: use the loop information specified in the sound file header, ignoring *ilps* and *ilpe*

0: no looping (*ilps* and *ilpe* are ignored)

1: forward looping (wrap around loop end if it is crossed in forward direction, and wrap around loop start if it is crossed in backward direction)

2: backward looping (change direction at loop end if it is crossed in forward direction, and wrap around loop start if it is crossed in backward direction)

3: forward-backward looping (change direction at both loop points if they are crossed as described above)

ilps (optional, defaults to 0) - loop start in sample frames (fractional values are allowed), or loop end if *ilps* is greater than *ilpe*. Ignored unless *ilpmod* is set to 1, 2, or 3. If the loop points are equal, the whole sample is looped.

ilpe (optional, defaults to 0) - loop end in sample frames (fractional values are allowed), or loop start if *ilps* is greater than *ilpe*. Ignored unless *ilpmod* is set to 1, 2, or 3. If the loop points are equal, the whole sample is looped.

Credits

Written by Istvan Varga.

2006

New in Csound 5.03

sndloop

sndloop — A sound looper with pitch control.

Description

This opcode records input audio and plays it back in a loop with user-defined duration and crossfade time. It also allows the pitch of the loop to be controlled, including reversed playback.

Syntax

```
asig, krec sndloop ain, kpitch, ktrig, idur, ifad
```

Initialisation

idur -- loop duration in seconds

ifad -- crossfade duration in seconds

Performance

asig -- output sig

krec -- 'rec on' signal, 1 when recording, 0 otherwise

kpitch -- pitch control (transposition ratio); negative values play the loop back in reverse

ktrig -- trigger signal: when 0, processing is bypassed. When switched on (*ktrig* \geq 1), the opcode starts recording until the loop memory is full. It then plays the looped sound until it is switched off again (*ktrig* = 0). Another recording can start again with *ktrig* \geq 1.

Examples

Example 474. Example

```
asig in                                ; get the signal in
ktrig line 0, 1, 1                      ; trigger signal
aout,krec sndloop asig, 1, ktrig, 4, 0.05 ; rec starts at 1 sec, for 4 secs 0.05 crossfade
printk 1, krec                          ; prints the recording signal
out aout
```

The example above shows the basic operation of `sndloop`. Pitch can be controlled at the k-rate, recording is started as soon as the trigger value is \geq 1. Recording can be restarted by making the trigger 0 and then 1 again.

Credits

Author: Victor Lazzarini;
April 2005

New in Version 5.00

sndwarp

sndwarp — Reads a mono sound sample from a table and applies time-stretching and/or pitch modification.

Description

sndwarp reads sound samples from a table and applies time-stretching and/or pitch modification. Time and frequency modification are independent from one another. For example, a sound can be stretched in time while raising the pitch!

The window size and overlap arguments are important to the result and should be experimented with. In general they should be as small as possible. For example, start with *iwsiz*e=*sr*/10 and *ioverlap*=15. Try *irandw*=*iwsiz*e*0.2. If you can get away with less overlaps, the program will be faster. But too few may cause an audible flutter in the amplitude. The algorithm reacts differently depending upon the input sound and there are no fixed rules for the best use in all circumstances. But with proper tuning, excellent results can be achieved.

Syntax

```
ares [, ac] sndwarp xamp, xtimewarp, xresample, ifn1, ibeg, iwsiz, \  
irandw, ioverlap, ifn2, itimemode
```

Initialization

ifn1 -- the number of the table holding the sound samples which will be subjected to the *sndwarp* processing. *GEN01* is the appropriate function generator to use to store the sound samples from a pre-existing soundfile.

ibeg -- the time in seconds to begin reading in the table (or soundfile). When *itimemode* is non-zero, the value of *xtimewarp* is offset by *ibeg*.

*iwsiz*e -- the window size in samples used in the time scaling algorithm.

irandw -- the bandwidth of a random number generator. The random numbers will be added to *iwsiz*e.

ioverlap -- determines the density of overlapping windows.

ifn2 -- a function used to shape the window. It is usually used to create a ramp of some kind from zero at the beginning and back down to zero at the end of each window. Try using a half sine (i.e.: f1 0 16384 9 .5 1 0) which works quite well. Other shapes can also be used.

Performance

ares -- the single channel of output from the *sndwarp* unit generator. *sndwarp* assumes that the function table holding the sampled signal is a mono one. This simply means that *sndwarp* will index the table by single-sample frame increments. The user must be aware then that if a stereo signal is used with *sndwarp*, time and pitch will be altered accordingly.

ac (optional) -- a single-layer (no overlaps), unwindowed versions of the time and/or pitch altered signal. They are supplied in order to be able to balance the amplitude of the signal output, which typically contains many overlapping and windowed versions of the signal, with a clean version of the time-scaled and pitch-shifted signal. The *sndwarp* process can cause noticeable changes in amplitude, (up and down), due to a time differential between the overlaps when time-shifting is being done. When used with a *balance* unit, *ac* can greatly enhance the quality of sound.

xamp -- the value by which to scale the amplitude (see note on the use of this when using *ac*).

xtimewarp -- determines how the input signal will be stretched or shrunk in time. There are two ways to use this argument depending upon the value given for *itimemode*. When the value of *itimemode* is 0, *xtimewarp* will scale the time of the sound. For example, a value of 2 will stretch the sound by 2 times. When *itimemode* is any non-zero value then *xtimewarp* is used as a time pointer in a similar way in which the time pointer works in *lpread* and *pvoc*. An example below illustrates this. In both cases, the pitch will *not* be altered by this process. Pitch shifting is done independently using *xresample*.

xresample -- the factor by which to change the pitch of the sound. For example, a value of 2 will produce a sound one octave higher than the original. The timing of the sound, however, will *not* be altered.

Examples

Here is an example of the *sndwarp* opcode. It uses the file *sndwarp.csd* [examples/sndwarp.csd], and *mary.wav* [examples/mary.wav].

Example 475. Example of the *sndwarp* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o sndwarp.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - play an audio file.
instr 1
; Use the audio file defined in Table #1.
a1 loscil 30000, 1, 1, 1

out a1
endin

; Instrument #2 - time-stretch an audio file.
instr 2
kamp init 6500
; Start at 1 second and end at 3.5 seconds.
ktimewarp line 1, p3, 3.5
; Playback at the normal speed.
kresample init 1
; Use the audio file defined in Table #1.
ifn1 = 1
ibeg = 0
iwsiz = 4410
irandw = 882
ioverlap = 15
; Use Table #2 for the windowing function.
ifn2 = 2
; Use the ktimewarp parameter as a "time" pointer.
itimemode = 1

a1 sndwarp kamp, ktimewarp, kresample, ifn1, ibeg, iwsiz, irandw, ioverlap, ifn2, itimemode
out a1
endin

</CsInstruments>
<CsScore>

; Table #1: an audio file.
f 1 0 262144 1 "mary.wav" 0 0 0
```

```
; Table #2: half of a sine wave.
f 2 0 16384 9 0.5 1 0

; Play Instrument #1 for 3.5 seconds.
i 1 0 3.5
; Play Instrument #2 for 7 seconds (time-stretched).
i 2 3.5 10.5
e

</CsScore>
</CsoundSynthesizer>
```

The below example shows a slowing down or stretching of the sound stored in the stored table (*ifn1*). Over the duration of the note, the stretching will grow from no change from the original to a sound which is ten times “slower” than the original. At the same time the overall pitch will move upward over the duration by an octave.

```
iwindfun = 1
isampfun = 2
ibeg = 0
iwindsize = 2000
iwindrand = 400
ioverlap = 10
awarp line 1, p3, 1
aresamp line 1, p3, 2
kenv line 1, p3, .1
asig sndwarp kenv, awarp, aresamp, isampfun, ibeg, iwindsize, iwindrand, ioverlap, iwindfun, 0
```

Now, here's an example using *xtimewarp* as a time pointer and using stereo:

```
itimemode = 1
atime line 0, p3, 10
ar1, ar2 sndwarpst kenv, atime, aresamp, sampfun, ibeg, iwindsize, iwindrand, ioverlap, \
iwindfun, itimemode
```

In the above, *atime* advances the time pointer used in the *sndwarpst* from 0 to 10 over the duration of the note. If *p3* is 20 then the sound will be two times slower than the original. Of course you can use a more complex function than just a single straight line to control the time factor.

Now the same as above but using the *balance* function with the optional outputs:

```
asig,acmp sndwarp 1, awarp, aresamp, isampfun, ibeg, iwindsize, iwindrand, ioverlap, iwindfun, it
abal balance asig, acmp

asig1,asig2,acmp1,acmp2 sndwarpst 1, atime, aresamp, sampfun, ibeg, iwindsize, iwindrand, ioverlap, \
iwindfun, itimemode
abal1 balance asig1, acmp1
abal2 balance asig2, acmp2
```

In the above two examples notice the use of the *balance* unit. The output of *balance* can then be scaled, enveloped, sent to an *out* or *outs*, and so on. Notice that the amplitude arguments to *sndwarp* and *sndwarpst* are “1” in these examples. By scaling the signal after the *sndwarp* process, *abal*, *abal1*, and *abal2* should contain signals that have nearly the same amplitude as the original input signal to the *sndwarp* process. This makes it much easier to predict the levels and avoid samples out of range or sample values that are too small.



More Advice

Only use the stereo version when you really need to be processing a stereo file. It is somewhat slower than the mono version and if you use the *balance* function it is slower again. There is nothing wrong with using a mono *sndwarp* in a stereo orchestra

and sending the result to one or both channels of the stereo output!

See Also

sndwarpst

Credits

Author: Richard Karpen
Seattle, WA USA
1997

Example written by Kevin Conder.

sndwarpst

sndwarpst — Reads a stereo sound sample from a table and applies time-stretching and/or pitch modification.

Description

sndwarpst reads stereo sound samples from a table and applies time-stretching and/or pitch modification. Time and frequency modification are independent from one another. For example, a sound can be stretched in time while raising the pitch!

The window size and overlap arguments are important to the result and should be experimented with. In general they should be as small as possible. For example, start with *iwsiz*e=st/10 and *ioverlap*=15. Try *irandw*=*iwsiz*e*.2. If you can get away with less overlaps, the program will be faster. But too few may cause an audible flutter in the amplitude. The algorithm reacts differently depending upon the input sound and there are no fixed rules for the best use in all circumstances. But with proper tuning, excellent results can be achieved.

Syntax

```
ar1, ar2 [,ac1] [, ac2] sndwarpst xamp, xtimewarp, xresample, ifn1, \
    ibeg, iwsiz, irandw, ioverlap, ifn2, itimemode
```

Initialization

ifn1 -- the number of the table holding the sound samples which will be subjected to the *sndwarpst* processing. *GEN01* is the appropriate function generator to use to store the sound samples from a pre-existing soundfile.

ibeg -- the time in seconds to begin reading in the table (or soundfile). When *itimemode* is non-zero, the value of *xtimewarp* is offset by *ibeg*.

*iwsiz*e -- the window size in samples used in the time scaling algorithm.

irandw -- the bandwidth of a random number generator. The random numbers will be added to *iwsiz*e.

ioverlap -- determines the density of overlapping windows.

ifn2 -- a function used to shape the window. It is usually used to create a ramp of some kind from zero at the beginning and back down to zero at the end of each window. Try using a half a sine (i.e.: f1 0 16384 9 .5 1 0) which works quite well. Other shapes can also be used.

Performance

ar1, *ar2* -- *ar1* and *ar2* are the stereo (left and right) outputs from *sndwarpst*. *sndwarpst* assumes that the function table holding the sampled signal is a stereo one. *sndwarpst* will index the table by a two-sample frame increment. The user must be aware then that if a mono signal is used with *sndwarpst*, time and pitch will be altered accordingly.

ac1, *ac2* -- *ac1* and *ac2* are single-layer (no overlaps), unwindowed versions of the time and/or pitch altered signal. They are supplied in order to be able to balance the amplitude of the signal output, which typically contains many overlapping and windowed versions of the signal, with a clean version of the time-scaled and pitch-shifted signal. The *sndwarpst* process can cause noticeable changes in amplitude, (up and down), due to a time differential between the overlaps when time-shifting is being done. When used with a *balance* unit, *ac1* and *ac2* can greatly enhance the quality of sound. They are optional, but note that they must both be present in the syntax (use both or neither). An example of how to use this is given below.

xamp -- the value by which to scale the amplitude (see note on the use of this when using *ac1* and *ac2*).

xtimewarp -- determines how the input signal will be stretched or shrunk in time. There are two ways to use this argument depending upon the value given for *itimemode*. When the value of *itimemode* is 0, *xtimewarp* will scale the time of the sound. For example, a value of 2 will stretch the sound by 2 times. When *itimemode* is any non-zero value then *xtimewarp* is used as a time pointer in a similar way in which the time pointer works in *lpread* and *pvoc*. An example below illustrates this. In both cases, the pitch will *not* be altered by this process. Pitch shifting is done independently using *xresample*.

xresample -- the factor by which to change the pitch of the sound. For example, a value of 2 will produce a sound one octave higher than the original. The timing of the sound, however, will *not* be altered.

Examples

The below example shows a slowing down or stretching of the sound stored in the stored table (*ifn1*). Over the duration of the note, the stretching will grow from no change from the original to a sound which is ten times “slower” than the original. At the same time the overall pitch will move upward over the duration by an octave.

```
iwindfun = 1
isampfun = 2
ibeg = 0
iwindsize = 2000
iwindrand = 400
ioverlap = 10
awarp line 1, p3, 1
aresamp line 1, p3, 2
kenv line 1, p3, .1
asig sndwarp kenv, awarp, aresamp, isampfun, ibeg, iwindsize, iwindrand, ioverlap, iwindfun, 0
```

Now, here's an example using *xtimewarp* as a time pointer and using stereo:

```
itimemode = 1
atime line 0, p3, 10
ar1, ar2 sndwarpst kenv, atime, aresamp, sampfun, ibeg, iwindsize, iwindrand, ioverlap, \
iwindfun, itimemode
```

In the above, *atime* advances the time pointer used in the *sndwarpst* from 0 to 10 over the duration of the note. If p3 is 20 then the sound will be two times slower than the original. Of course you can use a more complex function than just a single straight line to control the time factor.

Now the same as above but using the *balance* function with the optional outputs:

```
asig,acmp sndwarp 1, awarp, aresamp, isampfun, ibeg, iwindsize, iwindrand, ioverlap, iwindfun, i
abal balance asig, acmp
asig1,asig2,acmp1,acmp2 sndwarpst 1, atime, aresamp, sampfun, ibeg, iwindsize, iwindrand, ioverlap, \
iwindfun, itimemode
abal1 balance asig1, acmp1
abal2 balance asig2, acmp2
```

In the above two examples notice the use of the *balance* unit. The output of *balance* can then be scaled, enveloped, sent to an *out* or *outs*, and so on. Notice that the amplitude arguments to *sndwarp* and *sndwarpst* are “1” in these examples. By scaling the signal after the *sndwarp* process, *abal*, *abal1*, and *abal2* should contain signals that have nearly the same amplitude as the original input signal to the *sndwarp* process. This makes it much easier to predict the levels and avoid samples out

of range or sample values that are too small.



More Advice

Only use the stereo version when you really need to be processing a stereo file. It is somewhat slower than the mono version and if you use the *balance* function it is slower again. There is nothing wrong with using a mono *sndwarp* in a stereo orchestra and sending the result to one or both channels of the stereo output!

See Also

sndwarp

Credits

Author: Richard Karpen
Seattle, WA USA
1997

socksend

socksend — Sends data to other processes using the low-level UDP or TCP protocols

Description

Transmits data directly using the UDP (socksend and socksend) or TCP (stsend) protocol onto a network. The data is not subject to any encoding or special routing. The socksend opcode send a stereo signal interleaved.

Syntax

```
socksend asig, Sipaddr, iport, ilength
```

```
socksend asigl, asigr, Sipaddr, iport,  
          ilength
```

```
stsend asig, Sipaddr, iport
```

Initialization

Sipaddr -- a string that is the IP address of the receiver in standard 4-octet dotted form.

iport -- the number of the port that is used for the communication.

ilength -- the length of the individual packets in UDP transmission. This number must be sufficiently small to fit a single MTU, which is set to the save value of 1456. In UDP transmissions the receiver needs to know this value

Performance

asig, asigl, asigr -- audio data to be transmitted.

Example

The example shows a simple sine wave being sent just once to a computer called "172.16.0.255", on port 7777 using UDP. Note that .255 is often used for broadcasting.

```
sr = 44100  
ksmps = 100  
nchnls = 1  
  
instr 1  
a1 oscil 20000,441,1  
    socksend a1, "172.16.0.255",7777, 200  
endin
```

Credits

Author: John ffitch
2006

sockrecv

sockrecv — Receives data from other processes using the low-level UDP or TCP protocols

Description

Receives directly using the UDP (sockrecv and sockrecvs) or TCP (strecv) protocol onto a network. The data is not subject to any encoding or special routing. The sockrecvs opcode receives a stereo signal interleaved.

Syntax

```
asig sockrecv iport, ilength
```

```
asigl, asigr sockrecvs iport, ilength
```

```
asig strecv Sipaddr, iport
```

Initialization

Sipaddr -- a string that is the IP address of the sender in standard 4-octet dotted form.

iport -- the number of the port that is used for the communication.

ilength -- the length of the individual packets in UDP transmission. This number must be sufficiently small to fit a single MTU, which is set to the save value of 1456. In UDP transmissions the sender and receiver needs agree on this value

Performance

asig, asigl, asigr -- audio data to be received.

Example

The example shows a mono signal being received on port 7777 using UDP.

```
sr = 44100
ksmps = 100
nchnls = 1

instr 1
al sockrecv      7777, 200
  out  al
endin
```

Credits

Author: John ffitch
2006

soundin

soundin — Reads audio data from an external device or stream.

Description

Reads audio data from an external device or stream. Up to 24 channels may be read.

Syntax

```
ar1[, ar2[, ar3[, ... a24]]] soundin ifilcod [, iskptim] [, iformat] \  
[, iskipinit] [, ibufsize]
```

Initialization

ifilcod -- integer or character-string denoting the source soundfile name. An integer denotes the file soundin.filcod; a character-string (in double quotes, spaces permitted) gives the filename itself, optionally a full pathname. If not a full path, the named file is sought first in the current directory, then in that given by the environment variable SSDIR (if defined) then by SFDIR. See also *GEN01*.

iskptim (optional, default=0) -- time in seconds of input sound to be skipped. The default value is 0. In csound 5.00 and later, this may be negative to add a delay instead of skipping time.

iformat (optional, default=0) -- specifies the audio data file format:

- 1 = 8-bit signed char (high-order 8 bits of a 16-bit integer)
- 2 = 8-bit A-law bytes
- 3 = 8-bit U-law bytes
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats
- 7 = 8-bit unsigned int (not available in Csound versions older than 5.00)
- 8 = 24-bit int (not available in Csound versions older than 5.00)
- 9 = 64-bit doubles (not available in Csound versions older than 5.00)

iskipinit -- switches off all initialisation if non zero (default=0). This was introduced in 4_23f13 and csound5.

ibufsize -- buffer size in mono samples (not sample frames). Not available in Csound versions older than 5.00. The default buffer size is 2048.

If *iformat* = 0 it is taken from the soundfile header, and if no header from the Csound *-o* command-line flag. The default value is 0.

Performance

soundin is functionally an audio generator that derives its signal from a pre-existing file. The number of channels read in is controlled by the number of result cells, a1, a2, etc., which must match that of the input file. A *soundin* opcode opens this file whenever the host instrument is initialized,

then closes it again each time the instrument is turned off.

There can be any number of *soundin* opcodes within a single instrument or orchestra. Two or more of them can read simultaneously from the same external file.



Note to Windows users

Windows users typically use back-slashes, “\”, when specifying the paths of their files. As an example, a Windows user might use the path “c:\music\samples\loop001.wav”. This is problematic because back-slashes are normally used to specify special characters.

To correctly specify this path in Csound, one may alternately:

- *Use forward slashes:* c:/music/samples/loop001.wav
- *Use back-slash special characters, “\\”:* c:\\music\\samples\\loop001.wav

Examples

Here is an example of the *soundin* opcode. It uses the file *soundin.csd* [examples/soundin.csd], *beats.wav* [examples/beats.wav].

Example 476. Example of the *soundin* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o soundin.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1 - play an audio file.
instr 1
  asig soundin "beats.wav"
  out asig
endin

</CsInstruments>
<CsScore>

; Play Instrument #1, the audio file, for three seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

diskin, in, inh, ino, inq, ins

Credits

Authors: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

Example written by Kevin Conder.

Warning to Windows users added by Kevin Conder, April 2002

soundout

soundout — Deprecated. Writes audio output to a disk file.

Description



Note

The usage of *soundout* is discouraged. Please use *fout* instead.

Writes audio output to a disk file.

Syntax

```
soundout  asigl, ifilcod [, iformat]
```

Initialization

ifilcod -- integer or character-string denoting the destination soundfile name. An integer denotes the file soundin.filcod; a character-string (in double quotes, spaces permitted) gives the filename itself, optionally a full pathname. If not a full path, the named file is sought first in the current directory, then in that given by the environment variable SSDIR (if defined) then by SFDIR. See also *GEN01*.

iformat (optional, default=0) -- specifies the audio data file format:

- 1 = 8-bit signed char (high-order 8 bits of a 16-bit integer)
- 2 = 8-bit A-law bytes
- 3 = 8-bit U-law bytes
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats

If *iformat* = 0 it is taken from the soundfile header, and if no header from the Csound -o command-line flag. The default value is 0.

Performance

soundout writes audio output to a disk file.



Note

Use of *fout* is recommended instead of *soundout*

See Also

fout, *out*, *outh*, *outo*, *outq*, *outq1*, *outq2*, *outq3*, *outq4*, *outs*, *outs1*, *outs2* soundouts

Credits

Author: Barry L. Vercoe, Matt Ingalls/Mike Berry
MIT, Mills College
1993-1997

soundouts

soundouts — Deprecated. Writes audio output to a disk file.

Description



Note

The usage of *soundouts* is discouraged. Please use *fout* instead.

Writes audio output to a disk file.

Syntax

```
soundouts  asigl, asigr, ifilcod [, iformat]
```

Initialization

ifilcod -- integer or character-string denoting the destination soundfile name. An integer denotes the file soundout.ifilcod; a character-string (in double quotes, spaces permitted) gives the filename itself, optionally a full pathname. If not a full path, the named file is written relative to the directory given by the SFDIR environment variable if defined, or the current directory. See also *GEN01*.

iformat (optional, default=0) -- specifies the audio data file format:

- 1 = 8-bit signed char (high-order 8 bits of a 16-bit integer)
- 4 = 16-bit short integers
- 5 = 32-bit long integers
- 6 = 32-bit floats

If *iformat* = 0 it is taken from the Csound *-o* command-line flag. The default value is 0.

Performance

soundouts writes stereo audio output to a disk file in raw (headerless) format without 0dBFS scaling. The expected range of the audio signals depends on the selected sample format.



Note

Use of *fout* is recommended instead of *soundouts*

See Also

out, *outh*, *outo*, *outq*, *outq1*, *outq2*, *outq3*, *outq4*, *outs*, *outs1*, *outs2* *soundout*

Credits

Author: Istvan Varga

space

`space` — Distributes an input signal among 4 channels using cartesian coordinates.

Description

`space` takes an input signal and distributes it among 4 channels using Cartesian xy coordinates to calculate the balance of the outputs. The xy coordinates can be defined in a separate text file and accessed through a Function statement in the score using *Gen28*, or they can be specified using the optional *kx*, *ky* arguments. The advantages to the former are:

1. A graphic user interface can be used to draw and edit the trajectory through the Cartesian plane
2. The file format is in the form time1 X1 Y1 time2 X2 Y2 time3 X3 Y3 allowing the user to define a time-tagged trajectory

`space` then allows the user to specify a time pointer (much as is used for *pvoc*, *lpread* and some other units) to have detailed control over the final speed of movement.

Syntax

```
a1, a2, a3, a4 space asig, ifn, ktime, kverbsend, kx, ky
```

Initialization

ifn -- number of the stored function created using *Gen28*. This function generator reads a text file which contains sets of three values representing the xy coordinates and a time-tag for when the signal should be placed at that location. The file should look like:

```
0    -1    1
1     1    1
2     4     4
2.1  -4   -4
3    10  -10
5   -40    0
```

If that file were named “move” then the *Gen28* call in the score would like:

```
f1 0 0 28 "move"
```

Gen28 takes 0 as the size and automatically allocates memory. It creates values to 10 milliseconds of resolution. So in this case there will be 500 values created by interpolating X1 to X2 to X3 and so on, and Y1 to Y2 to Y3 and so on, over the appropriate number of values that are stored in the function table. In the above example, the sound will begin in the left front, over 1 second it will move to the right front, over another second it move further into the distance but still in the left front, then in just 1/10th of a second it moves to the left rear, a bit distant. Finally over the last .9 seconds the sound will move to the right rear, moderately distant, and it comes to rest between the two left channels (due west!), quite distant. Since the values in the table are accessed through the use of a time-pointer in the *space* unit, the actual timing can be made to follow the file's timing exactly or it can be made to go faster or slower through the same trajectory. If you have access to the GUI that allows

one to draw and edit the files, there is no need to create the text files manually. But as long as the file is ASCII and in the format shown above, it doesn't matter how it is made!



Important

If *ifn* is 0, then *space* will take its values for the xy coordinates from *kx* and *ky*.

Performance

The configuration of the xy coordinates in space places the signal in the following way:

- a1 is -1, 1
- a2 is 1, 1
- a3 is -1, -1
- a4 is 1, -1

This assumes a loudspeaker set up as a1 is left front, a2 is right front, a3 is left back, a4 is right back. Values greater than 1 will result in sounds being attenuated, as if in the distance. *space* considers the speakers to be at a distance of 1; smaller values of xy can be used, but *space* will not amplify the signal in this case. It will, however balance the signal so that it can sound as if it were within the 4 speaker *space*. x=0, y=1, will place the signal equally balanced between left and right front channels, x=y=0 will place the signal equally in all 4 channels, and so on. Although there must be 4 output signals from *space*, it can be used in a 2 channel orchestra. If the xy's are kept so that $Y \geq 1$, it should work well to do panning and fixed localization in a stereo field.

asig -- input audio signal.

ktime -- index into the table containing the xy coordinates. If used like:

```
ktime      line 0, 5, 5
a1, a2, a3, a4 space asig, 1, ktime, ...
```

with the file “move” described above, the speed of the signal's movement will be exactly as described in that file. However:

```
ktime      line 0, 10, 5
```

the signal will move at half the speed specified. Or in the case of:

```
ktime      line 5, 15, 0
```

the signal will move in the reverse direction as specified and 3 times slower! Finally:

```
ktime      line 2, 10, 3
```

will cause the signal to move only from the place specified in line 3 of the text file to the place spe-

cified in line 5 of the text file, and it will take 10 seconds to do it.

*kreverb*send -- the percentage of the direct signal that will be factored along with the distance as derived from the XY coordinates to calculate signal amounts that can be sent to reverb units such as reverb, or reverb2.

kx, *ky* -- when *ifn* is 0, *space* and *spdist* will use these values as the XY coordinates to localize the signal.

Examples

```
instr 1
  asig      ;some audio signal
  ktime      line 0, p3, p10
  a1, a2, a3, a4      space asig,1, ktime, .1
  ar1, ar2, ar3, ar4  spsend

  ga1 = ga1+ar1
  ga2 = ga2+ar2
  ga3 = ga3+ar3
  ga4 = ga4+ar4

                                outq a1, a2, a3, a4
endin

instr 99 ; reverb instrument

  a1 reverb2 ga1, 2.5, .5
  a2 reverb2 ga2, 2.5, .5
  a3 reverb2 ga3, 2.5, .5
  a4 reverb2 ga4, 2.5, .5

  outq a1, a2, a3, a4
  ga1=0
  ga2=0
  ga3=0
  ga4=0
endin
```

In the above example, the signal, *asig*, is moved according to the data in Function #1 indexed by *ktime*. *space* sends the appropriate amount of the signal internally to *spsend*. The outputs of the *spsend* are added to global accumulators in a common Csound style and the global signals are used as inputs to the reverb units in a separate instrument.

space can be useful for quad and stereo panning as well as fixed placement of sounds anywhere between two loudspeakers. Below is an example of the fixed placement of sounds in a stereo field using xy values from the score instead of a function table.

```
instr 1
  ...
  a1, a2, a3, a4      space asig, 0, 0, .1, p4, p5
  ar1, ar2, ar3, ar4  spsend

  ga1=ga1+ar1
  ga2=ga2+ar2

                                outs  a1, a2
endin

instr 99 ; reverb....
  ...
endin
```

A few notes: p4 and p5 are the X and Y values


```
;place the sound in the left speaker and near
i1 0 1 -1 1
;place the sound in the right speaker and far
i1 1 1 45 45
;place the sound equally between left and right and in the middle ground distance
i1 2 1 0 12
e
```

The next example shows a simple intuitive use of the distance values returned by *spdist* to simulate Doppler shift.

```
ktime          line 0, p3, 10
kdist          spdist 1, ktime
kfreq = (ifreq * 340) / (340 + kdist)
asig          oscili iamp, kfreq, 1

a1, a2, a3, a4  space asig, 1, ktime, .1
ar1, ar2, ar3, ar4 ssend
```

The same function and time values are used for both *spdist* and *space*. This insures that the distance values used internally in the *space* unit will be the same as those returned by *spdist* to give the impression of a Doppler shift!

See Also

spdist, *spsend*

Credits

Author: Richard Karpen
Seattle, WA USA
1998

New in Csound version 3.48

spat3d

spat3d — Positions the input sound in a 3D space and allows moving the sound at k-rate.

Description

This opcode positions the input sound in a 3D space, with optional simulation of room acoustics, in various output formats. *spat3d* allows moving the sound at k-rate (this movement is interpolated internally to eliminate "zipper noise" if sr not equal to kr).

Syntax

```
aW, aX, aY, aZ spat3d ain, kX, kY, kZ, idist, ift, imode, imdel, iovr [, istor]
```

Initialization

idist -- For modes 0 to 3, *idist* is the unit circle distance in meters. For mode 4, *idist* is the distance between microphones.

The following formulas describe amplitude and delay as a function of sound source distance from microphone(s):

$$\text{amplitude} = 1 / (0.1 + \text{distance})$$
$$\text{delay} = \text{distance} / 340 \text{ (in seconds)}$$

Distance can be calculated as:

$$\text{distance} = \sqrt{iX^2 + iY^2 + iZ^2}$$

In Mode 4, distance can be calculated as:

$$\begin{aligned} \text{distance from left mic} &= \sqrt{(iX + idist/2)^2 + iY^2 + iZ^2} \\ \text{distance from right mic} &= \sqrt{(iX - idist/2)^2 + iY^2 + iZ^2} \end{aligned}$$

With *spat3d* the distance between the sound source and any microphone should be at least $(340 * 18) / \text{sr}$ meters. Shorter distances will work, but may produce artifacts in some cases. There is no such limitation for *spat3di* and *spat3dt*.

Sudden changes or discontinuities in sound source location can result in pops or clicks. Very fast movement may also degrade quality.

ift -- Function table storing room parameters (for free field spatialization, set it to zero or negative). Table size is 54. The values in the table are:

Room Parameter	Purpose
0	Early reflection recursion depth (0 is the sound

Room Parameter	Purpose
	source, 1 is the first reflection etc.) for spat3d and spat3di. The number of echoes for four walls (front, back, right, left) is: $N = (2 * R + 2) * R$. If all six walls are enabled: $N = (((4 * R + 6) * R + 8) * R) / 3$
1	Late reflection recursion depth (used by spat3dt only). spat3dt skips early reflections and renders echoes up to this level. If early reflection depth is negative, spat3d and spat3di will output zero, while spat3dt will start rendering from the sound source.
2	imdel for spat3d. Overrides opcode parameter if non-negative.
3	irlen for spat3dt. Overrides opcode parameter if non-negative.
4	idist value. Overrides opcode parameter if ≥ 0 .
5	Random seed (0 - 65535) -1 seeds from current time.
6 - 53	wall parameters (w = 6: ceil, w = 14: floor, w = 22: front, w = 30: back, w = 38: right, w = 46: left)
w + 0	Enable reflections from this wall (0: no, 1: yes)
w + 1	Wall distance from listener (in meters)
w + 2	Randomization of wall distance (0 - 1) (in units of 1 / (wall distance))
w + 3	Reflection level (-1 - 1)
w + 4	Parametric equalizer frequency in Hz.
w + 5	Parametric equalizer level (1.0: no filtering)
w + 6	Parametric equalizer Q (0.7071: no resonance)
w + 7	Parametric equalizer mode (0: peak EQ, 1: low shelf, 2: high shelf)

imode -- Output mode

- 0: B format with W output only (mono)

$aout = aW$

- 1: B format with W and Y output (stereo)

$aleft = aW + 0.7071 * aY$
 $aright = aW - 0.7071 * aY$

- 2: B format with W, X, and Y output (2D). This can be converted to UHJ:

$aWre, aWim \quad \text{hilbert } aW$
 $aXre, aXim \quad \text{hilbert } aX$
 $aYre, aYim \quad \text{hilbert } aY$
 $aWXr = 0.0928 * aXre + 0.4699 * aWre$

```
aWxiYr = 0.2550*aXim - 0.1710*aWim + 0.3277*aYre
aleft  = aWxr + aWxiYr
aright = aWxr - aWxiYr
```

- 3: B format with all outputs (3D)
- 4: Simulates a pair of microphones (stereo output)

```
aW    butterlp aW, ifreq    ; recommended values for ifreq
aY    butterlp aY, ifreq    ; are around 1000 Hz
aleft = aW + aX
aright = aY + aZ
```

Mode 0 is the cheapest to calculate, while mode 4 is the most expensive.

In Mode 4, The optional lowpass filters can change the frequency response depending on direction. For example, if the sound source is located left to the listener then the high frequencies are attenuated in the right channel and slightly increased in the left. This effect can be disabled by not using filters. You can also experiment with other filters (tone etc.) for better effect.

Note that mode 4 is most useful for listening with headphones, and is also more expensive to calculate than the B-format (0 to 3) modes. The *idist* parameter in this case sets the distance between left and right microphone; for headphones, values between 0.2 - 0.25 are recommended, although higher settings up to 0.4 may be used for wide stereo effects.

More information about B format can be found here: http://www.york.ac.uk/inst/mustech/3d_audio/ambis2.htm

imdel -- Maximum delay time for spat3d in seconds. This has to be longer than the delay time of the latest reflection (depends on room dimensions, sound source distance, and recursion depth; using this formula gives a safe (although somewhat overestimated) value:

$$\text{imdel} = (R + 1) * \text{sqrt}(W*W + H*H + D*D) / 340.0$$

where R is the recursion depth, W, H, and D are the width, height, and depth of the room, respectively).

iovr -- Oversample ratio for spat3d (1 to 8). Setting it higher improves quality at the expense of memory and CPU usage. The recommended value is 2.

istor (optional, default=0) -- Skip initialization if non-zero (default: 0).

Performance

aW, *aX*, *aY*, *aZ* -- Output signals

	mode 0	mode 1	mode 2	mode 3	mode 4
aW	W out	W out	W out	W out	left chn / low freq.
aX	0	0	X out	X out	left chn / high freq.
aY	0	Y out	Y out	Y out	right chn / low freq.
aZ	0	0	0	Z out	right chn / high

	mode 0	mode 1	mode 2	mode 3	mode 4
					fr.

ain -- Input signal

kX, *kY*, *kZ* -- Sound source coordinates (in meters)

If you encounter very slow performance (up to 100 times slower), it may be caused by denormals (this is also true of many other IIR opcodes, including *butterlp*, *pareq*, *hilbert*, and many others). Underflows can be avoided by:

- Using the *denorm* opcode on *ain* before *spat3d*.
- mixing low level DC or noise to the input signal, e.g.

```
atmp rnd31 1/1e24, 0, 0
```

```
aW, aX, aY, aZ spa3di ain + atmp, ...
```

or

```
aW, aX, aY, aZ spa3di ain + 1/1e24, ...
```

- reducing *irlen* in the case of *spat3dt* (which does not have an input signal). A value of about 0.005 is suitable for most uses, although it also depends on EQ settings. If the equalizer is not used, “irlen” can be set to 0.

Examples

Here is a example of the *spat3d* opcode that outputs a stereo file. It uses the file *spat3d_stereo.csd* [examples/spat3d_stereo.csd].

Example 477. Stereo example of the *spat3d* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o spat3d_stereo.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Istvan Varga */
sr      = 48000
kr      = 1000
ksmps   = 48
nchnls  = 2

/* room parameters */
idep     = 3      /* early reflection depth      */

itmp     ftgen    1, 0, 64, -2,
/* depth1, depth2, max delay, IR length, idist, seed */ \
idep, 48, -1, 0.01, 0.25, 123, \
1, 21.982, 0.05, 0.87, 4000.0, 0.6, 0.7, 2, /* ceil */ \
1, 1.753, 0.05, 0.87, 3500.0, 0.5, 0.7, 2, /* floor */ \
```

```

1, 15.220, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* front */ \
1, 9.317, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* back */ \
1, 17.545, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* right */ \
1, 12.156, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* left */

instr 1

/* some source signal */

a1 phasor 150 ; oscillator
a1 butterbp a1, 500, 200 ; filter
a1 = taninv(a1 * 100)
a2 phasor 3 ; envelope
a2 mirror 40*a2, -100, 5
a2 limit a2, 0, 1
a1 = a1 * a2 * 9000

kazim line 0, 2.5, 360 ; move sound source around
kdist line 1, 10, 4 ; distance

; convert polar coordinates
kX = sin(kazim * 3.14159 / 180) * kdist
kY = cos(kazim * 3.14159 / 180) * kdist
kZ = 0

a1 = a1 + 0.000001 * 0.000001 ; avoid underflows

imode = 1 ; change this to 3 for 8 spk in a cube,
; or 1 for simple stereo

aW, aX, aY, aZ spat3d a1, kX, kY, kZ, 1.0, 1, imode, 2, 2

aW = aW * 1.4142

; stereo
;
aL = aW + aY /* left */
aR = aW - aY /* right */

; quad (square)
;
;aFL = aW + aX + aY /* front left */
;aFR = aW + aX - aY /* front right */
;aRL = aW - aX + aY /* rear left */
;aRR = aW - aX - aY /* rear right */

; eight channels (cube)
;
;aUFL = aW + aX + aY + aZ /* upper front left */
;aUFR = aW + aX - aY + aZ /* upper front right */
;aURL = aW - aX + aY + aZ /* upper rear left */
;aURR = aW - aX - aY + aZ /* upper rear right */
;aLFL = aW + aX + aY - aZ /* lower front left */
;aLFR = aW + aX - aY - aZ /* lower front right */
;aLRL = aW - aX + aY - aZ /* lower rear left */
;aLRR = aW - aX - aY - aZ /* lower rear right */

outs aL, aR

endin

</CsInstruments>
<CsScore>

/* Written by Istvan Varga */
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>

```

Here is a example of the spat3d opcode that outputs a UHJ file. It uses the file *spat3d_UHJ.csd* [examples/spat3d_UHJ.csd].

Example 478. UHJ example of the spat3d opcode.

```

<CsoundSynthesizer>
<CsOptions>

```

```
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-o dac        -i adc      -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o spat3d_UHJ.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Istvan Varga */
sr = 48000
kr = 750
ksmps = 64
nchnls = 2

itmp      ftgen      1, 0, 64, -2,
/* depth1, depth2, max delay, IR length, idist, seed */ \
3, 48, -1, 0.01, 0.25, 123, \
1, 21.982, 0.05, 0.87, 4000.0, 0.6, 0.7, 2, /* ceil */ \
1, 1.753, 0.05, 0.87, 3500.0, 0.5, 0.7, 2, /* floor */ \
1, 15.220, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* front */ \
1, 9.317, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* back */ \
1, 17.545, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* right */ \
1, 12.156, 0.05, 0.87, 5000.0, 0.8, 0.7, 2 /* left */

instr 1

p3 = p3 + 1.0

kazim line 0.0, 4.0, 360.0      ; azimuth
kelev line 40, p3 - 1.0, -20    ; elevation
kdist = 2.0                    ; distance
; convert coordinates
kX = kdist * cos(kelev * 0.01745329) * sin(kazim * 0.01745329)
kY = kdist * cos(kelev * 0.01745329) * cos(kazim * 0.01745329)
kZ = kdist * sin(kelev * 0.01745329)

; source signal
a1 phasor 160.0
a2 delayl a1
a1 = a1 - a2
kffrq1 port 200.0, 0.8, 12000.0
affrq upsamp kffrq1
affrq pareq affrq, 5.0, 0.0, 1.0, 2
kffrq downsamp affrq
aenv4 phasor 3.0
aenv4 limit 2.0 - aenv4 * 8.0, 0.0, 1.0
a1 butterbp a1 * aenv4, kffrq, 160.0
aenv linseg 1.0, p3 - 1.0, 1.0, 0.04, 0.0, 1.0, 0.0
a_ = 4000000 * a1 * aenv + 0.00000001

; spatialize
a_W, a_X, a_Y, a_Z spat3d a_, kX, kY, kZ, 1.0, 1, 2, 2.0, 2

; convert to UHJ format (stereo)
aWre, aWim hilbert a_W
aXre, aXim hilbert a_X
aYre, aYim hilbert a_Y

aWXre = 0.0928*aXre + 0.4699*aWre
aWXim = 0.2550*aXim - 0.1710*aWim

aL = aWXre + aWXim + 0.3277*aYre
aR = aWXre - aWXim - 0.3277*aYre

outs aL, aR

endin

</CsInstruments>
<CsScore>

/* Written by Istvan Varga */
t 0 60

i 1 0.0 8.0
e

</CsScore>
</CsoundSynthesizer>
```

Here is a example of the spat3d opcode that outputs a quadrophonic file. It uses the file *spat3d_quad.csd* [examples/spat3d_quad.csd].

Example 479. Quadrophonic example of the spat3d opcode.

```

<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o spat3d_quad.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

/* Written by Istvan Varga */
sr      = 48000
kr      = 1000
ksmps   = 48
nchnls  = 4

/* room parameters */

idep     = 3      /* early reflection depth          */

itmp     ftgen    1, 0, 64, -2,
/* depth1, depth2, max delay, IR length, idist, seed */ \
idep, 48, -1, 0.01, 0.25, 123,
1, 21.982, 0.05, 0.87, 4000.0, 0.6, 0.7, 2, /* ceiling */ \
1, 1.753, 0.05, 0.87, 3500.0, 0.5, 0.7, 2, /* floor */ \
1, 15.220, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* front */ \
1, 9.317, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* back */ \
1, 17.545, 0.05, 0.87, 5000.0, 0.8, 0.7, 2, /* right */ \
1, 12.156, 0.05, 0.87, 5000.0, 0.8, 0.7, 2 /* left */ \

instr 1

/* some source signal */

a1      phasor 150          ; oscillator
a1      butterbp a1, 500, 200 ; filter
a1      = taninv(a1 * 100)
a2      phasor 3            ; envelope
a2      mirror 40*a2, -100, 5
a2      limit a2, 0, 1
a1      = a1 * a2 * 9000

kazim    line 0, 2.5, 360      ; move sound source around
kdist    line 1, 10, 4         ; distance

; convert polar coordinates
kX       = sin(kazim * 3.14159 / 180) * kdist
kY       = cos(kazim * 3.14159 / 180) * kdist
kZ       = 0

a1       = a1 + 0.000001 * 0.000001 ; avoid underflows

imode    = 2 ; change this to 3 for 8 spk in a cube,
; or 1 for simple stereo

aW, aX, aY, aZ spat3d a1, kX, kY, kZ, 1.0, 1, imode, 2, 2

aW       = aW * 1.4142

; stereo
;
;aL      = aW + aY          /* left          */
;aR      = aW - aY          /* right         */

; quad (square)
;
aFL      = aW + aX + aY      /* front left    */
aFR      = aW + aX - aY      /* front right   */
aRL      = aW - aX + aY      /* rear left     */
aRR      = aW - aX - aY      /* rear right    */

; eight channels (cube)
;
;aUFL    = aW + aX + aY + aZ /* upper front left */
;aUFR    = aW + aX - aY + aZ /* upper front right */
;aURL    = aW - aX + aY + aZ /* upper rear left  */
;aURR    = aW - aX - aY + aZ /* upper rear right */
;aLFL    = aW + aX + aY - aZ /* lower front left  */
;aLFR    = aW + aX - aY - aZ /* lower front right */
;aLRL    = aW - aX + aY - aZ /* lower rear left   */
;aLRR    = aW - aX - aY - aZ /* lower rear right  */

```



```
;aLRR = aW - aX - aY - aZ /* lower rear right */
outq aFL, aFR, aRL, aRR
endin

</CsInstruments>
<CsScore>

/* Written by Istvan Varga */
t 0 60
i 1 0 10
e

</CsScore>
</CsoundSynthesizer>
```

See Also

spat3di, spat3dt

Credits

Author: Istvan Varga
2001

New in version 4.12

Updated April 2002 by Istvan Varga

spat3di

spat3di — Positions the input sound in a 3D space with the sound source position set at i-time.

Description

This opcode positions the input sound in a 3D space, with optional simulation of room acoustics, in various output formats. With *spat3di*, sound source position is set at i-time.

Syntax

```
aW, aX, aY, aZ spat3di ain, iX, iY, iZ, idist, ift, imode [, istor]
```

Initialization

iX -- Sound source X coordinate in meters (positive: right, negative: left)

iY -- Sound source Y coordinate in meters (positive: front, negative: back)

iZ -- Sound source Z coordinate in meters (positive: up, negative: down)

idist -- For modes 0 to 3, *idist* is the unit circle distance in meters. For mode 4, *idist* is the distance between microphones.

The following formulas describe amplitude and delay as a function of sound source distance from microphone(s):

$$\text{amplitude} = 1 / (0.1 + \text{distance})$$

$$\text{delay} = \text{distance} / 340 \text{ (in seconds)}$$

Distance can be calculated as:

$$\text{distance} = \sqrt{iX^2 + iY^2 + iZ^2}$$

In Mode 4, distance can be calculated as:

$$\begin{aligned} \text{distance from left mic} &= \sqrt{(iX + idist/2)^2 + iY^2 + iZ^2} \\ \text{distance from right mic} &= \sqrt{(iX - idist/2)^2 + iY^2 + iZ^2} \end{aligned}$$

With *spat3d* the distance between the sound source and any microphone should be at least $(340 * 18) / \text{sr}$ meters. Shorter distances will work, but may produce artifacts in some cases. There is no such limitation for *spat3di* and *spat3dt*.

Sudden changes or discontinuities in sound source location can result in pops or clicks. Very fast movement may also degrade quality.

ift -- Function table storing room parameters (for free field spatialization, set it to zero or negative).

Table size is 54. The values in the table are:

Room Parameter	Purpose
0	Early reflection recursion depth (0 is the sound source, 1 is the first reflection etc.) for spat3d and spat3di. The number of echoes for four walls (front, back, right, left) is: $N = (2 * R + 2) * R$. If all six walls are enabled: $N = (((4 * R + 6) * R + 8) * R) / 3$
1	Late reflection recursion depth (used by spat3dt only). spat3dt skips early reflections and renders echoes up to this level. If early reflection depth is negative, spat3d and spat3di will output zero, while spat3dt will start rendering from the sound source.
2	imdel for spat3d. Overrides opcode parameter if non-negative.
3	irlen for spat3dt. Overrides opcode parameter if non-negative.
4	idist value. Overrides opcode parameter if ≥ 0 .
5	Random seed (0 - 65535) -1 seeds from current time.
6 - 53	wall parameters (w = 6: ceil, w = 14: floor, w = 22: front, w = 30: back, w = 38: right, w = 46: left)
w + 0	Enable reflections from this wall (0: no, 1: yes)
w + 1	Wall distance from listener (in meters)
w + 2	Randomization of wall distance (0 - 1) (in units of $1 / (\text{wall distance})$)
w + 3	Reflection level (-1 - 1)
w + 4	Parametric equalizer frequency in Hz.
w + 5	Parametric equalizer level (1.0: no filtering)
w + 6	Parametric equalizer Q (0.7071: no resonance)
w + 7	Parametric equalizer mode (0: peak EQ, 1: low shelf, 2: high shelf)

imode -- Output mode

- 0: B format with W output only (mono)

aout = aW

- 1: B format with W and Y output (stereo)

aleft = aW + 0.7071*aY
aright = aW - 0.7071*aY

- 2: B format with W, X, and Y output (2D). This can be converted to UHJ:

aWre, aWim hilbert aW
aXre, aXim hilbert aX

```
aYre, aYim  hilbert aY
aWXr  = 0.0928*aXre + 0.4699*aWre
aWXiYr = 0.2550*aXim - 0.1710*aWim + 0.3277*aYre
aleft  = aWXr + aWXiYr
aright = aWXr - aWXiYr
```

- 3: B format with all outputs (3D)
- 4: Simulates a pair of microphones (stereo output)

```
aW  butterlp aW, ifreq  ; recommended values for ifreq
aY  butterlp aY, ifreq  ; are around 1000 Hz
aleft = aW + aX
aright = aY + aZ
```

Mode 0 is the cheapest to calculate, while mode 4 is the most expensive.

In Mode 4, The optional lowpass filters can change the frequency response depending on direction. For example, if the sound source is located left to the listener then the high frequencies are attenuated in the right channel and slightly increased in the left. This effect can be disabled by not using filters. You can also experiment with other filters (tone etc.) for better effect.

Note that mode 4 is most useful for listening with headphones, and is also more expensive to calculate than the B-format (0 to 3) modes. The *idist* parameter in this case sets the distance between left and right microphone; for headphones, values between 0.2 - 0.25 are recommended, although higher settings up to 0.4 may be used for wide stereo effects.

More information about B format can be found here: http://www.york.ac.uk/inst/mustech/3d_audio/ambis2.htm

istor (optional, default=0) -- Skip initialization if non-zero (default: 0).

Performance

ain -- Input signal

aW, *aX*, *aY*, *aZ* -- Output signals

	mode 0	mode 1	mode 2	mode 3	mode 4
aW	W out	W out	W out	W out	left chn / low freq.
aX	0	0	X out	X out	left chn / high freq.
aY	0	Y out	Y out	Y out	right chn / low freq.
aZ	0	0	0	Z out	right chn / high fr.

If you encounter very slow performance (up to 100 times slower), it may be caused by denormals (this is also true of many other IIR opcodes, including *butterlp*, *pareq*, *hilbert*, and many others). Underflows can be avoided by:

- Using the *denorm* opcode on *ain* before *spat3di*.

- mixing low level DC or noise to the input signal, e.g.

atmp rnd31 1/1e24, 0, 0

aW, aX, aY, aZ spat3di ain + atmp, ...

or

aW, aX, aY, aZ spa3di ain + 1/1e24, ...

- reducing *irlen* in the case of *spat3dt* (which does not have an input signal). A value of about 0.005 is suitable for most uses, although it also depends on EQ settings. If the equalizer is not used, “irlen” can be set to 0.

Examples

See the examples for *spat3d*.

See Also

spat3d, *spat3dt*

Credits

Author: Istvan Varga
2001

New in version 4.12

Updated April 2002 by Istvan Varga

spat3dt

spat3dt — Can be used to render an impulse response for a 3D space at i-time.

Description

This opcode positions the input sound in a 3D space, with optional simulation of room acoustics, in various output formats. *spat3dt* can be used to render the impulse response at i-time, storing output in a function table, suitable for convolution.

Syntax

```
spat3dt ioutft, iX, iY, iZ, idist, ift, imode, irlen [, iftnocl]
```

Initialization

ioutft -- Output ftable number for spat3dt. W, X, Y, and Z outputs are written interleaved to this table. If the table is too short, output will be truncated.

iX -- Sound source X coordinate in meters (positive: right, negative: left)

iY -- Sound source Y coordinate in meters (positive: front, negative: back)

iZ -- Sound source Z coordinate in meters (positive: up, negative: down)

idist -- For modes 0 to 3, *idist* is the unit circle distance in meters. For mode 4, *idist* is the distance between microphones.

The following formulas describe amplitude and delay as a function of sound source distance from microphone(s):

$$\text{amplitude} = 1 / (0.1 + \text{distance})$$

$$\text{delay} = \text{distance} / 340 \text{ (in seconds)}$$

Distance can be calculated as:

$$\text{distance} = \sqrt{iX^2 + iY^2 + iZ^2}$$

In Mode 4, distance can be calculated as:

$$\begin{aligned} \text{distance from left mic} &= \sqrt{(iX + idist/2)^2 + iY^2 + iZ^2} \\ \text{distance from right mic} &= \sqrt{(iX - idist/2)^2 + iY^2 + iZ^2} \end{aligned}$$

With *spat3d* the distance between the sound source and any microphone should be at least $(340 * 18) / \text{sr}$ meters. Shorter distances will work, but may produce artifacts in some cases. There is no such limitation for *spat3di* and *spat3dt*.

Sudden changes or discontinuities in sound source location can result in pops or clicks. Very fast movement may also degrade quality.

ift -- Function table storing room parameters (for free field spatialization, set it to zero or negative). Table size is 54. The values in the table are:

Room Parameter	Purpose
0	Early reflection recursion depth (0 is the sound source, 1 is the first reflection etc.) for spat3d and spat3di. The number of echoes for four walls (front, back, right, left) is: $N = (2 * R + 2) * R$. If all six walls are enabled: $N = (((4 * R + 6) * R + 8) * R) / 3$
1	Late reflection recursion depth (used by spat3dt only). spat3dt skips early reflections and renders echoes up to this level. If early reflection depth is negative, spat3d and spat3di will output zero, while spat3dt will start rendering from the sound source.
2	imdel for spat3d. Overrides opcode parameter if non-negative.
3	irlen for spat3dt. Overrides opcode parameter if non-negative.
4	idist value. Overrides opcode parameter if ≥ 0 .
5	Random seed (0 - 65535) -1 seeds from current time.
6 - 53	wall parameters (w = 6: ceil, w = 14: floor, w = 22: front, w = 30: back, w = 38: right, w = 46: left)
w + 0	Enable reflections from this wall (0: no, 1: yes)
w + 1	Wall distance from listener (in meters)
w + 2	Randomization of wall distance (0 - 1) (in units of $1 / (\text{wall distance})$)
w + 3	Reflection level (-1 - 1)
w + 4	Parametric equalizer frequency in Hz.
w + 5	Parametric equalizer level (1.0: no filtering)
w + 6	Parametric equalizer Q (0.7071: no resonance)
w + 7	Parametric equalizer mode (0: peak EQ, 1: low shelf, 2: high shelf)

imode -- Output mode

- 0: B format with W output only (mono)

aout = aW

- 1: B format with W and Y output (stereo)

aleft = aW + 0.7071*aY
aright = aW - 0.7071*aY

- 2: B format with W, X, and Y output (2D). This can be converted to UHJ:

```
aWre, aWim    hilbert aW
aXre, aXim    hilbert aX
aYre, aYim    hilbert aY
aWXr  = 0.0928*aXre + 0.4699*aWre
aWXiYr = 0.2550*aXim - 0.1710*aWim + 0.3277*aYre
aleft  = aWXr + aWXiYr
aright = aWXr - aWXiYr
```

- 3: B format with all outputs (3D)
- 4: Simulates a pair of microphones (stereo output)

```
aW    butterlp aW, ifreq    ; recommended values for ifreq
aY    butterlp aY, ifreq    ; are around 1000 Hz
aleft = aW + aX
aright = aY + aZ
```

Mode 0 is the cheapest to calculate, while mode 4 is the most expensive.

In Mode 4, The optional lowpass filters can change the frequency response depending on direction. For example, if the sound source is located left to the listener then the high frequencies are attenuated in the right channel and slightly increased in the left. This effect can be disabled by not using filters. You can also experiment with other filters (tone etc.) for better effect.

Note that mode 4 is most useful for listening with headphones, and is also more expensive to calculate than the B-format (0 to 3) modes. The *idist* parameter in this case sets the distance between left and right microphone; for headphones, values between 0.2 - 0.25 are recommended, although higher settings up to 0.4 may be used for wide stereo effects.

More information about B format can be found here: http://www.york.ac.uk/inst/mustech/3d_audio/ambis2.htm

irlen -- Impulse response length of echoes (in seconds). Depending on filter parameters, values around 0.005-0.01 are suitable for most uses (higher values result in more accurate output, but slower rendering)

iftnocl (optional, default=0) -- Do not clear output ftable (mix to existing data) if set to 1, clear table before writing if set to 0 (default: 0).

Examples

See the examples for *spat3d*.

See Also

spat3d, *spat3di*

Credits

Author: Istvan Varga
2001

New in version 4.12

Updated April 2002 by Istvan Varga

spdist

spdist — Calculates distance values from xy coordinates.

Description

spdist uses the same xy data as *space*, also either from a text file using *Gen28* or from x and y arguments given to the unit directly. The purpose of this unit is to make available the values for distance that are calculated from the xy coordinates.

In the case of *space*, the xy values are used to determine a distance which is used to attenuate the signal and prepare it for use in *spsend*. But it is also useful to have these values for distance available to scale the frequency of the signal before it is sent to the *space* unit.

Syntax

```
k1 spdist ifn, ktime, kx, ky
```

Initialization

ifn -- number of the stored function created using *Gen28*. This function generator reads a text file which contains sets of three values representing the xy coordinates and a time-tag for when the signal should be placed at that location. The file should look like:

0	-1	1
1	1	1
2	4	4
2.1	-4	-4
3	10	-10
5	-40	0

If that file were named "move" then the *Gen28* call in the score would like:

```
f1 0 0 28 "move"
```

Gen28 takes 0 as the size and automatically allocates memory. It creates values to 10 milliseconds of resolution. So in this case there will be 500 values created by interpolating X1 to X2 to X3 and so on, and Y1 to Y2 to Y3 and so on, over the appropriate number of values that are stored in the function table. In the above example, the sound will begin in the left front, over 1 second it will move to the right front, over another second it move further into the distance but still in the left front, then in just 1/10th of a second it moves to the left rear, a bit distant. Finally over the last .9 seconds the sound will move to the right rear, moderately distant, and it comes to rest between the two left channels (due west!), quite distant. Since the values in the table are accessed through the use of a time-pointer in the *space* unit, the actual timing can be made to follow the file's timing exactly or it can be made to go faster or slower through the same trajectory. If you have access to the GUI that allows one to draw and edit the files, there is no need to create the text files manually. But as long as the file is ASCII and in the format shown above, it doesn't matter how it is made!

IMPORTANT: If *ifn* is 0 then *space* will take its values for the xy coordinates from *kx* and *ky*.

Performance

The configuration of the xy coordinates in space places the signal in the following way:

- a1 is -1, 1
- a2 is 1, 1
- a3 is -1, -1
- a4 is 1, -1

This assumes a loudspeaker set up as a1 is left front, a2 is right front, a3 is left back, a4 is right back. Values greater than 1 will result in sounds being attenuated, as if in the distance. *space* considers the speakers to be at a distance of 1; smaller values of xy can be used, but *space* will not amplify the signal in this case. It will, however balance the signal so that it can sound as if it were within the 4 speaker *space*. x=0, y=1, will place the signal equally balanced between left and right front channels, x=y=0 will place the signal equally in all 4 channels, and so on. Although there must be 4 output signals from *space*, it can be used in a 2 channel orchestra. If the xy's are kept so that $Y \geq 1$, it should work well to do panning and fixed localization in a stereo field.

ktime -- index into the table containing the xy coordinates. If used like:

```
ktime      line 0, 5, 5
a1, a2, a3, a4 space asig, 1, ktime, ...
```

with the file "move" described above, the speed of the signal's movement will be exactly as described in that file. However:

```
ktime      line 0, 10, 5
```

the signal will move at half the speed specified. Or in the case of:

```
ktime      line 5, 15, 0
```

the signal will move in the reverse direction as specified and 3 times slower! Finally:

```
ktime      line 2, 10, 3
```

will cause the signal to move only from the place specified in line 3 of the text file to the place specified in line 5 of the text file, and it will take 10 seconds to do it.

kx, *ky* -- when *ifn* is 0, *space* and *spdist* will use these values as the XY coordinates to localize the signal.

Examples

```
instr 1
  asig      ;some audio signal
  ktime     line 0, p3, p10
```

```
a1, a2, a3, a4      space asig,1, ktime, .1
ar1, ar2, ar3, ar4 ssend

ga1 = ga1+ar1
ga2 = ga2+ar2
ga3 = ga3+ar3
ga4 = ga4+ar4

                                outq a1, a2, a3, a4
endin

instr 99 ; reverb instrument

a1 reverb2 ga1, 2.5, .5
a2 reverb2 ga2, 2.5, .5
a3 reverb2 ga3, 2.5, .5
a4 reverb2 ga4, 2.5, .5

    outq a1, a2, a3, a4
ga1=0
ga2=0
ga3=0
ga4=0
```

In the above example, the signal, *asig*, is moved according to the data in Function #1 indexed by *ktime*. *space* sends the appropriate amount of the signal internally to *ssend*. The outputs of the *ssend* are added to global accumulators in a common Csound style and the global signals are used as inputs to the reverb units in a separate instrument.

space can be useful for quad and stereo panning as well as fixed placement of sounds anywhere between two loudspeakers. Below is an example of the fixed placement of sounds in a stereo field using xy values from the score instead of a function table.

```
instr 1
...
a1, a2, a3, a4      space asig, 0, 0, .1, p4, p5
ar1, ar2, ar3, ar4 ssend

ga1=ga1+ar1
ga2=ga2+ar2

                                outs  a1, a2
endin

instr 99 ; reverb...
....
endin
```

A few notes: p4 and p5 are the X and Y values

```
;place the sound in the left speaker and near
i1 0 1 -1 1
;place the sound in the right speaker and far
i1 1 1 45 45
;place the sound equally between left and right and in the middle ground distance
i1 2 1 0 12
e
```

The next example shows a simple intuitive use of the distance values returned by *spdist* to simulate Doppler shift.

```
ktime      line 0, p3, 10
kdist      spdist 1, ktime
kfreq = (ifreq * 340) / (340 + kdist)
asig       oscili iamp, kfreq, 1
```

```
a1, a2, a3, a4      space asig, 1, ktime, .1  
ar1, ar2, ar3, ar4 spsend
```

The same function and time values are used for both *spdist* and *space*. This insures that the distance values used internally in the *space* unit will be the same as those returned by *spdist* to give the impression of a Doppler shift!

See Also

space, spsend

Credits

Author: Richard Karpen
Seattle, WA USA
1998

New in Csound version 3.48

specaddm

specaddm — Perform a weighted add of two input spectra.

Description

Perform a weighted add of two input spectra.

Syntax

```
wsig specaddm wsig1, wsig2 [, imul2]
```

Initialization

imul2 (optional, default=0) -- if non-zero, scale the *wsig2* magnitudes before adding. The default value is 0.

Performance

wsig1 -- the first input spectra.

wsig2 -- the second input spectra.

Do a weighted add of two input spectra. For each channel of the two input spectra, the two magnitudes are combined and written to the output according to:

$$\text{magout} = \text{mag1in} + \text{mag2in} * \text{imul2}$$

The operation is performed whenever the input *wsig1* is sensed to be new. This unit will (at Initialization) verify the consistency of the two spectra (equal size, equal period, equal mag types).

Examples

```
wsig2    specdiff      wsig1      ; sense onsets
wsig3    specfilt      wsig2, 2    ; absorb slowly
          specdisp      wsig2, .1   ; & display both spectra
          specdisp      wsig3, .1
```

See Also

specdiff, specfilt, spechist, specscal

specdiff

specdiff — Finds the positive difference values between consecutive spectral frames.

Description

Finds the positive difference values between consecutive spectral frames.

Syntax

```
wsig specdiff wsign
```

Performance

wsig -- the output spectrum.

wsign -- the input spectra.

Finds the positive difference values between consecutive spectral frames. At each new frame of *wsign*, each magnitude value is compared with its predecessor, and the positive changes written to the output spectrum. This unit is useful as an energy onset detector.

Examples

```
wsig2    specdiff      wsig1      ; sense onsets
wsig3    specfilt      wsig2, 2    ; absorb slowly
          specdisp      wsig2, .1   ; & display both spectra
          specdisp      wsig3, .1
```

See Also

specaddm, *specfilt*, *speechist*, *specscal*

specdisp

specdisp — Displays the magnitude values of the spectrum.

Description

Displays the magnitude values of the spectrum.

Syntax

```
specdisp wsig, iprd [, iwtflg]
```

Initialization

iprd -- the period, in seconds, of each new display.

iwtflg (optional, default=0) -- wait flag. If non-zero, hold each display until released by the user. The default value is 0 (no wait).

Performance

wsig -- the input spectrum.

Displays the magnitude values of spectrum *wsig* every *iprd* seconds (rounded to some integral number of *wsig*'s originating *iprd*).

Examples

```
ksum      specsum  wsig, 1      ; sum the spec bins, and ksmooth
          if      ksum < 2000  kgoto zero ; if sufficient amplitude
koct      specptrk wsig          ; pitch-track the signal
          kgoto    contin
zero:
  koct      =      0
contin:
```

See Also

specsum

specfilt

specfilt — Filters each channel of an input spectrum.

Description

Filters each channel of an input spectrum.

Syntax

```
wsig specfilt wsignin, ifhtim
```

Initialization

ifhtim -- half-time constant.

Performance

wsignin -- the input spectrum.

Filters each channel of an input spectrum. At each new frame of *wsignin*, each magnitude value is injected into a 1st-order lowpass recursive filter, whose half-time constant has been initially set by sampling the ftable *ifhtim* across the (logarithmic) frequency space of the input spectrum. This unit effectively applies a *persistence* factor to the data occurring in each spectral channel, and is useful for simulating the *energy integration* that occurs during auditory perception. It may also be used as a time-attenuated running *histogram* of the spectral distribution.

Examples

```
wsig2  specdiff      wsig1      ; sense onsets
wsig3  specfilt      wsig2, 2    ; absorb slowly
        specdisp      wsig2, .1  ; & display both spectra
        specdisp      wsig3, .1
```

See Also

specaddm, *specdiff*, *spechist*, *specscal*

spechist

spechist — Accumulates the values of successive spectral frames.

Description

Accumulates the values of successive spectral frames.

Syntax

```
wsig spechist wsignin
```

Performance

wsignin -- the input spectra.

Accumulates the values of successive spectral frames. At each new frame of *wsignin*, the accumulations-to-date in each magnitude track are written to the output spectrum. This unit thus provides a running *histogram* of spectral distribution.

Examples

```
wsig2    specdiff          wsig1          ; sense onsets
wsig3    specfilt         wsig2, 2         ; absorb slowly
          specdisp        wsig2, .1        ; & display both spectra
          specdisp        wsig3, .1
```

See Also

specaddm, *specdiff*, *specfilt*, *specscal*

specptrk

specptrk — Estimates the pitch of the most prominent complex tone in the spectrum.

Description

Estimate the pitch of the most prominent complex tone in the spectrum.

Syntax

```
koct, kamp specptrk wsig, kvar, ilo, ihi, istr, idbthresh, inptls, \  
irolloff [, iodd] [, iconfs] [, interp] [, ifprd] [, iwtflg]
```

Initialization

ilo, ihi, istr -- pitch range conditioners (low, high, and starting) expressed in decimal octave form.

idbthresh -- energy threshold (in decibels) for pitch tracking to occur. Once begun, tracking will be continuous until the energy falls below one half the threshold (6 dB down), whence the *koct* and *kamp* outputs will be zero until the full threshold is again surpassed. *idbthresh* is a guiding value. At initialization it is first converted to the *idbout* mode of the source spectrum (and the 6 dB down point becomes .5, .25, or 1/root 2 for modes 0, 2 and 3). The values are also further scaled to allow for the weighted partial summation used during correlation. The actual thresholding is done using the internal weighted and summed *kamp* value that is visible as the second output parameter.

inptls, irolloff -- number of harmonic partials used as a matching template in the spectrally-based pitch detection, and an amplitude rolloff for the set expressed as some fraction per octave (linear, so don't roll off to negative). Since the partials and rolloff fraction can affect the pitch following, some experimentation will be useful: try 4 or 5 partials with .6 rolloff as an initial setting; raise to 10 or 12 partials with rolloff .75 for complex timbres like the bassoon (weak fundamental). Computation time is dependent on the number of partials sought. The maximum number is 16.

ioodd (optional) -- if non-zero, employ only odd partials in the above set (e.g. *inptls* of 4 would employ partials 1,3,5,7). This improves the tracking of some instruments like the clarinet. The default value is 0 (employ all partials).

iconfs (optional) -- number of confirmations required for the pitch tracker to jump an octave, prorated for fractions of an octave (i.e. the value 12 implies a semitone change needs 1 confirmation (two hits) at the *spectrum* generating *iprd*). This parameter limits spurious pitch analyses such as octave errors. A value of 0 means no confirmations required; the default value is 10.

interp (optional) -- if non-zero, interpolate each output signal (*koct*, *kamp*) between incoming *wsig* frames. The default value is 0 (repeat the signal values between frames).

ifprd (optional) -- if non-zero, display the internally computed spectrum of candidate fundamentals. The default value is 0 (no display).

iwtflg (optional) -- wait flag. If non-zero, hold each display until released by the user. The default value is 0 (no wait).

Performance

At note initialization this unit creates a template of *inptls* harmonically related partials (odd partials, if *ioodd* non-zero) with amplitude rolloff to the fraction *irolloff* per octave. At each new frame of *wsig*, the spectrum is cross-correlated with this template to provide an internal spectrum of candidate fundamentals (optionally displayed). A likely pitch/amp pair (*koct*, *kamp*, in decimal octave and summed *idbout* form) is then estimated. *koct* varies from the previous *koct* by no more than plus or minus *kvar* decimal octave units. It is also guaranteed to lie within the hard limit range *ilo* -- *ihi*

(decimal octave low and high pitch). *kvar* can be dynamic, e.g. onset amp dependent. Pitch resolution uses the originating *spectrum ifrqs* bins/octave, with further parabolic interpolation between adjacent bins. Settings of root magnitude, *ifrqs* = 24, *iq* = 15 should capture all the inflections of interest. Between frames, the output is either repeated or interpolated at the k-rate. (See *spectrum*.)

Examples

```
al,a2    ins                                ; read a stereo clarinet input
krms      rms                               ; find a monaural rms value
kvar      =                                ; & use to gate the pitch var
wsig      spectrum                          ; get a 7-oct spectrum, 24 bins
          specdisp                          ; display this and now estimate
koct,ka   spectrk                          ; the pch and amp
aosc      oscil                             ; & generate \ new tone with
koct      =                                ; replace non pitch with low C
          display                          ; & display the pitch track
          display                          ; plus the summed root mag
          outs                             ; output 1 original and 1 new

          al, 20
          0.6 + krms/8000
          al, .01, 7, 24, 15, 0, 3
          wsig, .2
          wsig, kvar, 7.0, 10, 9, 20, 4, .7, 1, 5, 1, .2
          ka*ka*10, cpsoct(koct),2
          (kcoct<7.0?7.0:kcoct)
          kcoct-7.0, .25, 20
          ka, .25, 20
          al, aosc
```

specscal

specscal — Scales an input spectral datablock with spectral envelopes.

Description

Scales an input spectral datablock with spectral envelopes.

Syntax

```
wsig specscal wsignin, ifscale, ifthresh
```

Initialization

ifscale -- scale function table. A function table containing values by which a value's magnitude is rescaled.

ifthresh -- threshold function table. If *ifthresh* is non-zero, each magnitude is reduced by its corresponding table-value (to not less than zero)

Performance

wsig -- the output spectrum

wsignin -- the input spectra

Scales an input spectral datablock with spectral envelopes. Function tables *ifthresh* and *ifscale* are initially sampled across the (logarithmic) frequency space of the input spectrum; then each time a new input spectrum is sensed the sampled values are used to scale each of its magnitude channels as follows: if *ifthresh* is non-zero, each magnitude is reduced by its corresponding table-value (to not less than zero); then each magnitude is rescaled by the corresponding *ifscale* value, and the resulting spectrum written to *wsig*.

Examples

```
wsig2    specdiff      wsig1      ; sense onsets
wsig3    specfilt      wsig2, 2    ; absorb slowly
          specdisp      wsig2, .1   ; & display both spectra
          specdisp      wsig3, .1
```

See Also

specaddm, specdiff, specfilt, spechist

specsum

specsum — Sums the magnitudes across all channels of the spectrum.

Description

Sums the magnitudes across all channels of the spectrum.

Syntax

```
ksum specsum wsig [, interp]
```

Initialization

interp (optional, default-0) -- if non-zero, interpolate the output signal (*koct* or *ksum*). The default value is 0 (repeat the signal value between changes).

Performance

ksum -- the output signal.

wsig -- the input spectrum.

Sums the magnitudes across all channels of the spectrum. At each new frame of *wsig*, the magnitudes are summed and released as a scalar *ksum* signal. Between frames, the output is either repeated or interpolated at the k-rate. This unit produces a k-signal summation of the magnitudes present in the spectral data, and is thereby a running measure of its moment-to-moment overall strength.

Examples

```
ksum    specsum    wsig, 1          ; sum the spec bins, and ksmooth
        if         ksum < 2000    kgoto zero    ; if sufficient amplitude
koct    specptrk    wsig           ; pitch-track the signal
        kgoto      contin
zero:
koct    =           0              ; else output zero
contin:
```

See Also

specdisp

spectrum

spectrum — Generate a constant-Q, exponentially-spaced DFT.

Description

Generate a constant-Q, exponentially-spaced DFT across all octaves of a multiply-downsampled control or audio input signal.

Syntax

```
wsig spectrum xsig, iprd, iocts, ifrqa [, iq] [, ihann] [, idbout] \  
      [, idsprd] [, idsinrs]
```

Initialization

ihann (optional) -- apply a Hamming or Hanning window to the input. The default is 0 (Hamming window)

idbout (optional) -- coded conversion of the DFT output:

- 0 = magnitude
- 1 = dB
- 2 = mag squared
- 3 = root magnitude

The default value is 0 (magnitude).

idsprd (optional) -- if non-zero, display the composite downsampling buffer every *idsprd* seconds. The default value is 0 (no display).

idsines (optional) -- if non-zero, display the Hamming or Hanning windowed sinusoids used in DFT filtering. The default value is 0 (no sinusoid display).

Performance

This unit first puts signal *asig* or *ksig* through *iocts* of successive octave decimation and down-sampling, and preserves a buffer of down-sampled values in each octave (optionally displayed as a composite buffer every *idsprd* seconds). Then at every *iprd* seconds, the preserved samples are passed through a filter bank (*ifrqs* parallel filters per octave, exponentially spaced, with frequency/bandwidth Q of *iq*), and the output magnitudes optionally converted (*idbout*) to produce a band-limited spectrum that can be read by other units.

The stages in this process are computationally intensive, and computation time varies directly with *iocts*, *ifrqs*, *iq*, and inversely with *iprd*. Settings of *ifrqs* = 12, *iq* = 10, *idbout* = 3, and *iprd* = .02 will normally be adequate, but experimentation is encouraged. *ifrqs* currently has a maximum of 120 divisions per octave. For audio input, the frequency bins are tuned to coincide with A440.

This unit produces a self-defining spectral datablock *wsig*, whose characteristics used (*iprd*, *iocts*, *ifrqs*, *idbout*) are passed via the data block itself to all derivative *wsigs*. There can be any number of spectrum units in an instrument or orchestra, but all *wsig* names must be unique.

Examples

```
asig in                                ; get external audio
wsig spectrum asig,.02,6,12,33,0,1,1  ; downsample in 6 octs & calc a 72 pt dft (Q 33, dB out) eve
```


splitrig

splitrig — Split a trigger signal

Description

splitrig splits a trigger signal (i.e. a timed sequence of control-rate impulses) into several channels following a structure designed by the user.

Syntax

```
splitrig ktrig, kndx, imaxtics, ifn, kout1 [,kout2,...,koutN]
```

Initialization

imaxtics - number of tics belonging to largest pattern

ifn - number of table containing channel-data structuring

Performance

asig - incoming (input) signal

ktrig - trigger signal

The *splitrig* opcode splits a trigger signal into several output channels according to one or more patterns provided by the user. Normally the regular timed trigger signal generated by metro opcode is used to be transformed into rhythmic pattern that can trig several independent melodies or percussion riffs. But you can also start from non-isocronous trigger signals. This allows to use some "interpretative" and less "mechanic" groove variations. Patterns are looped and each *numtics_of_pattern_N* the cycle is repeated.

The scheme of patterns is defined by the user and is stored into *ifn* table according to the following format:

```
gil ftgen 1,0,1024, -2 \ ; table is generated with GEN02 in this case
\
numtics_of_pattern_1, \ ;pattern 1
  tic1_out1, tic1_out2, ... , tic1_outN,\
  tic2_out1, tic2_out2, ... , tic2_outN,\
  tic3_out1, tic3_out2, ... , tic3_outN,\
  .....
  ticN_out1, ticN_out2, ... , ticN_outN,\
\
numtics_of_pattern_2, \ ;pattern 2
  tic1_out1, tic1_out2, ... , tic1_outN,\
  tic2_out1, tic2_out2, ... , tic2_outN,\
  tic3_out1, tic3_out2, ... , tic3_outN,\
  .....
  ticN_out1, ticN_out2, ... , ticN_outN,\
  .....
\
numtics_of_pattern_N, \ ;pattern N
  tic1_out1, tic1_out2, ... , tic1_outN,\
  tic2_out1, tic2_out2, ... , tic2_outN,\
  tic3_out1, tic3_out2, ... , tic3_outN,\
  .....
  ticN_out1, ticN_out2, ... , ticN_outN,\
```

This scheme can contain more than one pattern, each one with a different number of rows. Each pattern is preceded by a special row containing a single *numtics_of_pattern_N* field; this field ex-

presses the number of tics that makes up the corresponding pattern. Each pattern's row makes up a tic. Each pattern's column corresponds to a channel, and each field of a row is a number that makes up the value outputted by the corresponding *koutXX* channel (if number is a zero, corresponding output channel will not trigger anything in that particular arguments). Obviously, all rows must contain the same number of fields that must be equal to the number of *koutXX* channel. All patterns must contain the same number of rows, this number must be equal to the largest pattern and is defined by *imaxtics* variable. Even if a pattern has less tics than the largest pattern, it must be made up of the same number of rows, in this case, some of these rows, at the end of the pattern itself, will not be used (and can be set to any value, because it doesn't matter).

The *kndx* variable chooses the number of the pattern to be played, zero indicating the first pattern. Each time the integer part of *kndx* changes, tic counter is reset to zero.

Patterns are looped and each *numtics_of_pattern_N* the cycle is repeated.

examples 4 - calculate average value of *asig* in the time interval

This opcode can be useful in several situations, for example to implement a vu-meter

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

spsend

spsend — Generates output signals based on a previously defined *space* opcode.

Description

spsend depends upon the existence of a previously defined *space*. The output signals from *spsend* are derived from the values given for xy and reverb in the *space* and are ready to be sent to local or global reverb units (see example below).

Syntax

```
a1, a2, a3, a4 spsend
```

Performance

The configuration of the xy coordinates in space places the signal in the following way:

- a1 is -1, 1
- a2 is 1, 1
- a3 is -1, -1
- a4 is 1, -1

This assumes a loudspeaker set up as a1 is left front, a2 is right front, a3 is left back, a4 is right back. Values greater than 1 will result in sounds being attenuated, as if in the distance. *space* considers the speakers to be at a distance of 1; smaller values of xy can be used, but *space* will not amplify the signal in this case. It will, however balance the signal so that it can sound as if it were within the 4 speaker *space*. x=0, y=1, will place the signal equally balanced between left and right front channels, x=y=0 will place the signal equally in all 4 channels, and so on. Although there must be 4 output signals from *space*, it can be used in a 2 channel orchestra. If the xy's are kept so that $Y \geq 1$, it should work well to do panning and fixed localization in a stereo field.

Examples

```
instr 1
  asig      ;some audio signal
  ktime          line 0, p3, p10
  a1, a2, a3, a4  space asig,1, ktime, .1
  ar1, ar2, ar3, ar4 spsend

  ga1 = ga1+ar1
  ga2 = ga2+ar2
  ga3 = ga3+ar3
  ga4 = ga4+ar4

                                outq a1, a2, a3, a4
endin

instr 99 ; reverb instrument

  a1 reverb2 ga1, 2.5, .5
  a2 reverb2 ga2, 2.5, .5
  a3 reverb2 ga3, 2.5, .5
  a4 reverb2 ga4, 2.5, .5

  outq a1, a2, a3, a4
```

```
ga1=0
ga2=0
ga3=0
ga4=0
```

In the above example, the signal, *asig*, is moved according to the data in Function #1 indexed by *ptime*. *space* sends the appropriate amount of the signal internally to *spsend*. The outputs of the *spsend* are added to global accumulators in a common Csound style and the global signals are used as inputs to the reverb units in a separate instrument.

space can be useful for quad and stereo panning as well as fixed placement of sounds anywhere between two loudspeakers. Below is an example of the fixed placement of sounds in a stereo field using xy values from the score instead of a function table.

```
instr 1
...
a1, a2, a3, a4      space asig, 0, 0, .1, p4, p5
ar1, ar2, ar3, ar4 spsend

ga1=ga1+ar1
ga2=ga2+ar2
                        outs  a1, a2
endin

instr 99 ; reverb...
...
endin
```

A few notes: p4 and p5 are the X and Y values

```
;place the sound in the left speaker and near
i1 0 1 -1 1
;place the sound in the right speaker and far
i1 1 1 45 45
;place the sound equally between left and right and in the middle ground distance
i1 2 1 0 12
e
```

The next example shows a simple intuitive use of the distance values returned by *spdist* to simulate Doppler shift.

```
ptime          line 0, p3, 10
kdist          spdist 1, ptime
kfreq = (ifreq * 340) / (340 + kdist)
asig          oscili iamp, kfreq, 1

a1, a2, a3, a4      space asig, 1, ptime, .1
ar1, ar2, ar3, ar4 spsend
```

The same function and time values are used for both *spdist* and *space*. This insures that the distance values used internally in the *space* unit will be the same as those returned by *spdist* to give the impression of a Doppler shift!

See Also

space, *spdist*

Credits

Author: Richard Karpen
Seattle, WA USA
1998

New in Csound version 3.48

sprintf

`sprintf` — printf-style formatted output to a string variable.

Description

sprintf write printf-style formatted output to a string variable, similarly to the C function `sprintf()`. `sprintf` runs at i-time only.

Syntax

```
Sdst sprintf Sfmt, xarg1[, xarg2[, ... ]]
```

```
Sdst sprintfk Sfmt, xarg1[, xarg2[, ... ]]
```

Initialization

Sfmt -- format string, has the same format as in `printf()` and other similar C functions, except length modifiers (l, ll, h, etc.) are not supported. The following conversion specifiers are allowed:

- d, i, o, u, x, X, e, E, f, F, g, G, c, s

xarg1, *xarg2*, ... -- input arguments (max. 30) for format, should be i-rate for all conversion specifiers except %s, which requires a string argument. Integer formats like %d round the input values to the nearest integer.

Performance

Sdst -- output string variable

Example

```
Sname  sprintf "soundin-%04d.wav", ifileno
Smsg    sprintf "The file name is: '%s'", Sname
        puts Smsg, 1
asig soundin Sname
```

See also

sprintfk

Credits

Author: Istvan Varga
2005

sprintfk

sprintfk — printf-style formatted output to a string variable at k-rate.

Description

sprintfk writes printf-style formatted output to a string variable, similarly to the C function `sprintf()`. *sprintfk* runs both at initialization and performance time.

Syntax

```
Sdst sprintfk Sfmt, xarg1[, xarg2[, ... ]]
```

Initialization

Sfmt -- format string, has the same format as in `printf()` and other similar C functions, except length modifiers (l, ll, h, etc.) are not supported. The following conversion specifiers are allowed:

- d, i, o, u, x, X, e, E, f, F, g, G, c, s

xarg1, *xarg2*, ... -- input arguments (max. 30) for format, should be i-rate for all conversion specifiers except %s, which requires a string argument. *sprintfk* also allows k-rate number arguments, but these should still be valid at init time as well (unless *sprintfk* is skipped with *igoto*). Integer formats like %d round the input values to the nearest integer.

Performance

Sdst -- output string variable

Examples

Here is an example of the *sprintfk* opcode. It uses the file *sprintfk.csd* [examples/sprintfk.csd].

Example 480. Example of the *sprintfk* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o sprintfk.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

sr      = 48000
ksmps   = 16
nchnls  = 2
0dbfs   = 1

; Example by Jonathan Murphy 2007

instr 1

sl      = "1"
```

```
S2      = " + 1"
ktrig   init      0
kval    init      2
if (ktrig == 1) then
  S1     strcatk  S1, S2
  kval   = kval + 1
endif
String  sprintfk  "%s = %d", S1, kval
        puts      String, kval
ktrig   metro     1

endin

</CsInstruments>
<CsScore>
i1 0 10
e
</CsScore>
</CsoundSynthesizer>
```

See also

sprintf, puts, strcat

Credits

Author: Istvan Varga
2005
Example by Jonathan Murphy

sqrt

sqrt — Returns a square root value.

Description

Returns the square root of x (x non-negative).

The argument value is restricted for *log*, *log10*, and *sqrt*.

Syntax

sqrt(*x*) (no rate restriction)

where the argument within the parentheses may be an expression. Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

Examples

Here is an example of the sqrt opcode. It uses the file *sqrt.csd* [examples/sqrt.csd].

Example 481. Example of the sqrt opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o sqrt.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  i1 = sqrt(64)
  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
instr 1:  i1 = 8.000
```

See Also

abs, exp, frac, int, log, log10, i

Credits

Example written by Kevin Conder.

sr

sr — Sets the audio sampling rate.

Description

These statements are global value *assignments*, made at the beginning of an orchestra, before any instrument block is defined. Their function is to set certain *reserved symbol variables* that are required for performance. Once set, these reserved symbols can be used in expressions anywhere in the orchestra.

Syntax

```
sr = iarg
```

Initialization

sr = (optional) -- set sampling rate to *iarg* samples per second per channel. The default value is 44100.

In addition, any *global variable* [48] can be initialized by an *init-time assignment* anywhere before the first *instr statement*. All of the above assignments are run as instrument 0 (i-pass only) at the start of real performance.

Beginning with Csound version 3.46, *sr* may be omitted. The sample rate will be calculated from *kr* and *ksmps*, but this must evaluate to an integer. If none of these global values is defined, the sample rate will default to 44100. You will usually want to use a value that your soundcard supports, like 44100 or 48000, otherwise, the audio generated by csound may be unplayable, or you will get an error if you attempt to run in real-time. You may naturally use a sample rate like 96000, for off-line rendering even if your soundcard doesn't support it. Csound will generate a valid file that can be played on capable systems.

Examples

```
sr = 10000
kr = 500
ksmps = 20
gil = sr/2.
ga init 0
itranspose = octpch(.01)
```

See Also

kr, *ksmps*, *nchnls*

stack

stack — Initializes the stack.

Description

Initializes and sets the size of the global stack.

Syntax

```
stack iStackSize
```

Initialization

iStackSize - size of the stack in bytes.

Performance

Csound implements a single global stack. Initializing the stack with the *stack* opcode is not required - it is optional, and if not done, the first use of *push* or *push_f* will automatically create a stack of 32768 bytes. Otherwise, *stack* is normally called from the orchestra header, and takes a stack size parameter in bytes (there is an upper limit of about 16 MB). Once set, the stack size is fixed and cannot be changed during performance.

The global stack works in LIFO order: after multiple *push* calls, *pop* should be used in reverse order.

Each *push* or *pop* operation can work on a "bundle" of multiple variables. When using *pop*, the number, type, and order of items must match those used by the corresponding *push*. That is, after a 'push Sfoo, ibar', you must call something like 'pop Sbar, ifoo', and not e.g. two separate 'pop' statements.

push and *pop* opcodes can take variables of any type (i-, k-, a- and strings). Variables of type 'a' and 'k' are passed at performance time only, while 'i' and 'S' are passed at init time only.

push/pop for a, k, i, and S types copy data by value. By contrast, *push_f* only pushes a "reference" to the f-signal, and then the corresponding *pop_f* will copy directly from the original variable to its output signal. For this reason, changing the source f-signal of *push_f* before *pop_f* is called is not recommended, and if the instrument instance owning the variable that was passed by *push_f* is deactivated before *pop_f* is called, undefined behavior may occur.

Any stack errors (trying to push when there is no more space, or pop from an empty stack, inconsistent number or type of arguments, etc.) are fatal and terminate performance.

See also

pop, *push*, *pop_f* and *push_f*.

Credits

By: Istvan Varga.

2006

statevar

statevar — State-variable filter.

Description

Statevar is a new digital implementation of the analogue state-variable filter. This filter has four simultaneous outputs: high-pass, low-pass, band-pass and band-reject. This filter uses oversampling for sharper resonance (default: 3 times oversampling). It includes a resonance limiter that prevents the filter from getting unstable.

Syntax

```
ahp,alp,abp,abr statevar ain, kcf, kq [, iosamps, istor]
```

Initialization

iosamps -- number of times of oversampling used in the filtering process. This will determine the maximum sharpness of the filter resonance (Q). More oversampling allows higher Qs, less oversampling will limit the resonance. The default is 3 times (iosamps=0).

istor --initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

ahp -- high-pass output signal.

alp -- low-pass output signal.

abp -- band-pass signal.

abr -- band-reject signal.

asig -- input signal.

kcf -- filter cutoff frequency

kq -- filter Q. This value is limited internally depending on the frequency and the number of times of oversampling used in the process (3-times oversampling by default).

Examples

Example 482. Example

```
kenv          linseg 0,0.1,1, p3-0.2,1, 0.1, 0
asig          buzz 16000*kenv, 100, 100, 1;
kf           expseg 100, p3/2, 5000, p3/2, 1000
ahp,alp,abp,abr statevar asig, kf, 200

              outs alp,ahp
```

Credits

Author: Victor Lazzarini;
January 2005

New plugin in version 5

January 2005.

stix

stix — Semi-physical model of a stick sound.

Description

stix is a semi-physical model of a stick sound. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares stix iamp, idettack [, inum] [, idamp] [, imaxshake]
```

Initialization

iamp -- Amplitude of output. Note: As these instruments are stochastic, this is only a approximation.

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 30.

idamp (optional) -- the damping factor, as part of this equation:

$$\text{damping_amount} = 0.998 + (\text{idamp} * 0.002)$$

The default *damping_amount* is 0.998 which means that the default value of *idamp* is 0. The maximum *damping_amount* is 1.0 (no damping). This means the maximum value for *idamp* is 1.0.

The recommended range for *idamp* is usually below 75% of the maximum value.

imaxshake (optional) -- amount of energy to add back into the system. The value should be in range 0 to 1.

Examples

Here is an example of the *stix* opcode. It uses the file *stix.csd* [examples/stix.csd].

Example 483. Example of the *stix* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o stix.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

;orchestra -----

sr =          44100
kr =          4410
ksmps =        10
nchnls =        1
```

```
instr 01                                ;an example of stix
  a1  line 20, p3, 20                    ;preset amplitude increase
  a2  stix p4, 0.01                      ;stix needs a little amp help at these settings
  a3  product a1, a2                    ;increase amplitude
      out a3
      endin

</CsInstruments>
<CsScore>

;score -----

i1 0 1 26000
e

</CsScore>
</CsoundSynthesizer>
```

See Also

cabasa, crunch, sandpaper, sekere

Credits

Author: Perry Cook, part of the PhOLIES (Physically-Oriented Library of Imitated Environmental Sounds)

Adapted by John ffitch

University of Bath, Codemist Ltd.

Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

strchar

strchar — Return the ASCII code of a character in a string

Description

Return the ASCII code of the character in Sstr at ipos (defaults to zero which means the first character), or zero if ipos is out of range. strchar runs at init time only.

Syntax

```
ichr strchar Sstr[, ipos]
```

See also

strchark

Credits

Author: Istvan Varga
2006

strchark

strchark — Return the ASCII code of a character in a string

Description

Return the ASCII code of the character in Sstr at kpos (defaults to zero which means the first character), or zero if kpos is out of range. strchark runs both at init and performance time.

Syntax

```
kchr strchark Sstr[, kpos]
```

See also

strchar

Credits

Author: Istvan Varga
2006

New in version 5.02

strcpy

strcpy — Assign value to a string variable

Description

Assign to a string variable by copying the source which may be a constant or another string variable. strcpy and = copy the string at i-time only.

Syntax

```
Sdst strcpy Ssrc
```

```
Sdst = Ssrc
```

Example

```
Sfoo  strcpy "Hello, world !"  
      puts Sfoo, 1
```

See also

strcpyk

Credits

Author: Istvan Varga
2005

strcpyk

strcpyk — Assign value to a string variable (k-rate)

Description

Assign to a string variable by copying the source which may be a constant or another string variable. *strcpyk* does the assignment both at initialization and performance time.

Syntax

```
Sdst strcpyk Ssrc
```

See also

strcpy

Credits

Author: Istvan Varga
2005

strcat

strcat — Concatenate strings

Description

Concatenate two strings and store the result in a variable. *strcat* runs at i-time only. It is allowed for any of the input arguments to be the same as the output variable.

Syntax

```
Sdst strcat Ssrc1, Ssrc2
```

Example

```
Sname = "beats"  
Sname strcat Sname, ".wav"  
asig soundin Sname
```

See also

strcatk

Credits

Author: Istvan Varga
2005

New in version 5.02

strcatk

strcatk — Concatenate strings (k-rate)

Description

Concatenate two strings and store the result in a variable. *strcatk* does the concatenation both at initialization and performance time. It is allowed for any of the input arguments to be the same as the output variable.

Syntax

```
Sdst strcatk Ssrc1, Ssrc2
```

See also

strcat

Credits

Author: Istvan Varga
2005

New in version 5.02

strcmp

strcmp — Compare strings

Description

Compare strings and set the result to -1, 0, or 1 if the first string is less than, equal to, or greater than the second, respectively. strcmp compares at i-time only.

Syntax

ires **strcmp** S1, S2

See also

strcmpk

Credits

Author: Istvan Varga
2005

strcmpk

strcmp — Compare strings

Description

Compare strings and set the result to -1, 0, or 1 if the first string is less than, equal to, or greater than the second, respectively. *strcmpk* does the comparison both at initialization and performance time.

Syntax

```
kres strcmpk S1, S2
```

See also

strcmp

Credits

Author: Istvan Varga
2005

streson

streson — A string resonator with variable fundamental frequency.

Description

An audio signal is modified by a string resonator with variable fundamental frequency.

Syntax

```
ares streson asig, kfr, ifdbgain
```

Initialization

ifdbgain -- feedback gain, between 0 and 1, of the internal delay line. A value close to 1 creates a slower decay and a more pronounced resonance. Small values may leave the input signal unaffected. Depending on the filter frequency, typical values are $> .9$.

Performance

asig -- the input audio signal.

kfr -- the fundamental frequency of the string.

streson passes the input *asig* through a network composed of comb, low-pass and all-pass filters, similar to the one used in some versions of the Karplus-Strong algorithm, creating a string resonator effect. The fundamental frequency of the “string” is controlled by the k-rate variable *kfr*. This opcode can be used to simulate sympathetic resonances to an input signal.

See *Modal Frequency Ratios* for frequency ratios of real instruments which can be used to determine the values of *kfrq*.

streson is an adaptation of the StringFlt object of the SndObj Sound Object Library developed by the author.

Examples

Here is an example of the streson opcode. It uses the file *streson.csd* [examples/streson.csd].

Example 484. Example of the streson opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o streson.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

```
; Instrument #1.
instr 1
  ; Generate a normal sine wave.
  asig oscils 8000, 440, 1

  ; Vary the fundamental frequency of the string
  ; resonator linearly from 220 to 880 Hertz.
  kfr line 220, p3, 880
  ifdbgain = 0.95

  ; Run our sine wave through the string resonator.
  astres streson asig, kfr, ifdbgain

  ; The resonance can get quite loud.
  ; So we'll clip the signal at 30,000.
  al clip astres, 1, 30000
  out al
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for five seconds.
i 1 0 5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Victor Lazzarini
Music Department
National University of Ireland, Maynooth
Maynooth, Co. Kildare
1998

Example written by Kevin Conder.

New in Csound version 3.494

strget

strget — Set string variable to value from strset table or string p-field

Description

strget sets a string variable at initialization time to the value stored in *strset* table at the specified index, or a string p-field from the score. If there is no string defined for the index, the variable is set to an empty string.

Syntax

Sdst **strget** *indx*

Initialization

indx -- strset index, or score p-field

Sdst -- destination string variable

See also

strset

Credits

Author: Istvan Varga

2005

strindex

strindex — Return the position of the first occurrence of a string in another string

Description

Return the position of the first occurrence of S2 in S1, or -1 if not found. If S2 is empty, 0 is returned. strindex runs at init time only.

Syntax

```
ipos strindex S1, S2
```

See also

strindexk

Credits

Author: Istvan Varga
2006

New in version 5.02

strindexk

strindexk — Return the position of the first occurrence of a string in another string

Description

Return the position of the first occurrence of S2 in S1, or -1 if not found. If S2 is empty, 0 is returned. strindexk runs both at init and performance time.

Syntax

kpos **strindexk** S1, S2

See also

strindex

Credits

Author: Istvan Varga
2006

New in version 5.02

strlen

strlen — Return the length of a string

Description

Return the length of a string, or zero if it is empty. strlen runs at init time only.

Syntax

```
ilen strlen Sstr
```

See also

strlenk

Credits

Author: Istvan Varga
2006

New in version 5.02

strlenk

strlenk — Return the length of a string

Description

Return the length of a string, or zero if it is empty. strlenk runs both at init and performance time.

Syntax

```
klen strlenk Sstr
```

See also

strlen

Credits

Author: Istvan Varga
2006

New in version 5.02

strlower

strlower — Convert a string to lower case

Description

Convert Ssrc to lower case, and write the result to Sdst. strlower runs at init time only.

Syntax

Sdst **strlower** Ssrc

See also

strlowerk

Credits

Author: Istvan Varga
2006

New in version 5.02

strlowerk

strlowerk — Convert a string to lower case

Description

Convert Ssrc to lower case, and write the result to Sdst. strlowerk runs both at init and performance time.

Syntax

Sdst **strlowerk** Ssrc

See also

strlower

Credits

Author: Istvan Varga
2006

New in version 5.02

strrindex

strrindex — Return the position of the last occurrence of a string in another string

Description

Return the position of the last occurrence of S2 in S1, or -1 if not found. If S2 is empty, the length of S1 is returned. strrindex runs at init time only.

Syntax

```
ipos strrindex S1, S2
```

See also

strrindexk

Credits

Author: Istvan Varga
2006

New in version 5.02

strrindexk

strrindexk — Return the position of the last occurrence of a string in another string

Description

Return the position of the last occurrence of S2 in S1, or -1 if not found. If S2 is empty, the length of S1 is returned. strrindexk runs both at init and performance time.

Syntax

kpos **strrindexk** S1, S2

See also

strindex

Credits

Author: Istvan Varga
2006

New in version 5.02

strset

strset — Allows a string to be linked with a numeric value.

Description

Allows a string to be linked with a numeric value.

Syntax

```
strset iarg, istring
```

Initialization

iarg -- the numeric value.

istring -- the alphanumeric string (in double-quotes).

strset (optional) allows a string, such as a filename, to be linked with a numeric value. Its use is optional.

Examples

The following statement, used in the orchestra header, will allow the numeric value 10 to be substituted anywhere the soundfile *asound.wav* is called for.

```
strset 10, "asound.wav"
```

Examples

Here is an example of the strset opcode. It uses the file *strset.csd* [examples/strset.csd].

Example 485. Example of the strset opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 1
nchnls = 1

;Example by Andres Cabrera 2008

; \\n is used to denote "new line"
strset 1, "String 1\\n"
strset 2, "String 2\\n"

instr 1
```

```
Str strget p4
prints Str
endin

</CsInstruments>
<CsScore>
;      p4 is used to select string
i 1 0 1 1
i 1 3 1 2
</CsScore>
</CsoundSynthesizer>
```

See Also

pset and *strget*

strsub

strsub — Extract a substring

Description

Return a substring of the source string. strsub runs at init time only.

Syntax

Sdst **strsub** Ssrc[, istart[, iend]]

Initialization

istart (optional, defaults to 0) -- start position in Ssrc, counting from 0. A negative value means the end of the string.

iend (optional, defaults to -1) -- end position in Ssrc, counting from 0. A negative value means the end of the string. If iend is less than istart, the output is reversed.

Examples

Here is an example of the strsub opcode. It uses the file *strsub.csd* [examples/strsub.csd].

Example 486. Example of the strsub opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      ;;-d      RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o strsub.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
; By: Jonathan Murphy 2007

instr 1
  Smember strget p4

  ; Parse Smember
  istrlen  strlen  Smember
  idelimiter strindex Smember, ":"

  S1      strsub Smember, 0, idelimiter ; "String1"
  S2      strsub Smember, idelimiter + 1, istrlen ; "String2"

  printf "First string: %s\nSecond string: %s\n", 1, S1, S2

endin

</CsInstruments>
<CsScore>
i 1 0 1 "String1:String2"
</CsScore>
</CsoundSynthesizer>
```

See also

strsubk

Credits

Author: Istvan Varga
2006

strsubk

strsubk — Extract a substring

Description

Return a substring of the source string. strsubk runs both at init and performance time.

Syntax

Sdst **strsubk** Ssrc, kstart, kend

Performance

kstart -- start position in Ssrc, counting from 0. A negative value means the end of the string.

kend -- end position in Ssrc, counting from 0. A negative value means the end of the string. If *kend* is less than *kstart*, the output is reversed.

See also

strsub

Credits

Author: Istvan Varga
2006

strtod

strtod — Converts a string to a float (i-rate).

Description

Convert a string to a floating point value. It is also possible to pass an strset index or a string p-field from the score instead of a string argument. If the string cannot be parsed as a floating point or integer number, an init or perf error occurs and the instrument is deactivated.

Syntax

```
ir strtod Sstr
```

```
ir strtod indx
```

Initialization

Sstr -- String to convert.

indx -- index of string set by strset

Performance

ir -- Value of string as float.

Credits

Author: Istvan Varga
2005

strtodk

strtodk — Converts a string to a float (k-rate).

Description

Convert a string to a floating point value at i- or k-rate. It is also possible to pass an strset index or a string p-field from the score instead of a string argument. If the string cannot be parsed as a floating point or integer number, an init or perf error occurs and the instrument is deactivated.



Note

If a k-rate index variable is used, it should be valid at i-time as well.

Syntax

```
kr strtodk Sstr
```

```
kr strtodk kndx
```

Performance

kr -- Value of string as float.

Sstr -- String to convert.

indx -- index of string set by strset

Credits

Author: Istvan Varga
2005

strtol

strtol — Converts a string to a signed integer (i-rate).

Description

Convert a string to a signed integer value. It is also possible to pass an strset index or a string p-field from the score instead of a string argument. If the string cannot be parsed as a floating point or integer number, an init or perf error occurs and the instrument is deactivated.

Syntax

```
ir strtol Sstr
```

```
ir strtol indx
```

Initialization

Sstr -- String to convert.

indx -- index of string set by strset

strtol can parse numbers in decimal, octal (prefixed by 0), and hexadecimal (with a prefix of 0x) format.

Performance

ir -- Value of string as signed integer.

Credits

Author: Istvan Varga
2005

strtolk

strtolk — Converts a string to a signed integer (k-rate).

Description

Convert a string to a floating point value at i- or k-rate. It is also possible to pass an strset index or a string p-field from the score instead of a string argument. If the string cannot be parsed as a floating point or integer number, an init or perf error occurs and the instrument is deactivated.



Note

If a k-rate index variable is used, it should be valid at i-time as well.

Syntax

```
kr strtolk Sstr
```

```
kr strtolk kndx
```

strtolk can parse numbers in decimal, octal (prefixed by 0), and hexadecimal (with a prefix of 0x) format.

Performance

kr -- Value of string as signed integer.

Sstr -- String to convert.

indx -- index of string set by strset

Credits

Author: Istvan Varga
2005

strupper

strupper — Convert a string to upper case

Description

Convert Ssrc to upper case, and write the result to Sdst. strupper runs at init time only.

Syntax

Sdst **strupper** Ssrc

See also

strupperk

Credits

Author: Istvan Varga
2006

New in version 5.02

strupperk

strupperk — Convert a string to upper case

Description

Convert Ssrc to upper case, and write the result to Sdst. strupperk runs both at init and performance time.

Syntax

Sdst **strupperk** Ssrc

See also

strupper

Credits

Author: Istvan Varga
2006

New in version 5.02

subinstr

subinstr — Creates and runs a numbered instrument instance.

Description

Creates an instance of another instrument and is used as if it were an opcode.

Syntax

```
a1, [...] [, a8] subinstr instrnum [, p4] [, p5] [...]
```

```
a1, [...] [, a8] subinstr "insname" [, p4] [, p5] [...]
```

Initialization

instrnum -- Number of the instrument to be called.

"insname" -- A string (in double-quotes) representing a named instrument.

For more information about specifying input and output interfaces, see *Calling an Instrument within an Instrument*.

Performance

a1, ..., a8 -- The audio output from the called instrument. This is generated using the *signal output* opcodes.

p4, p5, ... -- Additional input values the are mapped to the called instrument p-fields, starting with p4.

The called instrument's p2 and p3 values will be identical to the host instrument's values. While the host instrument can *control its own duration*, any such attempts inside the called instrument will most likely have no effect.

See Also

Calling an Instrument within an Instrument, event, schedule, subinstrinit

Examples

Here is an example of the subinstr opcode. It uses the file *subinstr.csd* [examples/subinstr.csd].

Example 487. Example of the subinstr opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o subinstr.wav -W ;; for file output any platform
</CsOptions>
```

```
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - Creates a basic tone.
instr 1
; Print the value of p4, should be equal to
; Instrument #2's iamp field.
print p4

; Print the value of p5, should be equal to
; Instrument #2's ipitch field.
print p5

; Create a tone.
asig oscils p4, p5, 0

out asig
endin

; Instrument #2 - Demonstrates the subinstr opcode.
instr 2
iamp = 20000
ipitch = 440

; Use Instrument #1 to create a basic sine-wave tone.
; Its p4 parameter will be set using the iamp variable.
; Its p5 parameter will be set using the ipitch variable.
abasic subinstr 1, iamp, ipitch

; Output the basic tone that we have created.
out abasic
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #2 for one second.
i 2 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Here is an example of the subinstr opcode using a named instrument. It uses the file *subinstr_named.csd* [examples/subinstr_named.csd].

Example 488. Example of the subinstr opcode using a named instrument.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o subinstr_named.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument "basic_tone" - Creates a basic tone.
instr basic_tone
; Print the value of p4, should be equal to
; Instrument #2's iamp field.
print p4
```



```
; Print the value of p5, should be equal to
; Instrument #2's ipitch field.
print p5

; Create a tone.
asig oscils p4, p5, 0

out asig
endin

; Instrument #1 - Demonstrates the subinstr opcode.
instr 1
  iamp = 20000
  ipitch = 440

  ; Use the "basic_tone" named instrument to create a
  ; basic sine-wave tone.
  ; Its p4 parameter will be set using the iamp variable.
  ; Its p5 parameter will be set using the ipitch variable.
  abasic subinstr "basic_tone", iamp, ipitch

  ; Output the basic tone that we have created.
  out abasic
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

New in version 4.21

subinstrinit

subinstrinit — Creates and runs a numbered instrument instance at init-time.

Description

Same as *subinstr*, but init-time only and has no output arguments.

Syntax

```
subinstrinit instrnum [, p4] [, p5] [...]
```

```
subinstrinit "insname" [, p4] [, p5] [...]
```

Initialization

instrnum -- Number of the instrument to be called.

"insname" -- A string (in double-quotes) representing a named instrument.

For more information about specifying input and output interfaces, see *Calling an Instrument within an Instrument*.

Performance

p4, p5, ... -- Additional input values the are mapped to the called instrument p-fields, starting with p4.

The called instrument's p2 and p3 values will be identical to the host instrument's values. While the host instrument can *control its own duration*, any such attempts inside the called instrument will most likely have no effect.

See Also

Calling an Instrument within an Instrument, event, schedule, subinstr

Credits

New in version 4.23

sum

sum — Sums any number of a-rate signals.

Description

Sums any number of a-rate signals.

Syntax

```
ares sum asig1 [, asig2] [, asig3] [...]
```

Performance

asig1, asig2, ... -- a-rate signals to be summed (mixed or added).

Credits

Author: Gabriel Maldonado
Italy
April 1999

New in Csound version 3.54

svfilter

svfilter — A resonant second order filter, with simultaneous lowpass, highpass and bandpass outputs.

Description

Implementation of a resonant second order filter, with simultaneous lowpass, highpass and bandpass outputs.

Syntax

```
alow, ahigh, aband svfilter asig, kcf, kq [, iscl]
```

Initialization

iscl -- coded scaling factor, similar to that in *reson*. A non-zero value signifies a peak response factor of 1, i.e. all frequencies other than *kcf* are attenuated in accordance with the (normalized) response curve. A zero value signifies no scaling of the signal, leaving that to some later adjustment (see *balance*). The default value is 0.

Performance

svfilter is a second order state-variable filter, with k-rate controls for cutoff frequency and Q. As Q is increased, a resonant peak forms around the cutoff frequency. *svfilter* has simultaneous lowpass, highpass, and bandpass filter outputs; by mixing the outputs together, a variety of frequency responses can be generated. The state-variable filter, or "multimode" filter was a common feature in early analog synthesizers, due to the wide variety of sounds available from the interaction between cutoff, resonance, and output mix ratios. *svfilter* is well suited to the emulation of "analog" sounds, as well as other applications where resonant filters are called for.

asig -- Input signal to be filtered.

kcf -- Cutoff or resonant frequency of the filter, measured in Hz.

kq -- Q of the filter, which is defined (for bandpass filters) as bandwidth/cutoff. *kq* should be in a range between 1 and 500. As *kq* is increased, the resonance of the filter increases, which corresponds to an increase in the magnitude and "sharpness" of the resonant peak. When using *svfilter* without any scaling of the signal (where *iscl* is either absent or 0), the volume of the resonant peak increases as Q increases. For high values of Q, it is recommended that *iscl* be set to a non-zero value, or that an external scaling function such as *balance* is used.

svfilter is based upon an algorithm in Hal Chamberlin's *Musical Applications of Microprocessors* (Hayden Books, 1985).

Examples

Here is an example of the *svfilter* opcode. It uses the file *svfilter.csd* [examples/svfilter.csd].

Example 489. Example of the *svfilter* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

<CsoundSynthesizer>

```
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o svfilter.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Orchestra file for resonant filter sweep of a sawtooth-like waveform.
; The seperate outputs of the filter are scaled by values from the score,
; and are mixed together.
sr = 44100
kr = 2205
ksmps = 20
nchnls = 1

instr 1

    idur      = p3
    ifreq     = p4
    iamp      = p5
    ilowamp   = p6           ; determines amount of lowpass output in signal
    ihighamp  = p7           ; determines amount of highpass output in signal
    ibandamp  = p8           ; determines amount of bandpass output in signal
    iq        = p9           ; value of q

    iharms    = (sr*.4) / ifreq

    asig      gbuzz 1, ifreq, iharms, 1, .9, 1           ; Sawtooth-like waveform
    kfreq     linseg 1, idur * 0.5, 4000, idur * 0.5, 1 ; Envelope to control filter cutoff

    alow, ahigh, aband svfilter asig, kfreq, iq

    aout1     = alow * ilowamp
    aout2     = ahigh * ihighamp
    aout3     = aband * ibandamp
    asum      = aout1 + aout2 + aout3
    kenv      linseg 0, .1, iamp, idur -.2, iamp, .1, 0 ; Simple amplitude envelope
    out      asum * kenv

endin

</CsInstruments>
<CsScore>

f1 0 8192 9 1 1 .25

i1 0 5 100 1000 1 0 0 5 ; lowpass sweep
i1 5 5 200 1000 1 0 0 30 ; lowpass sweep, octave higher, higher q
i1 10 5 100 1000 0 1 0 5 ; highpass sweep
i1 15 5 200 1000 0 1 0 30 ; highpass sweep, octave higher, higher q
i1 20 5 100 1000 0 0 1 5 ; bandpass sweep
i1 25 5 200 1000 0 0 1 30 ; bandpass sweep, octave higher, higher q
i1 30 5 200 2000 .4 .6 0 ; notch sweep - notch formed by combining highpass and lowpass outputs
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Sean Costello
Seattle, Washington
1999

New in Csound version 3.55

syncgrain

syncgrain — Synchronous granular synthesis.

Description

syncgrain implements synchronous granular synthesis. The source sound for the grains is obtained by reading a function table containing the samples of the source waveform. For sampled-sound sources, *GEN01* is used. *syncgrain* will accept deferred allocation tables.

The grain generator has full control of frequency (grains/sec), overall amplitude, grain pitch (a sampling increment) and grain size (in secs), both as fixed or time-varying (signal) parameters. An extra parameter is the grain pointer speed (or rate), which controls which position the generator will start reading samples in the table for each successive grain. It is measured in fractions of grain size, so a value of 1 (the default) will make each successive grain read from where the previous grain should finish. A value of 0.5 will make the next grain start at the midway position from the previous grain start and finish, etc.. A value of 0 will make the generator read always from a fixed position of the table (wherever the pointer was last at). A negative value will decrement pointer positions. This control gives extra flexibility for creating timescale modifications in the resynthesis.

syncgrain will generate any number of parallel grain streams (which will depend on grain density/frequency), up to the *iolaps* value (default 100). The number of streams (overlapped grains) is determined by $\text{grainsize} * \text{grain_freq}$. More grain overlaps will demand more calculations and the synthesis might not run in realtime (depending on processor power).

syncgrain can simulate FOF-like formant synthesis, provided that a suitable shape is used as grain envelope and a sinewave as the grain wave. For this use, grain sizes of around 0.04 secs can be used. The formant centre frequency is determined by the grain pitch. Since this is a sampling increment, in order to use a frequency in Hz, that value has to be scaled by $\text{tablesize}/\text{sr}$. Grain frequency will determine the fundamental.

syncgrain uses floating-point indexing, so its precision is not affected by large-size tables. This opcode is based on the SndObj library SyncGrain class.

Syntax

```
asig syncgrain kamp, kfreq, kpitch, kgrsize, kprate, ifun1, \  
      ifun2, iolaps
```

Initialization

ifun1 -- source signal function table. Deferred-allocation tables (see *GEN01*) are accepted, but the opcode expects a mono source.

ifun2 -- grain envelope function table.

iolaps -- maximum number of overlaps, $\max(kfreq) * \max(kgrsize)$. Estimating a large value should not affect performance, but exceeding this value will probably have disastrous consequences.

Performance

kamp -- amplitude scaling

kfreq -- frequency of grain generation, or density, in grains/sec.

kpitch -- grain pitch scaling (1=normal pitch, < 1 lower, > 1 higher; negative, backwards)

kgrsize -- grain size in secs.

kprate -- readout pointer rate, in grains. The value of 1 will advance the reading pointer 1 grain ahead in the source table. Larger values will time-compress and smaller values will time-expand the source signal. Negative values will cause the pointer to run backwards and zero will freeze it.

Examples

Example 490. Example

```
iolaps = 2
igrsize = 0.04
ifreq = iolaps/igrsize
ips = 1/iolaps

istr = .5 /* timescale */
ipitch = 1 /* pitchscale */

a1 syncgrain 16000, ifreq, ipitch, igrsize, ips*istr, 1, 2, iolaps
out al
```

Credits

Author: Victor Lazzarini;
January 2005

New plugin in version 5

January 2005.

syncloop

syncloop — Synchronous granular synthesis.

Description

syncloop is a variation on *syncgrain*, which implements synchronous granular synthesis. *syncloop* adds loop start and end points and an optional start position. Loop start and end control grain start positions, so the actual grains can go beyond the loop points (if the loop points are not at the extremes of the table), enabling seamless crossfading. For more information on the granular synthesis process, check the *syncgrain* manual page.

Syntax

```
asig syncloop kamp, kfreq, kpitch, kgrsize, kprate, klstart, \
      klend, ifun1, ifun2, iolaps[,istart, iskip]
```

Initialization

ifun1 -- source signal function table. Deferred-allocation tables (see *GEN01*) are accepted, but the opcode expects a mono source.

ifun2 -- grain envelope function table.

iolaps -- maximum number of overlaps, $\max(kfreq) \cdot \max(kgrsize)$. Estimating a large value should not affect performance, but exceeding this value will probably have disastrous consequences.

istart -- starting point of synthesis in secs (defaults to 0).

iskip -- if 1, the opcode initialisation is skipped, for tied notes, performance continues from the position in the loop where the previous note stopped. The default, 0, does not skip initialisation

Performance

kamp -- amplitude scaling

kfreq -- frequency of grain generation, or density, in grains/sec.

kpitch -- grain pitch scaling (1=normal pitch, < 1 lower, > 1 higher; negative, backwards)

kgrsize -- grain size in secs.

kprate -- readout pointer rate, in grains. The value of 1 will advance the reading pointer 1 grain ahead in the source table. Larger values will time-compress and smaller values will time-expand the source signal. Negative values will cause the pointer to run backwards and zero will freeze it.

klstart -- loop start in secs.

klend -- loop end in secs.

Examples

Example 491. Example


```
iolaps = 2
igrsize = 0.04
ifreq = iolaps/igrsize
ips = 1/iolaps

istr = .5 /* timescale */
ipitch = 1 /* pitchscale */

a1 syncloop 16000, ifreq, ipitch, igrsize, ips*istr, 1, 2, 1, 2, iolaps
out      a1
```

Credits

Author: Victor Lazzarini;
January 2005

New plugin in version 5

January 2005.

syncphasor

syncphasor — Produces a normalized moving phase value with sync input and output.

Description

Produces a moving phase value between zero and one and an extra impulse output ("sync out") whenever its phase value crosses or is reset to zero. The phase can be reset at any time by an impulse on the "sync in" parameter.

Syntax

```
aphase, asyncout syncphasor xcps, asyncin, [, iphs]
```

Initialization

iphs (optional) -- initial phase, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is zero.

Performance

aphase -- the output phase value; always between 0 and 1.

asyncout -- the sync output has a value of 1.0 for one sample whenever the phase value crosses zero or whenever the sync input is non-zero. It is zero at all other times.

asyncin -- the sync input causes the phase to reset to zero whenever *asyncin* is non-zero.

xcps -- frequency of the phasor in cycles-per-second. If *xcps* is negative, the phase value will decrease from 1 to 0 instead of increasing.

An internal phase is successively accumulated in accordance with the *xcps* frequency to produce a moving phase value, normalized to lie in the range $0 \leq \text{phs} < 1$. When used as the index to a *table* unit, this phase (multiplied by the desired function table length) will cause it to behave like an oscillator.

The phase of *syncphasor* though can be synced to another phasor (or other signal) using the *asyncin* parameter. Any time that *asyncin* is a non-zero value, the value of *aphase* will be reset to zero. *syncphasor* also outputs its own "sync" signal that consists of a one-sample impulse whenever its phase crosses zero or is reset. This makes it easy to chain together multiple *syncphasor* opcodes to create an oscillator "hard sync" effect.

Examples

Here is an example of the syncphasor opcode. It uses the file *syncphasor.csd* [examples/syncphasor.csd].

Example 492. Example of the syncphasor opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
```

```
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o abs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

instr 1
    ; Use two syncphasors - one is the "master",
    ; the other the "slave"

    ; master's frequency determines pitch
    imastercps =          cspch(p4)
    imaxamp    =          10000

    ; the slave's frequency affects the timbre
    kslavecps  line       imastercps, p3, imastercps * 3

    ; the master "oscillator"
    ; the master has no sync input
    anosync    init       0.0
    am, async  syncphasor imastercps, anosync

    ; the slave "oscillator"
    aout, as    syncphasor kslavecps, async

    adeclick   linseg      0.0, 0.05, 1.0, p3 - 0.1, 1.0, 0.05, 0.0

    ; Output the slave's phase value which is a rising
    ; sawtooth wave. This produces aliasing, but hey, this
    ; this is just an example ;)

    out        aout * adeclick * imaxamp

endin

</CsInstruments>
<CsScore>

i1 0 1      7.00
i1 + 0.5    7.02
i1 + .      7.05
i1 + .      7.07
i1 + .      7.09
i1 + 2      7.06

e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the syncphasor opcode. It uses the file *syncphasor-CZresonance.csd* [examples/syncphasor-CZresonance.csd].

Example 493. Another example of the syncphasor opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o syncphasor-CZresonance.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
; by Anthony Kozar. February 2008
; http://www.anthonykozar.net/

; Imitation of the Casio CZ-series synthesizer's "Resonance" waveforms
; using a synced phasor to read a sinusoid table. The jumps at the sync
; points are smoothed by multiplying with a windowing function controlled
; by the master phasor.

; Based on information from the Wikipedia article on phase distortion:
; http://en.wikipedia.org/wiki/Phase_distortion_synthesis

; Sawtooth Resonance waveform. Smoothing function is just the inverted
; master phasor.

; The Wikipedia article shows an inverted cosine as the stored waveform,
; which implies that it must be unipolar for the smoothing to work.
; I have substituted a sine wave in the first phrase to keep the output
```

```
; bipolar. The second phrase demonstrates the much "rezzier" sound of the
; bipolar cosine due to discontinuities.

instr 1
  ifreq      =      cpspch(p4)
  initReson  =      p5
  itable     =      p6
  imaxamp    =      10000
  anosync    init   0.0

  kslavecps  line    ifreq * initReson, p3, ifreq
  amaster, async syncphasor ifreq, anosync ; pair of phasors
  aslave, async2 syncphasor kslavecps, async ; slave synced to master
  aosc       tablei   aslave, itable, 1 ; use slave phasor to read a (co)sine table
  aout       =        aosc * (1.0 - amaster) ; inverted master smoothes jumps
  adeclick   linseg   0.0, 0.05, 1.0, p3 - 0.1, 1.0, 0.05, 0.0

                                out      aout * adeclick * imaxamp

endin

; Triangle or Trapezoidal Resonance waveform. Uses a second table to change
; the shape of the smoothing function. (This is my best guess so far as to
; how these worked). The cosine table works fine with the triangular smoothing
; but we once again need to use a sine table with the trapezoidal smoothing.

; (It might be interesting to be able to vary the "width" of the trapezoid.
; This could be done with the pdhalf opcode).

instr 2
  ifreq      =      cpspch(p4)
  initReson  =      p5
  itable     =      p6
  ismoothtbl =      p7
  imaxamp    =      10000
  anosync    init   0.0

  kslavecps  line    ifreq * initReson, p3, ifreq
  amaster, async syncphasor ifreq, anosync ; pair of phasors
  aslave, async2 syncphasor kslavecps, async ; slave synced to master
  aosc       tablei   aslave, itable, 1 ; use slave phasor to read a (co)sine table
  asmooth    tablei   amaster, ismoothtbl, 1 ; use master phasor to read smoothing table
  aout       =        aosc * asmooth
  adeclick   linseg   0.0, 0.05, 1.0, p3 - 0.1, 1.0, 0.05, 0.0

                                out      aout * adeclick * imaxamp

endin

</CsInstruments>
<CsScore>
f1 0 16385 10 1
f3 0 16385 9 1 1 270 ; inverted cosine
f5 0 4097 7 0.0 2048 1.0 2049 0.0 ; unipolar triangle
f6 0 4097 7 1.0 2048 1.0 2049 0.0 ; "trapezoid"

; Sawtooth resonance with a sine table
i1 0 1 7.00 5.0 1
i. + 0.5 7.02 4.0
i. + . 7.05 3.0
i. + . 7.07 2.0
i. + . 7.09 1.0
i. + 2 7.06 12.0
f0 6
s

; Sawtooth resonance with a cosine table
i1 0 1 7.00 5.0 3
i. + 0.5 7.02 4.0
i. + . 7.05 3.0
i. + . 7.07 2.0
i. + . 7.09 1.0
i. + 2 7.06 12.0
f0 6
s

; Triangle resonance with a cosine table
i2 0 1 7.00 5.0 3 5
i. + 0.5 7.02 4.0
i. + . 7.05 3.0
i. + . 7.07 2.0
i. + . 7.09 1.0
i. + 2 7.06 12.0
f0 6
s

; Trapezoidal resonance with a sine table
i2 0 1 7.00 5.0 1 6
i. + 0.5 7.02 4.0
i. + . 7.05 3.0
```

```
i. + .      7.07  2.0
i. + .      7.09  1.0
i. + 2      7.06  12.0

e

</CsScore>
</CsoundSynthesizer>
```

See also

phasor.

And the *Table Access* opcodes like: *table*, *tablei*, *table3* and *tab*.

Credits

Adapted from the *phasor* opcode by Anthony Kozar
January 2008

New in Csound version 5.08

system

system — Call an external program via the system call

Description

system and **system_i** call any external command understood by the operating system, similarly to the C function `system()`. **system_i** runs at i-time only, while **system** runs both at initialization and performance time.

Syntax

```
ires system_i itrig, Scmd, [inowait]
```

```
kres system ktrig, Scmd, [knowait]
```

Initialization

Scmd -- command string

itrig -- if greater than zero the opcode performs the printing; otherwise it is a null operation.

Performance

ktrig -- if greater than zero and different from the value on the previous control cycle the opcode performs the requested printing. Initially this previous value is taken as zero.

inowait, knowait -- if given a non zero the command is run in the background and the command does not wait for the result. (default = 0)

ires, kres -- the return code of the command in wait mode and if the command is run. In other cases returns zero.

More than one system command (a script) can be executed with a single **system** opcode by using double braces strings `{{ }}`.



Note

This opcode is very system dependant, so should be used with extreme care (or not used) if platform neutrality is desired.

Example

Here is an example of the `system_i` opcode. It uses the file `system.csd` [examples/system.csd].

Example 494. Example of the system opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             ; -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
```

```
; -o system.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

instr 1
; Waits for command to execute before continuing
ires system_i 1, {{      ps
    date
    cd ~/Desktop
    pwd
    ls -l
    whois csounds.com
}}
print ires
turnoff
endin

instr 2
; Runs command in a separate thread
ires system_i 1, {{      ps
    date
    cd ~/Desktop
    pwd
    ls -l
    whois csounds.com
}}, 1

print ires
turnoff
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for thirty seconds.
i 1 0 1
i 2 5 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John fitch
2007

New in version 5.06

tb

tb0, tb1, tb2, tb3, tb4, tb5, tb6, tb7, tb8, tb9, tb10, tb11, tb12, tb13, tb14, tb15, tb0_init, tb1_init, tb2_init, tb3_init, tb4_init, tb5_init, tb6_init, tb7_init, tb8_init, tb9_init, tb10_init, tb11_init, tb12_init, tb13_init, tb14_init, tb15_init — Table Read Access inside expressions.

Description

Allow to read tables in function fashion, to be used inside expressions. At present time Csound only supports functions with a single input argument. However, to access table elements, user must provide two numbers, i.e. the number of table and the index of element. So, in order to allow to access a table element with a function, a previous preparation step should be done.

Syntax

```
tb0_init ifn
```

```
tb1_init ifn
```

```
tb2_init ifn
```

```
tb3_init ifn
```

```
tb4_init ifn
```

```
tb5_init ifn
```

```
tb6_init ifn
```

```
tb7_init ifn
```

```
tb8_init ifn
```

```
tb9_init ifn
```

```
tb10_init ifn
```

```
tb11_init ifn
```

```
tb12_init ifn
```

```
tb13_init ifn
```

```
tb14_init ifn
```

```
tb15_init ifn
```

```
iout = tb0(iIndex)
```

```
kout = tb0(kIndex)
```

```
iout = tb1(iIndex)
```

```
kout = tb1(kIndex)
```



```
iout = tb2(iIndex)

kout = tb2(kIndex)

iout = tb3(iIndex)

kout = tb3(kIndex)

iout = tb4(iIndex)

kout = tb4(kIndex)

iout = tb5(iIndex)

kout = tb5(kIndex)

iout = tb6(iIndex)

kout = tb6(kIndex)

iout = tb7(iIndex)

kout = tb7(kIndex)

iout = tb8(iIndex)

kout = tb8(kIndex)

iout = tb9(iIndex)

kout = tb9(kIndex)

iout = tb10(iIndex)

kout = tb10(kIndex)

iout = tb11(iIndex)

kout = tb11(kIndex)

iout = tb12(iIndex)

kout = tb12(kIndex)

iout = tb13(iIndex)

kout = tb13(kIndex)

iout = tb14(iIndex)

kout = tb14(kIndex)

iout = tb15(iIndex)

kout = tb15(kIndex)
```

Performance

There are 16 different opcodes whose name is associated with a number from 0 to 15. User can associate a specific table with each opcode (so the maximum number of tables that can be accessed in function fashion is 16). Prior to access a table, user must associate the table with one of the 16 opcodes by means of an opcode chosen among `tb0_init...tb15_init`. For example,

```
tb0_init 1
```

associates table 1 with `tb0()` function, so that, each element of table 1 can be accessed (in function fashion) with:

```
kvar = tb0(k_some_index_of_table1) * k_some_other_var
```

```
ivar = tb0(i_some_index_of_table1) + i_some_other_var etc...
```

By using these opcodes, user can drastically reduce the number of lines of an orchestra, improving its readability.

Credits

Written by Gabriel Maldonado.

tab

tab — Fast table opcodes.

Description

Fast table opcodes. Faster than *table* and *tablew* because don't allow wrap-around and limit and don't check index validity. Have been implemented in order to provide fast access to arrays. Support non-power of two tables (can be generated by any GEN function by giving a negative length value).

Syntax

```
ir tab_i indx, ifn[, ixmode]
```

```
kr tab kndx, ifn[, ixmode]
```

```
ar tab xndx, ifn[, ixmode]
```

```
tabw_i isig, indx, ifn [,ixmode]
```

```
tabw ksig, kndx, ifn [,ixmode]
```

```
tabw asig, andx, ifn [,ixmode]
```

Initialization

ifn -- table number

ixmode -- defaults to zero. If zero *xndx* and *ixoff* ranges match the length of the table; if non zero *xndx* and *ixoff* have a 0 to 1 range.

isig -- input value to write.

indx -- table index

Performance

asig, *ksig* -- input signal to write.

andx, *kndx* -- table index.

tab and *tabw* opcodes are similar to *table* and *tablew*, but are faster and support tables having non-power-of-two length.

Special care of index value must be taken into account. Index values out of the table allocated space will crash Csound.

Credits

Written by Gabriel Maldonado.

tabrec

tabrec — Recording of control signals.

Description

Records control-rate signals on trigger-temporization basis.

Syntax

```
tabrec    ktrig_start, ktrig_stop, knumtics, kfn, kin1 [,kin2,...,kinN]
```

Performance

ktrig_start -- start recording when non-zero.

ktrig_stop -- stop recording when knumtics trigger impulses are received by this input argument.

knumtics -- stop recording or reset playing pointer to zero when the number of tics defined by this argument is reached.

kfn -- table where k-rate signals are recorded.

kin1,...,kinN -- input signals to record.

The *tabrec* and *tabplay* opcodes allow to record/playback control signals on trigger-temporization basis.

tabrec opcode records a group of k-rate signals by storing them into *kfn* table. Each time *ktrig_start* is triggered, *tabrec* resets the table pointer to zero and begins to record. Recording phase stops after *knumtics* trigger impluses have been received by *ktrig_stop* argument.

These opcodes can be used like a sort of ``middle-term" memory that ``remembers" generated signals. Such memory can be used to supply generative music with a coherent iterative compositional structure.

See Also

tabplay

Credits

Written by Gabriel Maldonado.

table

table — Accesses table values by direct indexing.

Description

Accesses table values by direct indexing.

Syntax

```
ares table andx, ifn [, ixmode] [, ixoff] [, iwrap]
```

```
ires table indx, ifn [, ixmode] [, ixoff] [, iwrap]
```

```
kres table kndx, ifn [, ixmode] [, ixoff] [, iwrap]
```

Initialization

ifn -- function table number.

ixmode (optional) -- index data mode. The default value is 0.

- 0 = raw index
- 1 = normalized (0 to 1)

ixoff (optional) -- amount by which index is to be offset. For a table with origin at center, use `tables-ize/2` (raw) or `.5` (normalized). The default value is 0.

iwrap (optional) -- wraparound index flag. The default value is 0.

- 0 = nowrap (index < 0 treated as index=0; index tablesizes sticks at index=size)
- 1 = wraparound.

Performance

table invokes table lookup on behalf of init, control or audio indices. These indices can be raw entry numbers (0,1,2...size - 1) or scaled values (0 to 1-e). Indices are first modified by the offset value then checked for range before table lookup (see *iwrap*). If index is likely to be full scale, or if interpolation is being used, the table should have an extended guard point. *table* indexed by a periodic phasor (see *phasor*) will simulate an oscillator.

Examples

Here is an example of the table opcode. It uses the file *table.csd* [examples/table.csd].

Example 495. Example of the table opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o table.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Vary our index linearly from 0 to 1.
kndx line 0, p3, 1

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kfreq table kndx, ifn, ixmode

; Generate a sine waveform, use our table values
; to vary its frequency.
a1 oscil 20000, kfreq, 2
out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a line from 200 to 2,000.
f 1 0 1025 -7 200 1024 2000
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

tablei, table3, oscill, oscilli, osciln

Credits

Example written by Kevin Conder.

table3

table3 — Accesses table values by direct indexing with cubic interpolation.

Description

Accesses table values by direct indexing with cubic interpolation.

Syntax

```
ares table3 andx, ifn [, ixmode] [, ixoff] [, iwrap]
```

```
ires table3 indx, ifn [, ixmode] [, ixoff] [, iwrap]
```

```
kres table3 kndx, ifn [, ixmode] [, ixoff] [, iwrap]
```

Initialization

ifn -- function table number.

ixmode (optional) -- index data mode. The default value is 0.

- 0 = raw index
- 1 = normalized (0 to 1)

ixoff (optional) -- amount by which index is to be offset. For a table with origin at center, use `tablesize/2` (raw) or `.5` (normalized). The default value is 0.

iwrap (optional) -- wraparound index flag. The default value is 0.

- 0 = nowrap (index < 0 treated as index=0; index tablesize sticks at index=size)
- 1 = wraparound.

Performance

table3 is identical to *tablei*, except that it uses cubic interpolation. (New in Csound version 3.50.)

See Also

table, *tablei*, *oscil1*, *oscil1i*, *osciln*

tablecopy

tablecopy — Simple, fast table copy opcode.

Description

Simple, fast table copy opcode.

Syntax

```
tablecopy kdft, ksft
```

Performance

kdft -- Destination function table.

ksft -- Number of source function table.

tablecopy -- Simple, fast table copy opcode. Takes the table length from the destination table, and reads from the start of the source table. For speed reasons, does not check the source length - just copies regardless - in “wrap” mode. This may read through the source table several times. A source table with length 1 will cause all values in the destination table to be written to its value.

tablecopy cannot read or write the guardpoint. To read it use *table*, with *ndx* = the table length. Likewise use *table* write to write it.

To write the guardpoint to the value in location 0, use *tablegpw*.

This is primarily to change function tables quickly in a real-time situation.

See Also

tablegpw, *tablemix*, *tableicopy*, *tableigpw*, *tableimix*

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

tablegpw

tablegpw — Writes a table's guard point.

Description

Writes a table's guard point.

Syntax

```
tablegpw kfn
```

Performance

kfn -- Table number to be interrogated

tablegpw -- For writing the table's guard point, with the value which is in location 0. Does nothing if table does not exist.

Likely to be useful after manipulating a table with *tablemix* or *tablecopy*.

See Also

tablecopy, *tablemix*, *tablecopy*, *tableigpw*, *tableimix*

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

tablei

tablei — Accesses table values by direct indexing with linear interpolation.

Description

Accesses table values by direct indexing with linear interpolation.

Syntax

```
ares tablei andx, ifn [, ixmode] [, ixoff] [, iwrap]
```

```
ires tablei indx, ifn [, ixmode] [, ixoff] [, iwrap]
```

```
kres tablei kndx, ifn [, ixmode] [, ixoff] [, iwrap]
```

Initialization

ifn -- function table number. *tablei* requires the extended guard point.

ixmode (optional) -- index data mode. The default value is 0.

- 0 = raw index
- 1 = normalized (0 to 1)

ixoff (optional) -- amount by which index is to be offset. For a table with origin at center, use $\text{tablesize}/2$ (raw) or .5 (normalized). The default value is 0.

iwrap (optional) -- wraparound index flag. The default value is 0.

- 0 = nowrap (index < 0 treated as index=0; index tablesize sticks at index=size)
- 1 = wraparound.

Performance

tablei is a interpolating unit in which the fractional part of index is used to interpolate between adjacent table entries. The smoothness gained by interpolation is at some small cost in execution time (see also *oscili*, etc.), but the interpolating and non-interpolating units are otherwise interchangeable. Note that when *tablei* uses a periodic index whose modulo n is less than the power of 2 table length, the interpolation process requires that there be an $(n+1)$ th table value that is a repeat of the 1st (see *f Statement* in score).

See Also

table, *table3*, *oscil1*, *oscil1i*, *osciln*

tablecopy

tablecopy — Simple, fast table copy opcode.

Description

Simple, fast table copy opcode.

Syntax

```
tablecopy idft, isft
```

Initialization

idft -- Destination function table.

isft -- Number of source function table.

Performance

tablecopy -- Simple, fast table copy opcodes. Takes the table length from the destination table, and reads from the start of the source table. For speed reasons, does not check the source length - just copies regardless - in "wrap" mode. This may read through the source table several times. A source table with length 1 will cause all values in the destination table to be written to its value.

tablecopy cannot read or write the guardpoint. To read it use *table*, with *ndx* = the table length. Likewise use *table write* to write it.

To write the guardpoint to the value in location 0, use *tablegpw*.

This is primarily to change function tables quickly in a real-time situation.

See Also

tablecopy, *tablegpw*, *tablemix*, *tableigpw*, *tableimix*

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

tableigpw

tableigpw — Writes a table's guard point.

Description

Writes a table's guard point.

Syntax

```
tableigpw ifn
```

Initialization

ifn -- Table number to be interrogated

Performance

tableigpw -- For writing the table's guard point, with the value which is in location 0. Does nothing if table does not exist.

Likely to be useful after manipulating a table with *tablemix* or *tablecopy*.

See Also

tablecopy, *tablegpw*, *tablemix*, *tableicopy*, *tableimix*

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

tableikt

tableikt — Provides k-rate control over table numbers.

Description

k-rate control over table numbers.

The standard Csound opcode *tablei*, when producing a k- or a-rate result, can only use an init-time variable to select the table number. *tableikt* accepts k-rate control as well as i-time. In all other respects they are similar to the original opcodes.

Syntax

```
ares tableikt xndx, kfn [, ixmode] [, ixoff] [, iwrap]
```

```
kres tableikt kndx, kfn [, ixmode] [, ixoff] [, iwrap]
```

Initialization

ixmode -- if 0, *xndx* and *ixoff* ranges match the length of the table. if non-zero *xndx* and *ixoff* have a 0 to 1 range. Default is 0

ixoff -- if 0, total index is controlled directly by *xndx*, ie. the indexing starts from the start of the table. If non-zero, start indexing from somewhere else in the table. Value must be positive and less than the table length (*ixmode* = 0) or less than 1 (*ixmode* not equal to 0). Default is 0.

iwrap -- if *iwrap* = 0, *Limit mode*: when total index is below 0, then final index is 0. Total index above table length results in a final index of the table length - high out of range total indexes stick at the upper limit of the table. If *iwrap* not equal to 0, *Wrap mode*: total index is wrapped modulo the table length so that all total indexes map into the table. For instance, in a table of length 8, *xndx* = 5 and *ixoff* = 6 gives a total index of 11, which wraps to a final index of 3. Default is 0.

Performance

kndx -- Index into table, either a positive number range

xndx -- matching the table length (*ixmode* = 0) or a 0 to 1 range (*ixmode* not equal to 0)

kfn -- Table number. Must be ≥ 1 . Floats are rounded down to an integer. If a table number does not point to a valid table, or the table has not yet been loaded (*GEN01*) then an error will result and the instrument will be de-activated.



Caution with k-rate table numbers

At k-rate, if a table number of < 1 is given, or the table number points to a non-existent table, or to one which has a length of 0 (it is to be loaded from a file later) then an error will result and the instrument will be deactivated. *kfn* must be initialized at the appropriate rate using *init*. Attempting to load an i-rate value into *kfn* will result in an error.

See Also

tablekt

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

tablemix

tablemix — Mixes two tables.

Description

Mixes two tables.

Syntax

```
tablemix idft, idoff, ilen, is1ft, is1off, is1g, is2ft, is2off, is2g
```

Initialization

idft -- Destination function table.

idoff -- Offset to start writing from. Can be negative.

ilen -- Number of write operations to perform. Negative means work backwards.

is1ft, is2ft -- Source function tables. These can be the same as the destination table, if care is exercised about direction of copying data.

is1off, is2off -- Offsets to start reading from in source tables.

is1g, is2g -- Gains to apply when reading from the source tables. The results are added and the sum is written to the destination table.

Performance

tablemix -- This opcode mixes from two tables, with separate gains into the destination table. Writing is done for *klen* locations, usually stepping forward through the table - if *klen* is positive. If it is negative, then the writing and reading order is backwards - towards lower indexes in the tables. This bi-directional option makes it easy to shift the contents of a table sideways by reading from it and writing back to it with a different offset.

If *klen* is 0, no writing occurs. Note that the internal integer value of *klen* is derived from the ANSI C floor() function - which returns the next most negative integer. Hence a fractional negative *klen* value of -2.3 would create an internal length of 3, and cause the copying to start from the offset locations and proceed for two locations to the left.

The total index for table reading and writing is calculated from the starting offset for each table, plus the index value, which starts at 0 and then increments (or decrements) by 1 as mixing proceeds.

These total indexes can potentially be very large, since there is no restriction on the offset or the *klen*. However each total index for each table is ANDed with a length mask (such as 0000 0111 for a table of length 8) to form a final index which is actually used for reading or writing. So no reading or writing can occur outside the tables. This is the same as “wrap” mode in table read and write. These opcodes do not read or write the guardpoint. If a table has been rewritten with one of these, then if it has a guardpoint which is supposed to contain the same value as the location 0, then call *tablegpw* afterwards.

The indexes and offsets are all in table steps - they are not normalized to 0 - 1. So for a table of length 256, *klen* should be set to 256 if all the table was to be read or written.

The tables do not need to be the same length - wrapping occurs individually for each table.

See Also

tablecopy, *tablegpw*, *tablemix*, *tableicopy*, *tableigpw*

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

tableiw

tableiw — Change the contents of existing function tables.

Description

This opcode operates on existing function tables, changing their contents. *tableiw* is used when all inputs are init time variables or constants and you only want to run it at the initialization of the instrument. The valid combinations of variable types are shown by the first letter of the variable names.

Syntax

```
tableiw isig, indx, ifn [, ixmode] [, ixoff] [, iwgmode]
```

Initialization

isig -- Input value to write to the table.

indx -- Index into table, either a positive number range matching the table length (*ixmode* = 0) or a 0 to 1 range (*ixmode* not equal to 0)

ifn -- Table number. Must be ≥ 1 . Floats are rounded down to an integer. If a table number does not point to a valid table, or the table has not yet been loaded (*GEN01*) then an error will result and the instrument will be de-activated.

ixmode (optional, default=0) -- index mode.

- 0 = *indx* and *ixoff* ranges match the length of the table.
- not equal to 0 = *indx* and *ixoff* have a 0 to 1 range.

ixoff (optional, default=0) -- index offset.

- 0 = Total index is controlled directly by *indx*, i.e. the indexing starts from the start of the table.
- Not equal to 0 = Start indexing from somewhere else in the table. Value must be positive and less than the table length (*ixmode* = 0) or less than 1 (*ixmode* not equal to 0).

iwgmode (optional, default=0) -- Wrap and guard point mode.

- 0 = Limit mode.
- 1 = Wrap mode.
- 2 = Guardpoint mode.

Performance

Limit mode (0)

Limit the total index (*indx* + *ixoff*) to between 0 and the guard point. For a table of length 5, this

means that locations 0 to 3 and location 4 (the guard point) can be written. A negative total index writes to location 0.

Wrap mode (1)

Wrap total index value into locations 0 to E, where E is either one less than the table length or the factor of 2 number which is one less than the table length. For example, wrap into a 0 to 3 range - so that total index 6 writes to location 2.

Guardpoint mode (2)

The guardpoint is written at the same time as location 0 is written - with the same value.

This facilitates writing to tables which are intended to be read with interpolation for producing smooth cyclic waveforms. In addition, before it is used, the total index is incremented by half the range between one location and the next, before being rounded down to the integer address of a table location.

Normally (*igwmode* = 0 or 1) for a table of length 5 - which has locations 0 to 3 as the main table and location 4 as the guard point, a total index in the range of 0 to 0.999 will write to location 0. ("0.999" means just less than 1.0.) 1.0 to 1.999 will write to location 1 etc. A similar pattern holds for all total indexes 0 to 4.999 (*igwmode* = 0) or to 3.999 (*igwmode* = 1). *igwmode* = 0 enables locations 0 to 4 to be written - with the guardpoint (4) being written with a potentially different value from location 0.

With a table of length 5 and the *igwmode* = 2, then when the total index is in the range 0 to 0.499, it will write to locations 0 and 4. Range 0.5 to 1.499 will write to location 1 etc. 3.5 to 4.0 will also write to locations 0 and 4.

This way, the writing operation most closely approximates the results of interpolated reading. Guard point mode should only be used with tables that have a guardpoint.

Guardpoint mode is accomplished by adding 0.5 to the total index, rounding to the next lowest integer, wrapping it modulo the factor of two which is one less than the table length, writing the table (locations 0 to 3 in our example) and then writing to the guard point if index = 0.

See Also

tablew, *tablewkt*

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

Updated August 2002, thanks go to Abram Hindle for pointing out the correct syntax.

tablekt

tablekt — Provides k-rate control over table numbers.

Description

k-rate control over table numbers.

The standard Csound opcode *table* when producing a k- or a-rate result, can only use an init-time variable to select the table number. *tablekt* accepts k-rate control as well as i-time. In all other respects they are similar to the original opcodes.

Syntax

```
ares tablekt xndx, kfn [, ixmode] [, ixoff] [, iwrap]
```

```
kres tablekt kndx, kfn [, ixmode] [, ixoff] [, iwrap]
```

Initialization

ixmode -- if 0, *xndx* and *ixoff* ranges match the length of the table. if non-zero *xndx* and *ixoff* have a 0 to 1 range. Default is 0

ixoff -- if 0, total index is controlled directly by *xndx*, ie. the indexing starts from the start of the table. If non-zero, start indexing from somewhere else in the table. Value must be positive and less than the table length (*ixmode* = 0) or less than 1 (*ixmode* not equal to 0). Default is 0.

iwrap -- if *iwrap* = 0, *Limit mode*: when total index is below 0, then final index is 0. Total index above table length results in a final index of the table length - high out of range total indexes stick at the upper limit of the table. If *iwrap* not equal to 0, *Wrap mode*: total index is wrapped modulo the table length so that all total indexes map into the table. For instance, in a table of length 8, *xndx* = 5 and *ixoff* = 6 gives a total index of 11, which wraps to a final index of 3. Default is 0.

Performance

kndx -- Index into table, either a positive number range

xndx -- matching the table length (*ixmode* = 0) or a 0 to 1 range (*ixmode* not equal to 0)

kfn -- Table number. Must be ≥ 1 . Floats are rounded down to an integer. If a table number does not point to a valid table, or the table has not yet been loaded (*GEN01*) then an error will result and the instrument will be de-activated.



Caution with k-rate table numbers

At k-rate, if a table number of < 1 is given, or the table number points to a non-existent table, or to one which has a length of 0 (it is to be loaded from a file later) then an error will result and the instrument will be deactivated. *kfn* must be initialized at the appropriate rate using *init*. Attempting to load an i-rate value into *kfn* will result in an error.

See Also

tableikt

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

tablemix

tablemix — Mixes two tables.

Description

Mixes two tables.

Syntax

tablemix *kdft*, *kdoff*, *klen*, *ks1ft*, *ks1off*, *ks1g*, *ks2ft*, *ks2off*, *ks2g*

Performance

kdft -- Destination function table.

kdoff -- Offset to start writing from. Can be negative.

klen -- Number of write operations to perform. Negative means work backwards.

ks1ft, *ks2ft* -- Source function tables. These can be the same as the destination table, if care is exercised about direction of copying data.

ks1off, *ks2off* -- Offsets to start reading from in source tables.

ks1g, *ks2g* -- Gains to apply when reading from the source tables. The results are added and the sum is written to the destination table.

tablemix -- This opcode mixes from two tables, with separate gains into the destination table. Writing is done for *klen* locations, usually stepping forward through the table - if *klen* is positive. If it is negative, then the writing and reading order is backwards - towards lower indexes in the tables. This bi-directional option makes it easy to shift the contents of a table sideways by reading from it and writing back to it with a different offset.

If *klen* is 0, no writing occurs. Note that the internal integer value of *klen* is derived from the ANSI C floor() function - which returns the next most negative integer. Hence a fractional negative *klen* value of -2.3 would create an internal length of 3, and cause the copying to start from the offset locations and proceed for two locations to the left.

The total index for table reading and writing is calculated from the starting offset for each table, plus the index value, which starts at 0 and then increments (or decrements) by 1 as mixing proceeds.

These total indexes can potentially be very large, since there is no restriction on the offset or the *klen*. However each total index for each table is ANDed with a length mask (such as 0000 0111 for a table of length 8) to form a final index which is actually used for reading or writing. So no reading or writing can occur outside the tables. This is the same as “wrap” mode in table read and write. These opcodes do not read or write the guardpoint. If a table has been rewritten with one of these, then if it has a guardpoint which is supposed to contain the same value as the location 0, then call *tablegpw* afterwards.

The indexes and offsets are all in table steps - they are not normalized to 0 - 1. So for a table of length 256, *klen* should be set to 256 if all the table was to be read or written.

The tables do not need to be the same length - wrapping occurs individually for each table.

See Also

tablecopy, *tablegpw*, *tableicopy*, *tableigpw*, *tableimix*

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

tableng

tableng — Interrogates a function table for length.

Description

Interrogates a function table for length.

Syntax

```
ires tableng ifn
```

```
kres tableng kfn
```

Initialization

ifn -- Table number to be interrogated

Performance

kfn -- Table number to be interrogated

tableng returns the length of the specified table. This will be a power of two number in most circumstances. It will not show whether a table has a guardpoint or not. It seems this information is not available in the table's data structure. If the specified table is not found, then 0 will be returned.

Likely to be useful for setting up code for table manipulation operations, such as *tablemix* and *table-copy*.

Examples

Here is an example of the *tableng* opcode. It uses the file *tableng.csd* [examples/*tableng.csd*].

Example 496. Example of the *tableng* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsSoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac            -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o tableng.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Let's look at Table #1.
ifn = 1
ilen tableng ifn
```

```
    print ilen
  endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

The table is 16,384 samples long. So its output should include a line like this:

```
instr 1:  ilen = 16384.000
```

Credits

Author: Robin Whittle
Australia
May 1997

Example written by Kevin Conder.

tablera

tablera — Reads tables in sequential locations.

Description

These opcode reads tables in sequential locations to an a-rate variable. Some thought is required before using it. It has at least two major, and quite different, applications which are discussed below.

Syntax

```
ares tablera kfn, kstart, koff
```

Performance

ares -- a-rate destination for reading *ksmps* values from a table.

kfn -- i- or k-rate number of the table to read or write.

kstart -- Where in table to read or write.

koff -- i- or k-rate offset into table. Range unlimited - see explanation at end of this section.

In one application, *tablera* is intended to be used in pair with *tablewa*, or with several *tablera* opcodes before a *tablewa* -- all sharing the same *kstart* variable.

These read from and write to sequential locations in a table at audio rates, with *ksmps* floats being written and read each cycle.

tablera starts reading from location *kstart*. *tablewa* starts writing to location *kstart*, and then writes to *kstart* with the number of the location one more than the one it last wrote. (Note that for *tablewa*, *kstart* is both an input and output variable.) If the writing index reaches the end of the table, then no further writing occurs and zero is written to *kstart*.

For instance, if the table's length was 16 (locations 0 to 15), and *ksmps* was 5. Then the following steps would occur with repetitive runs of the *tablewa* opcode, assuming that *kstart* started at 0.

Run Number	Initial kstart	Final kstart	Locations Written
1	0	5	0 1 2 3 4
2	5	10	5 6 7 8 9
3	10	15	10 11 12 13 14
4	15	0	15

This is to facilitate processing table data using standard a-rate orchestra code between the *tablera* and *tablewa* opcodes. They allow all Csound k-rate operators to be used (with caution) on a-rate variables - something that would only be possible otherwise by *ksmps* = 1, *downsamp* and *upsamp*.



Several cautions

- The k-rate code in the processing loop is really running at a-rate, so time dependent functions like *port* and *oscil* work faster than normal - their code is expecting to be running at k-rate.

- This system will produce undesirable results unless the *ksmps* fits within the table length. For instance a table of length 16 will accommodate 1 to 16 samples, so this example will work with *ksmps* = 1 to 16.

Both these opcodes generate an error and deactivate the instrument if a table with length < *ksmps* is selected. Likewise an error occurs if *kstart* is below 0 or greater than the highest entry in the table - if *kstart* = table length.

- *kstart* is intended to contain integer values between 0 and (table length - 1). Fractional values above this should not affect operation but do not achieve anything useful.
- These opcodes are not interpolating, and the *kstart* and *koff* parameters always have a range of 0 to (table length - 1) - not 0 to 1 as is available in other table read/write opcodes. *koff* can be outside this range but it is wrapped around by the final AND operation.
- These opcodes are permanently in wrap mode. When *koff* is 0, no wrapping needs to occur, since the *kstart++* index will always be within the table's normal range. *koff* not equal to 0 can lead to wrapping.
- The offset does not affect the number of read/write cycles performed, or the value written to *kstart* by *tablewa*.
- These opcodes cannot read or write the guardpoint. Use *tablegpw* to write the guardpoint after manipulations have been done with *tablewa*.

Examples

```
kstart    =      0

lab1:
  atemp    tablera ktabsource, kstart, 0 ; Read 5 values from table into an
      ; a-rate variable.

  atemp    =      log(atemp) ; Process the values using a-rate
      ; code.

  kstart    tablewa ktabdest, atemp, 0 ; Write it back to the table

if ktemp 0 goto lab1 ; Loop until all table locations
      ; have been processed.
```

The above example shows a processing loop, which runs every k-cycle, reading each location in the table *ktabsource*, and writing the log of those values into the same locations of table *ktabdest*.

This enables whole tables, parts of tables (with offsets and different control loops) and data from several tables at once to be manipulated with a-rate code and written back to another (or to the same) table. This is a bit of a fudge, but it is faster than doing it with k-rate table read and write code.

Another application is:

```
kzero = 0
kloop = 0
```

```
kzero tablewa 23, asignal, 0 ; ksmps a-rate samples written
    ; into locations 0 to (ksmps -1) of table 23.

lab1: ktemp table kloop, 23 ; Start a loop which runs ksmps times,
    ; in which each cycle processes one of
    [ Some code to manipulate ] ; table 23's values with k-rate orchestra
    [ the value of ktemp. ] ; code.

    tablew ktemp, kloop, 23 ; Write the processed value to the table.

kloop = kloop + 1 ; Increment the kloop, which is both the
    ; pointer into the table and the loop
if kloop < ksmps goto lab1 ; counter. Keep looping until all values
    ; in the table have been processed.

asignal tablea 23, 0, 0 ; Copy the table contents back
    ; to an a-rate variable.
```

koff -- This is an offset which is added to the sum of *kstart* and the internal index variable which steps through the table. The result is then ANDed with the lengthmask (000 0111 for a table of length 8 - or 9 with guardpoint) and that final index is used to read or write to the table. *koff* can be any value. It is converted into a long using the ANSI floor() function so that -4.3 becomes -5. This is what we would want when using offsets which range above and below zero.

Ideally this would be an optional variable, defaulting to 0, however with the existing Csound orchestra read code, such default parameters must be init time only. We want k-rate here, so we cannot have a default.

See Also

tablewa

tableseg

tableseg — Creates a new function table by making linear segments between values in stored function tables.

Description

tableseg is like *linseg* but interpolate between values in a stored function tables. The result is a new function table passed internally to any following *vpvoc* which occurs before a subsequent *tableseg* (much like *lpread/lpreson* pairs work). The uses of these are described below under *vpvoc*.

Syntax

```
tableseg ifn1, idur1, ifn2 [, idur2] [, ifn3] [...]
```

Initialization

ifn1, *ifn2*, *ifn3*, etc. -- function table numbers. *ifn1*, *ifn2*, and so on, must be the same size.

idur1, *idur2*, etc. -- durations during which interpolation from one table to the next will take place.

See Also

pvbufread, *pvcross*, *pvinterp*, *pvread*, *tablexseg*

Credits

Author: Richard Karpen
Seattle, Wash
1997

New in version 3.44

tablew

tablew — Change the contents of existing function tables.

Description

This opcode operates on existing function tables, changing their contents. *tablew* is for writing at k- or at a-rates, with the table number being specified at init time. The valid combinations of variable types are shown by the first letter of the variable names.

Syntax

```
tablew asig, andx, ifn [, ixmode] [, ixoff] [, iwgmodes]
```

```
tablew isig, indx, ifn [, ixmode] [, ixoff] [, iwgmodes]
```

```
tablew ksig, kndx, ifn [, ixmode] [, ixoff] [, iwgmodes]
```

Initialization

asig, isig, ksig -- The value to be written into the table.

andx, indx, kndx -- Index into table, either a positive number range matching the table length (*ixmode* = 0) or a 0 to 1 range (*ixmode* != 0)

ifn -- Table number. Must be >= 1. Floats are rounded down to an integer. If a table number does not point to a valid table, or the table has not yet been loaded (*GEN01*) then an error will result and the instrument will be de-activated.

ixmode (optional, default=0) -- index mode.

- 0 = *xndx* and *ixoff* ranges match the length of the table.
- !=0 = *xndx* and *ixoff* have a 0 to 1 range.

ixoff (optional, default=0) -- index offset.

- 0 = Total index is controlled directly by *xndx*, i.e. the indexing starts from the start of the table.
- !=0 = Start indexing from somewhere else in the table. Value must be positive and less than the table length (*ixmode* = 0) or less than 1 (*ixmode* != 0).

iwgmodes (optional, default=0) -- Wrap and guardpoint mode.

- 0 = Limit mode.
- 1 = Wrap mode.
- 2 = Guardpoint mode.

Performance

Limit mode (0)

Limit the total index ($ndx + ixoff$) to between 0 and the guard point. For a table of length 5, this means that locations 0 to 3 and location 4 (the guard point) can be written. A negative total index writes to location 0.

Wrap mode (1)

Wrap total index value into locations 0 to E, where E is either one less than the table length or the factor of 2 number which is one less than the table length. For example, wrap into a 0 to 3 range - so that total index 6 writes to location 2.

Guardpoint mode (2)

The guardpoint is written at the same time as location 0 is written - with the same value.

This facilitates writing to tables which are intended to be read with interpolation for producing smooth cyclic waveforms. In addition, before it is used, the total index is incremented by half the range between one location and the next, before being rounded down to the integer address of a table location.

Normally ($igwmode = 0$ or 1) for a table of length 5 - which has locations 0 to 3 as the main table and location 4 as the guard point, a total index in the range of 0 to 0.999 will write to location 0. ("0.999" means just less than 1.0.) 1.0 to 1.999 will write to location 1 etc. A similar pattern holds for all total indexes 0 to 4.999 ($igwmode = 0$) or to 3.999 ($igwmode = 1$). $igwmode = 0$ enables locations 0 to 4 to be written - with the guardpoint (4) being written with a potentially different value from location 0.

With a table of length 5 and the $iwgmode = 2$, then when the total index is in the range 0 to 0.499, it will write to locations 0 and 4. Range 0.5 to 1.499 will write to location 1 etc. 3.5 to 4.0 will also write to locations 0 and 4.

This way, the writing operation most closely approximates the results of interpolated reading. Guard point mode should only be used with tables that have a guardpoint.

Guardpoint mode is accomplished by adding 0.5 to the total index, rounding to the next lowest integer, wrapping it modulo the factor of two which is one less than the table length, writing the table (locations 0 to 3 in our example) and then writing to the guard point if index = 0.

tablew has no output value. The last three parameters are optional and have default values of 0.

Caution with k-rate table numbers

At k-rate or a-rate, if a table number of < 1 is given, or the table number points to a non-existent table, or to one which has a length of 0 (it is to be loaded from a file later) then an error will result and the instrument will be deactivated. *kfn* and *afn* must be initialized at the appropriate rate using *init*. Attempting to load an i-rate value into *kfn* or *afn* will result in an error.

See Also

tableiw, *tablewkt*

Credits

Author: Robin Whittle
Australia
May 1997

tablewa

tablewa — Writes tables in sequential locations.

Description

This opcode writes to a table in sequential locations to and from an a-rate variable. Some thought is required before using it. It has at least two major, and quite different, applications which are discussed below.

Syntax

```
kstart tablewa kfn, asig, koff
```

Performance

kstart -- Where in table to read or write.

kfn -- i- or k-rate number of the table to read or write.

asig -- a-rate signal to read from when writing to the table.

koff -- i- or k-rate offset into table. Range unlimited - see explanation at end of this section.

In one application, it is intended to be used with one or with several *tablera* opcodes before a *tablewa* -- all sharing the same *kstart* variable.

These read from and write to sequential locations in a table at audio rates, with *ksmps* floats being written and read each cycle.

tablera starts reading from location *kstart*. *tablewa* starts writing to location *kstart*, and then writes to *kstart* with the number of the location one more than the one it last wrote. (Note that for *tablewa*, *kstart* is both an input and output variable.) If the writing index reaches the end of the table, then no further writing occurs and zero is written to *kstart*.

For instance, if the table's length was 16 (locations 0 to 15), and *ksmps* was 5. Then the following steps would occur with repetitive runs of the *tablewa* opcode, assuming that *kstart* started at 0.

Run Number	Initial kstart	Final kstart	Locations Written
1	0	5	0 1 2 3 4
2	5	10	5 6 7 8 9
3	10	15	10 11 12 13 14
4	15	0	15

This is to facilitate processing table data using standard a-rate orchestra code between the *tablera* and *tablewa* opcodes. They allow all Csound k-rate operators to be used (with caution) on a-rate variables - something that would only be possible otherwise by *ksmps* = 1, *downsamp* and *upsamp*.



Several cautions

- The k-rate code in the processing loop is really running at a-rate, so time dependent functions like *port* and *oscil* work faster than normal - their code is expecting to be running at k-rate.

- This system will produce undesirable results unless the *ksmps* fits within the table length. For instance a table of length 16 will accommodate 1 to 16 samples, so this example will work with *ksmps* = 1 to 16.

Both these opcodes generate an error and deactivate the instrument if a table with length < *ksmps* is selected. Likewise an error occurs if *kstart* is below 0 or greater than the highest entry in the table - if *kstart* = table length.

- *kstart* is intended to contain integer values between 0 and (table length - 1). Fractional values above this should not affect operation but do not achieve anything useful.
- These opcodes are not interpolating, and the *kstart* and *koff* parameters always have a range of 0 to (table length - 1) - not 0 to 1 as is available in other table read/write opcodes. *koff* can be outside this range but it is wrapped around by the final AND operation.
- These opcodes are permanently in wrap mode. When *koff* is 0, no wrapping needs to occur, since the *kstart++* index will always be within the table's normal range. *koff* not equal to 0 can lead to wrapping.
- The offset does not affect the number of read/write cycles performed, or the value written to *kstart* by *tablewa*.
- These opcodes cannot read or write the guardpoint. Use *tablegpw* to write the guardpoint after manipulations have been done with *tablewa*.

Examples

```
kstart    =      0

lab1:
  atemp    tablea ktabsource, kstart, 0 ; Read 5 values from table into an
      ; a-rate variable.

  atemp    =      log(atemp) ; Process the values using a-rate
      ; code.

  kstart    tablewa ktabdest, atemp, 0 ; Write it back to the table

if ktemp 0 goto lab1 ; Loop until all table locations
      ; have been processed.
```

The above example shows a processing loop, which runs every k-cycle, reading each location in the table *ktabsource*, and writing the log of those values into the same locations of table *ktabdest*.

This enables whole tables, parts of tables (with offsets and different control loops) and data from several tables at once to be manipulated with a-rate code and written back to another (or to the same) table. This is a bit of a fudge, but it is faster than doing it with k-rate table read and write code.

Another application is:

```
kzero = 0
kloop = 0
```



```
kzero tablewa 23, asignal, 0 ; ksmps a-rate samples written
    ; into locations 0 to (ksmps -1) of table 23.

lab1: ktemp table kloop, 23 ; Start a loop which runs ksmps times,
    ; in which each cycle processes one of
    [ Some code to manipulate ] ; table 23's values with k-rate orchestra
    [ the value of ktemp. ] ; code.

    tablew ktemp, kloop, 23 ; Write the processed value to the table.

kloop = kloop + 1 ; Increment the kloop, which is both the
    ; pointer into the table and the loop
if kloop < ksmps goto lab1 ; counter. Keep looping until all values
    ; in the table have been processed.

asignal tablea 23, 0, 0 ; Copy the table contents back
    ; to an a-rate variable.
```

koff -- This is an offset which is added to the sum of *kstart* and the internal index variable which steps through the table. The result is then ANDed with the lengthmask (000 0111 for a table of length 8 - or 9 with guardpoint) and that final index is used to read or write to the table. *koff* can be any value. It is converted into a long using the ANSI floor() function so that -4.3 becomes -5. This is what we would want when using offsets which range above and below zero.

Ideally this would be an optional variable, defaulting to 0, however with the existing Csound orchestra read code, such default parameters must be init time only. We want k-rate here, so we cannot have a default.

Credits

Author: Robin Whittle
Australia

tablewkt

tablewkt — Change the contents of existing function tables.

Description

This opcode operates on existing function tables, changing their contents. *tablewkt* uses a k-rate variable for selecting the table number. The valid combinations of variable types are shown by the first letter of the variable names.

Syntax

```
tablewkt asig, andx, kfn [, ixmode] [, ixoff] [, iwgmodes]
```

```
tablewkt ksig, kndx, kfn [, ixmode] [, ixoff] [, iwgmodes]
```

Initialization

asig, ksig -- The value to be written into the table.

andx, kndx -- Index into table, either a positive number range matching the table length (*ixmode* = 0) or a 0 to 1 range (*ixmode* != 0)

kfn -- Table number. Must be >= 1. Floats are rounded down to an integer. If a table number does not point to a valid table, or the table has not yet been loaded (*GEN01*) then an error will result and the instrument will be de-activated.

ixmode -- index mode. Default is zero.

- 0 = *xndx* and *ixoff* ranges match the length of the table.
- Not equal to 0 = *xndx* and *ixoff* have a 0 to 1 range.

ixoff -- index offset. Default is 0.

- 0 = Total index is controlled directly by *xndx*, i.e. the indexing starts from the start of the table.
- Not equal to 0 = Start indexing from somewhere else in the table. Value must be positive and less than the table length (*ixmode* = 0) or less than 1 (*ixmode* != 0).

iwgmodes -- table writing mode. Default is 0.

- 0 = Limit mode.
- 1 = Wrap mode.
- 2 = Guardpoint mode.

Performance

Limit mode (0)

Limit the total index ($ndx + ixoff$) to between 0 and the guard point. For a table of length 5, this means that locations 0 to 3 and location 4 (the guard point) can be written. A negative total index writes to location 0.

Wrap mode (1)

Wrap total index value into locations 0 to E, where E is one less than either the table length or the factor of 2 number which is one less than the table length. For example, wrap into a 0 to 3 range - so that total index 6 writes to location 2.

Guardpoint mode (2)

The guardpoint is written at the same time as location 0 is written - with the same value.

This facilitates writing to tables which are intended to be read with interpolation for producing smooth cyclic waveforms. In addition, before it is used, the total index is incremented by half the range between one location and the next, before being rounded down to the integer address of a table location.

Normally ($igwmode = 0$ or 1) for a table of length 5 - which has locations 0 to 3 as the main table and location 4 as the guard point, a total index in the range of 0 to 0.999 will write to location 0. ("0.999" means just less than 1.0.) 1.0 to 1.999 will write to location 1 etc. A similar pattern holds for all total indexes 0 to 4.999 ($igwmode = 0$) or to 3.999 ($igwmode = 1$). $igwmode = 0$ enables locations 0 to 4 to be written - with the guardpoint (4) being written with a potentially different value from location 0.

With a table of length 5 and the $igwmode = 2$, then when the total index is in the range 0 to 0.499, it will write to locations 0 and 4. Range 0.5 to 1.499 will write to location 1 etc. 3.5 to 4.0 will also write to locations 0 and 4.

This way, the writing operation most closely approximates the results of interpolated reading. Guard point mode should only be used with tables that have a guardpoint.

Guardpoint mode is accomplished by adding 0.5 to the total index, rounding to the next lowest integer, wrapping it modulo the factor of two which is one less than the table length, writing the table (locations 0 to 3 in our example) and then writing to the guard point if index = 0.

Caution with k-rate table numbers

At k-rate or a-rate, if a table number of < 1 is given, or the table number points to a non-existent table, or to one which has a length of 0 (it is to be loaded from a file later) then an error will result and the instrument will be deactivated. kfn and afn must be initialized at the appropriate rate using *init*. Attempting to load an i-rate value into kfn or afn will result in an error.

See Also

tableiw, *tablew*

Credits

Author: Robin Whittle
Australia
May 1997

tablexkt

tablexkt — Reads function tables with linear, cubic, or sinc interpolation.

Description

Reads function tables with linear, cubic, or sinc interpolation.

Syntax

```
ares tablexkt xndx, kfn, kwrap, iwsiz [ , ixmod] [ , ixoff] [ , iwrap]
```

Initialization

iwsiz -- This parameter controls the type of interpolation to be used:

- 2: Use linear interpolation. This is the lowest quality, but also the fastest mode.
- 4: Cubic interpolation. Slightly better quality than *iwsiz* = 2, at the expense of being somewhat slower.
- 8 and above (up to 1024): sinc interpolation with window size set to *iwsiz* (should be an integer multiply of 4). Better quality than linear or cubic interpolation, but very slow. When transposing up, a *kwrap* value above 1 can be used for anti-aliasing (this is even slower).

ixmod (optional) -- index data mode. The default value is 0.

- 0: raw index
- any non-zero value: normalized (0 to 1)



Notes

if *tablexkt* is used to play back samples with looping (e.g. table index is generated by *lphasor*), there must be at least *iwsiz* / 2 extra samples after the loop end point for interpolation, otherwise audible clicking may occur (also, at least *iwsiz* / 2 samples should be before the loop start point).

ixoff (optional) -- amount by which index is to be offset. For a table with origin at center, use *tablesiz* / 2 (raw) or 0.5 (normalized). The default value is 0.

iwrap (optional) -- wraparound index flag. The default value is 0.

- 0: Nowrap (index < 0 treated as index = 0; index >= *tablesiz* (or 1.0 in normalized mode) sticks at the guard point).
- any non-zero value: Index is wrapped to the allowed range (not including the guard point in this case).



Note

iwrap also applies to extra samples for interpolation.

Performance

ares -- audio output

xndx -- table index

kfn -- function table number

kwarp -- if greater than 1, use $\sin(x / \text{kwarp}) / x$ function for sinc interpolation, instead of the default $\sin(x) / x$. This is useful to avoid aliasing when transposing up (*kwarp* should be set to the transpose factor in this case, e.g. 2.0 for one octave), however it makes rendering up to twice as slow. Also, *iwsiz*e should be at least $\text{kwarp} * 8$. This feature is experimental, and may be optimized both in terms of speed and quality in new versions.



Note

kwarp has no effect if it is less than, or equal to 1, or linear or cubic interpolation is used.

Examples

Here is an example of the *tablexkt* opcode. It uses the file *tablexkt.csd* [examples/tablexkt.csd].

Example 497. Example of the *tablexkt* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o tablexkt.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
;Example by Jonathan Murphy

sr      = 44100
ksmps   = 10
nchnls  = 1

instr 1

ifn      = 1      ; query f1 as to number of samples
ilen     = nsamp(ifn)

itrns    = 4      ; transpose up 4 octaves
ilps     = 16     ; allow iwsiz/2 samples at start
ilpe     = ilen - 16 ; and at end
imode    = 3      ; loop forwards and backwards
istrt    = 16     ; start 16 samples into loop

alphs    lphasor  itrns, ilps, ilpe, imode, istrt
; use lphasor as index
andx     = alphs

kfn      = 1      ; read f1
kwarp    = 4      ; anti-aliasing, should be same value as itrns above
iwsiz    = 32     ; iwsiz must be at least 8 * kwarp

atab     tablexkt andx, kfn, kwarp, iwsiz

atab     = atab * 10000

        out      atab

endin

</CsInstruments>
```

```
<CsScore>  
f 1 0 262144 1 "beats.wav" 0 4 1  
i1 0 60  
e  
</CsScore>  
</CsoundSynthesizer>
```

Credits

Author: Istvan Varga

January 2002

Example by: Jonathan Murphy 2006

New in version 4.18

tablexseg

tablexseg — Creates a new function table by making exponential segments between values in stored function tables.

Description

tablexseg is like *expseg* but interpolate between values in a stored function tables. The result is a new function table passed internally to any following *vpvoc* which occurs before a subsequent *tablexseg* (much like *lpread/lpreson* pairs work). The uses of these are described below under *vpvoc*.

Syntax

```
tablexseg ifn1, idur1, ifn2 [, idur2] [, ifn3] [...]
```

Initialization

ifn1, *ifn2*, *ifn3*, etc. -- function table numbers. *ifn1*, *ifn2*, and so on, must be the same size.

idur1, *idur2*, etc. -- durations during which interpolation from one table to the next will take place.

See Also

pvbufread, *pvcross*, *pvinterp*, *pvread*, *tableseg*

Credits

Author: Richard Karpen
Seattle, WA USA
1997

tabmorph

tabmorph — Allow morphing between a set of tables.

Description

tabmorph allows morphing between a set of tables of the same size, by means of a weighted average between two currently selected tables.

Syntax

```
kout tabmorph kindex, kweightpoint, ktabnum1, ktabnum2, \  
    ifn1, ifn2 [, ifn3, ifn4, ... ifnN]
```

Initialization

ifn1, ifn2 [, ifn3, ifn4,...ifnN] - function table numbers. This is a set of chosen tables the user want to use in the morphing. All tables must have the same length. Be aware that only two of these tables can be chosen for the morphing at one time. Since it is possible to use non-integer numbers for the *ktabnum1* and *ktabnum2* arguments, the morphing is the result from the interpolation between adjacent consecutive tables of the set.

Performance

kout - The output value for index *kindex*, resulting from morphing two tables (see below).

kindex - main index index of the morphed resultant table. The range is 0 to table_length (not included).

kweightpoint - the weight of the influence of a pair of selected tables in the morphing. The range of this argument is 0 to 1. A zero makes it output the first table unaltered, a 1 makes it output the second table of the pair unaltered. All intermediate values between 0 and 1 determine the gradual morphing between the two tables of the pair.

ktabnum1 - the first table chosen for the morphing. This number doesn't express the table number directly, but the position of the table in the set sequence (starting from 0 to N-1). If this number is an integer, the corresponding table will be chosen unaltered. If it contains fractional values, then an interpolation with the next adjacent table will result.

ktabnum2 - the second table chosen for the morphing. This number doesn't express the table number directly, but the position of the table in the set sequence (starting from 0 to N-1). If this number is an integer, corresponding table will be chosen unaltered. If it contains fractional values, then an interpolation with the next adjacent table will result.

The *tabmorph* family of opcodes is similar to the *table* family, but allows morphing between two tables chosen into a set of tables. Firstly the user has to provide a set of tables of equal length (*ifn2 [, ifn3, ifn4,...ifnN]*). Then he can choose a pair of tables in the set in order to perform the morphing: *ktabnum1* and *ktabnum2* are filled with numbers (zero represents the first table in the set, 1 the second, 2 the third and so on). Then determine the morphing between the two chosen tables with the *kweightpoint* parameter. After that the resulting table can be indexed with the *kindex* parameter like a normal table opcode. If the value of this parameter surpasses the length of tables (which must be the same for all tables), then it is wrapped around.

tabmorph acts similarly to the *table* opcode, that is, without using interpolation. This means that it truncates the fractional part of the *kindex* argument. Anyway, fractional parts of *ktabnum1* and *ktabnum2* are significant, resulting in linear interpolation between the same element of two adjacent subsequent tables.

See Also

table, tabmorphi, tabmorpha, tabmorphak, ftmorf,

Credits

Author: Gabriel Maldonado

New in version 5.06

tabmorpha

tabmorpha — Allow morphing between a set of tables at audio rate with interpolation.

Description

tabmorpha allows morphing between a set of tables of the same size, by means of a weighted average between two currently selected tables.

Syntax

```
aout tabmorpha aindex, aweightpoint, atabnum1, atabnum2, \  
ifn1, ifn2 [, ifn3, ifn4, ... ifnN]
```

Initialization

ifn1, ifn2, ifn3, ifn4, ... ifnN - function table numbers. This is a set of chosen tables the user want to use in the morphing. All tables must have the same length. Be aware that only two of these tables can be chosen for the morphing at one time. Since it is possible to use non-integer numbers for the *atabnum1* and *atabnum2* arguments, the morphing is the result from the interpolation between adjacent consecutive tables of the set.

Performance

aout - The output value for index *aindex*, resulting from morphing two tables (see below).

aindex - main index index of the morphed resultant table. The range is 0 to table_length (not included).

aweightpoint - the weight of the influence of a pair of selected tables in the morphing. The range of this argument is 0 to 1. A zero makes it output the first table unaltered, a 1 makes it output the second table of the pair unaltered. All intermediate values between 0 and 1 determine the gradual morphing between the two tables of the pair.

atabnum1 - the first table chosen for the morphing. This number doesn't express the table number directly, but the position of the table in the set sequence (starting from 0 to N-1). If this number is an integer, the corresponding table will be chosen unaltered. If it contains fractional values, then an interpolation with the next adjacent table will result.

atabnum2 - the second table chosen for the morphing. This number doesn't express the table number directly, but the position of the table in the set sequence (starting from 0 to N-1). If this number is an integer, corresponding table will be chosen unaltered. If it contains fractional values, then an interpolation with the next adjacent table will result.

The *tabmorpha* family of opcodes is similar to the *table* family, but allows morphing between two tables chosen into a set of tables. Firstly the user has to provide a set of tables of equal length (*ifn2* [, *ifn3*, *ifn4*,...*ifnN*]). Then he can choose a pair of tables in the set in order to perform the morphing: *atabnum1* and *aatabnum2* are filled with numbers (zero represents the first table in the set, 1 the second, 2 the third and so on). Then determine the morphing between the two chosen tables with the *aweightpoint* parameter. After that the resulting table can be indexed with the *aindex* parameter like a normal table opcode. If the value of this parameter surpasses the length of tables (which must be the same for all tables), then it is wrapped around.

tabmorpha is the audio-rate version of *tabmorphi* (it uses interpolation). All input arguments work at a-rate.

See Also

table, tabmorph, tabmorphi, tabmorphak, fmorf,

Credits

Author: Gabriel Maldonado

New in version 5.06

tabmorphak

tabmorphak — Allow morphing between a set of tables at audio rate with interpolation.

Description

tabmorphak allows morphing between a set of tables of the same size, by means of a weighted average between two currently selected tables.

Syntax

```
aout tabmorphak aindex, kweightpoint, ktabnum1, ktabnum2, \  
    ifn1, ifn2 [, ifn3, ifn4, ... ifnN]
```

Initialization

ifn1, ifn2, ifn3, ifn4, ... ifnN - function table numbers. This is a set of chosen tables the user want to use in the morphing. All tables must have the same length. Be aware that only two of these tables can be chosen for the morphing at one time. Since it is possible to use non-integer numbers for the *atabnum1* and *atabnum2* arguments, the morphing is the result from the interpolation between adjacent consecutive tables of the set.

Performance

aout - The output value for index *aindex*, resulting from morphing two tables (see below).

aindex - main index index of the morphed resultant table. The range is 0 to table_length (not included).

kweightpoint - the weight of the influence of a pair of selected tables in the morphing. The range of this argument is 0 to 1. A zero makes it output the first table unaltered, a 1 makes it output the second table of the pair unaltered. All intermediate values between 0 and 1 determine the gradual morphing between the two tables of the pair.

ktabnum1 - the first table chosen for the morphing. This number doesn't express the table number directly, but the position of the table in the set sequence (starting from 0 to N-1). If this number is an integer, the corresponding table will be chosen unaltered. If it contains fractional values, then an interpolation with the next adjacent table will result.

ktabnum2 - the second table chosen for the morphing. This number doesn't express the table number directly, but the position of the table in the set sequence (starting from 0 to N-1). If this number is an integer, corresponding table will be chosen unaltered. If it contains fractional values, then an interpolation with the next adjacent table will result.

The *tabmorphak* family of opcodes is similar to the *table* family, but allows morphing between two tables chosen into a set of tables. Firstly the user has to provide a set of tables of equal length (*ifn2* [, *ifn3*, *ifn4*, ... *ifnN*]). Then he can choose a pair of tables in the set in order to perform the morphing: *atabnum1* and *atabnum2* are filled with numbers (zero represents the first table in the set, 1 the second, 2 the third and so on). Then determine the morphing between the two chosen tables with the *kweightpoint* parameter. After that the resulting table can be indexed with the *aindex* parameter like a normal table opcode. If the value of this parameter surpasses the length of tables (which must be the same for all tables), then it is wrapped around.

tabmorphak works at a-rate, but *kweightpoint*, *ktabnum1* and *ktabnum2* are working at k-rate, making it more efficient than *tabmorphak*, since there are less calculations. Except the rate of these three arguments, it is identical to *tabmorphak*.

See Also

table, tabmorph, tabmorphi, tabmorpha, ftmorf,

Credits

Author: Gabriel Maldonado

New in version 5.06

tabmorphi

tabmorphi — Allow morphing between a set of tables with interpolation.

Description

tabmorphi allows morphing between a set of tables of the same size, by means of a weighted average between two currently selected tables.

Syntax

```
kout tabmorphi kindex, kweightpoint, ktabnum1, ktabnum2, \  
ifn1, ifn2 [, ifn3, ifn4, ... ifnN]
```

Initialization

ifn1, ifn2 [, ifn3, ifn4,...ifnN] - function table numbers. This is a set of chosen tables the user want to use in the morphing. All tables must have the same length. Be aware that only two of these tables can be chosen for the morphing at one time. Since it is possible to use non-integer numbers for the *ktabnum1* and *ktabnum2* arguments, the morphing is the result from the interpolation between adjacent consecutive tables of the set.

Performance

kout - The output value for index *kindex*, resulting from morphing two tables (see below).

kindex - main index index of the morphed resultant table. The range is 0 to table_length (not included).

kweightpoint - the weight of the influence of a pair of selected tables in the morphing. The range of this argument is 0 to 1. A zero makes it output the first table unaltered, a 1 makes it output the second table of the pair unaltered. All intermediate values between 0 and 1 determine the gradual morphing between the two tables of the pair.

ktabnum1 - the first table chosen for the morphing. This number doesn't express the table number directly, but the position of the table in the set sequence (starting from 0 to N-1). If this number is an integer, the corresponding table will be chosen unaltered. If it contains fractional values, then an interpolation with the next adjacent table will result.

ktabnum2 - the second table chosen for the morphing. This number doesn't express the table number directly, but the position of the table in the set sequence (starting from 0 to N-1). If this number is an integer, corresponding table will be chosen unaltered. If it contains fractional values, then an interpolation with the next adjacent table will result.

The *tabmorphi* family of opcodes is similar to the *table* family, but allows morphing between two tables chosen into a set of tables. Firstly the user has to provide a set of tables of equal length (*ifn2 [, ifn3, ifn4,...ifnN]*). Then he can choose a pair of tables in the set in order to perform the morphing: *ktabnum1* and *ktabnum2* are filled with numbers (zero represents the first table in the set, 1 the second, 2 the third and so on). Then determine the morphing between the two chosen tables with the *kweightpoint* parameter. After that the resulting table can be indexed with the *kindex* parameter like a normal table opcode. If the value of this parameter surpasses the length of tables (which must be the same for all tables), then it is wrapped around.

tabmorphi is identical to *tabmorph*, but it performs linear interpolation for non-integer values of *kindex*, much like *tablei*.

See Also

table, tabmorph, tabmorpha, tabmorphak, ftmorf,

Credits

Author: Gabriel Maldonado

New in version 5.06

tabplay

tabplay — Playing-back control signals.

Description

Plays-back control-rate signals on trigger-temporization basis.

Syntax

```
tabplay ktrig, knumtics, kfn, kout1 [,kout2,..., koutN]
```

Performance

ktrig -- starts playing when non-zero.

knumtics -- stop recording or reset playing pointer to zero when the number of tics defined by this argument is reached.

kfn -- table where k-rate signals are recorded.

kout1,...,koutN -- playback output signals.

The *tabplay* and *tabrec* opcodes allow to record/playback control signals on trigger-temporization basis.

tabplay plays back a group of k-rate signals, previously recorded by *tabrec* into a table. Each time *ktrig* argument is triggered, an internal counter is increased of one unit. After *knumtics* trigger impluses are received by *ktrig* argument, the internal counter is zeroed and playback is restarted from the beginning, in looping style.

These opcodes can be used like a sort of ``middle-term" memory that ``remembers" generated signals. Such memory can be used to supply generative music with a coherent iterative compositional structure.

See Also

tabrec

Credits

Written by Gabriel Maldonado.

tambourine

tambourine — Semi-physical model of a tambourine sound.

Description

tambourine is a semi-physical model of a tambourine sound. It is one of the PhISEM percussion opcodes. PhISEM (Physically Informed Stochastic Event Modeling) is an algorithmic approach for simulating collisions of multiple independent sound producing objects.

Syntax

```
ares tambourine kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] \  
      [, ifreq1] [, ifreq2]
```

Initialization

idettack -- period of time over which all sound is stopped

inum (optional) -- The number of beads, teeth, bells, timbrels, etc. If zero, the default value is 32.

idamp (optional) -- the damping factor, as part of this equation:

$\text{damping_amount} = 0.9985 + (\text{idamp} * 0.002)$

The default *damping_amount* is 0.9985 which means that the default value of *idamp* is 0. The maximum *damping_amount* is 1.0 (no damping). This means the maximum value for *idamp* is 0.75.

The recommended range for *idamp* is usually below 75% of the maximum value.

imaxshake (optional, default=0) -- amount of energy to add back into the system. The value should be in range 0 to 1.

ifreq (optional) -- the main resonant frequency. The default value is 2300.

ifreq1 (optional) -- the first resonant frequency. The default value is 5600.

ifreq2 (optional) -- the second resonant frequency. The default value is 8100.

Performance

kamp -- Amplitude of output. Note: As these instruments are stochastic, this is only an approximation.

Examples

Here is an example of the tambourine opcode. It uses the file *tambourine.csd* [examples/tambourine.csd].

Example 498. Example of the tambourine opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o tambourine.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1: An example of a tambourine.
instr 01
  al tambourine 15000, 0.01

  out a1
endin

</CsInstruments>
<CsScore>

i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

bamboo, dripwater, guiro, sleighbells

Credits

Author: Perry Cook, part of the PhISEM (Physically Informed Stochastic Event Modeling)
Adapted by John ffitch
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 4.07

Added notes by Rasmus Ekman on May 2002.

tan

tan — Performs a tangent function.

Description

Returns the tangent of x (x in radians).

Syntax

`tan(x)` (no rate restriction)

Examples

Here is an example of the tan opcode. It uses the file *tan.csd* [examples/tan.csd].

Example 499. Example of the tan opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o tan.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  irad = 25
  i1 = tan(irad)

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = -0.134
```

See Also

cos, cosh, cosinv, sin, sinh, sininv, tan, taninv

Credits

Written by John ffitch.

New in version 3.47

Example written by Kevin Conder.

tanh

tanh — Performs a hyperbolic tangent function.

Description

Returns the hyperbolic tangent of x (x in radians).

Syntax

`tanh(x)` (no rate restriction)

Examples

Here is an example of the tanh opcode. It uses the file *tanh.csd* [examples/tanh.csd].

Example 500. Example of the tanh opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o tanh.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  irad = 1
  i1 = tanh(irad)

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 0.762
```

See Also

cos, cosh, cosinv, sin, sinh, sininv, tan, taninv

Credits

Author: John ffitch

New in version 3.47

Example written by Kevin Conder.

taninv

taninv — Performs an arctangent function.

Description

Returns the arctangent of x (x in radians).

Syntax

taninv(x) (no rate restriction)

Examples

Here is an example of the taninv opcode. It uses the file *taninv.csd* [examples/taninv.csd].

Example 501. Example of the taninv opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o taninv.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  irad = 0.5
  i1 = taninv(irad)

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 0.464
```

See Also

cos, cosh, cosinv, sin, sinh, sininv, tan, tanh, taninv2

Credits

Author: John ffitch

New in version 3.48

Example written by Kevin Conder.

taninv2

taninv2 — Returns an arctangent.

Description

Returns the arctangent of iy/ix , ky/kx , or ay/ax .

Syntax

ares **taninv2** ay, ax

ires **taninv2** iy, ix

kres **taninv2** ky, kx

Returns the arctangent of iy/ix , ky/kx , or ay/ax . If y is zero, *taninv2* returns zero regardless of the value of x . If x is zero, the return value is:

- $\pi/2$, if y is positive.
- $-\pi/2$, if y is negative.
- 0 , if y is 0 .

Initialization

iy, *ix* -- values to be converted

Performance

ky, *kx* -- control rate signals to be converted

ay, *ax* -- audio rate signals to be converted

Examples

Here is an example of the taninv2 opcode. It uses the file *taninv2.csd* [examples/taninv2.csd].

Example 502. Example of the taninv2 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o taninv2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
```

```
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Returns the arctangent for 1/2.
il taninv2 1, 2

print il
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  il = 0.464
```

See Also

taninv

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
April 1998

Example written by Kevin Conder.

New in Csound version 3.48

Corrected on May 2002, thanks to Istvan Varga.

tbvcf

tbvcf — Models some of the filter characteristics of a Roland TB303 voltage-controlled filter.

Description

This opcode attempts to model some of the filter characteristics of a Roland TB303 voltage-controlled filter. Euler's method is used to approximate the system, rather than traditional filter methods. Cutoff frequency, Q, and distortion are all coupled. Empirical methods were used to try to untwine, but frequency is only approximate as a result. Future fixes for some problems with this opcode may break existing orchestras relying on this version of *tbvcf*.

Syntax

```
ares tbvcf asig, xfco, xres, kdist, kasym [, iskip]
```

Initialization

iskip (optional, default=0) -- if non zero skip the initialisation of the filter. (New in Csound version 4.23f13 and 5.0)

Performance

asig -- input signal. Should be normalized to ± 1 .

xfco -- filter cutoff frequency. Optimum range is 10,000 to 1500. Values below 1000 may cause problems.

xres -- resonance or Q. Typically in the range 0 to 2.

kdist -- amount of distortion. Typical value is 2. Changing *kdist* significantly from 2 may cause odd interaction with *xfco* and *xres*.

kasym -- asymmetry of resonance. Typically in the range 0 to 1.

Examples

Here is an example of the *tbvcf* opcode. It uses the file *tbvcf.csd* [examples/tbvcf.csd].

Example 503. Example of the *tbvcf* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o tbvcf.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

;-----
; TBVCF Test
; Coded by Hans Mikelson December, 2000
;-----
sr = 44100 ; Sample rate
```

```
kr = 4410 ; Kontrol rate
ksmps = 10 ; Samples/Kontrol period
nchnls = 2 ; Normal stereo
zakinit 50, 50

instr 10

idur = p3 ; Duration
iamp = p4 ; Amplitude
ifqc = cpspch(p5) ; Pitch to frequency
ipanl = sqrt(p6) ; Pan left
ipanr = sqrt(1-p6) ; Pan right
iq = p7
idist = p8
iasym = p9

kdclock linseg 0, .002, 1, idur-.004, 1, .002, 0 ; Declick envelope

kfco expseg 10000, idur, 1000 ; Frequency envelope

ax vco 1, ifqc, 2, 1 ; Square wave
ay tbvcf ax, kfco, iq, idist, iasym ; TB-VCF
ay buthp ay/1, 100 ; Hi-pass

outs ay*iamp*ipanl*kdclock, ay*iamp*ipanr*kdclock
endin

</CsInstruments>
<CsScore>

f1 0 65536 10 1

; TeeBee Test
; Sta Dur Amp Pitch Pan Q Dist1 Asym
i10 0 0.2 32767 7.00 .5 0.0 2.0 0.0
i10 0.3 0.2 32767 7.00 .5 0.8 2.0 0.0
i10 0.6 0.2 32767 7.00 .5 1.6 2.0 0.0
i10 0.9 0.2 32767 7.00 .5 2.4 2.0 0.0
i10 1.2 0.2 32767 7.00 .5 3.2 2.0 0.0
i10 1.5 0.2 32767 7.00 .5 4.0 2.0 0.0
i10 1.8 0.2 32767 7.00 .5 0.0 2.0 0.25
i10 2.1 0.2 32767 7.00 .5 0.8 2.0 0.25
i10 2.4 0.2 32767 7.00 .5 1.6 2.0 0.25
i10 2.7 0.2 32767 7.00 .5 2.4 2.0 0.25
i10 3.0 0.2 32767 7.00 .5 3.2 2.0 0.25
i10 3.3 0.2 32767 7.00 .5 4.0 2.0 0.25
i10 3.6 0.2 32767 7.00 .5 0.0 2.0 0.5
i10 3.9 0.2 32767 7.00 .5 0.8 2.0 0.5
i10 4.2 0.2 32767 7.00 .5 1.6 2.0 0.5
i10 4.5 0.2 32767 7.00 .5 2.4 2.0 0.5
i10 4.8 0.2 32767 7.00 .5 3.2 2.0 0.5
i10 5.1 0.2 32767 7.00 .5 4.0 2.0 0.5
i10 5.4 0.2 32767 7.00 .5 0.0 2.0 0.75
i10 5.7 0.2 32767 7.00 .5 0.8 2.0 0.75
i10 6.0 0.2 32767 7.00 .5 1.6 2.0 0.75
i10 6.3 0.2 32767 7.00 .5 2.4 2.0 0.75
i10 6.6 0.2 32767 7.00 .5 3.2 2.0 0.75
i10 6.9 0.2 32767 7.00 .5 4.0 2.0 0.75
i10 7.2 0.2 32767 7.00 .5 0.0 2.0 1.0
i10 7.5 0.2 32767 7.00 .5 0.8 2.0 1.0
i10 7.8 0.2 32767 7.00 .5 1.6 2.0 1.0
i10 8.1 0.2 32767 7.00 .5 2.4 2.0 1.0
i10 8.4 0.2 32767 7.00 .5 3.2 2.0 1.0
i10 8.7 0.2 32767 7.00 .5 4.0 2.0 1.0
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Hans Mikelson
December, 2000 -- January, 2001

New in Csound 4.10

tempest

tempest — Estimate the tempo of beat patterns in a control signal.

Description

Estimate the tempo of beat patterns in a control signal.

Syntax

```
ktemp tempest kin, iprd, imindur, imemdur, ihp, ithresh, ihtim, ixfdbak, \  
istartempo, ifn [, idisprd] [, itweek]
```

Initialization

iprd -- period between analyses (in seconds). Typically about .02 seconds.

imindur -- minimum duration (in seconds) to serve as a unit of tempo. Typically about .2 seconds.

imemdur -- duration (in seconds) of the *kin* short-term memory buffer which will be scanned for periodic patterns. Typically about 3 seconds.

ihp -- half-power point (in Hz) of a low-pass filter used to smooth input *kin* prior to other processing. This will tend to suppress activity that moves much faster. Typically 2 Hz.

ithresh -- loudness threshold by which the low-passed *kin* is center-clipped before being placed in the short-term buffer as tempo-relevant data. Typically at the noise floor of the incoming data.

ihtim -- half-time (in seconds) of an internal forward-masking filter that masks new *kin* data in the presence of recent, louder data. Typically about .005 seconds.

ixfdbak -- proportion of this unit's *anticipated value* to be mixed with the incoming *kin* prior to all processing. Typically about .3.

istartempo -- initial tempo (in beats per minute). Typically 60.

ifn -- table number of a stored function (drawn left-to-right) by which the short-term memory data is attenuated over time.

idisprd (optional) -- if non-zero, display the short-term past and future buffers every *idisprd* seconds (normally a multiple of *iprd*). The default value is 0 (no display).

itweek (optional) -- fine-tune adjust this unit so that it is stable when analyzing events controlled by its own output. The default value is 1 (no change).

Performance

tempest examines *kin* for amplitude periodicity, and estimates a current tempo. The input is first low-pass filtered, then center-clipped, and the residue placed in a short-term memory buffer (attenuated over time) where it is analyzed for periodicity using a form of autocorrelation. The period, expressed as a *tempo* in beats per minute, is output as *ktemp*. The period is also used internally to make predictions about future amplitude patterns, and these are placed in a buffer adjacent to that of the input. The two adjacent buffers can be periodically displayed, and the predicted values optionally mixed with the incoming signal to simulate expectation.

This unit is useful for sensing the metric implications of any k-signal (e.g.- the RMS of an audio signal, or the second derivative of a conducting gesture), before sending to a *tempo* statement.

Examples

Here is an example of the `tempest` opcode. It uses the file `tempest.csd` [examples/tempest.csd], and `beats.wav` [examples/beats.wav].

Example 504. Example of the `tempest` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o tempest.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use the "beats.wav" sound file.
asig soundin "beats.wav"
; Extract the pitch and the envelope.
kcps, krms pitchamdf asig, 150, 500, 200

iprd = 0.01
imindur = 0.1
imemdur = 3
ihp = 1
ithresh = 30
ihtim = 0.005
ixfdbak = 0.05
istartempo = 110
ifn = 1

; Estimate its tempo.
k1 tempest krms, iprd, imindur, imemdur, ihp, ithresh, ihtim, ixfdbak, istartempo, ifn
printk2 k1

out asig
endin

</CsInstruments>
<CsScore>

; Table #1, a declining line.
f 1 0 128 16 1 128 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

The tempo of the audio file “beats.wav” is 120 beats per minute. In this examples, `tempest` will print out its best guess as the audio file plays. Its output should include lines like this:

```
. i1 118.24654
. i1 121.72949
```

tempo

tempo — Apply tempo control to an uninterpreted score.

Description

Apply tempo control to an uninterpreted score.

Syntax

```
tempo ktempo, istartempo
```

Initialization

istartempo -- initial tempo (in beats per minute). Typically 60.

Performance

ktempo -- The tempo to which the score will be adjusted.

tempo allows the performance speed of Csound scored events to be controlled from within an orchestra. It operates only in the presence of the Csound *-t* flag. When that flag is set, scored events will be performed from their uninterpreted p2 and p3 (beat) parameters, initially at the given command-line tempo. When a *tempo* statement is activated in any instrument (*ktempo* 0.), the operating tempo will be adjusted to *ktempo* beats per minute. There may be any number of *tempo* statements in an orchestra, but coincident activation is best avoided.

Examples

Here is an example of the tempo opcode. Remember, it only works if you use the *-t* flag with Csound. The example uses the file *tempo.csd* [examples/tempo.csd].

Example 505. Example of the tempo opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o tempo.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kval tempoval

  printk 0.1, kval

  ; If the fourth p-field is 1, increase the tempo.
  if (p4 == 1) kgoto speedup
```

```
    kgoto playit

speedup:
    ; Increase the tempo to 150 beats per minute.
    tempo 150, 60

playit:

    al oscil 10000, 440, 1
    out al
    endin

</CsInstruments>
<CsScore>

    ; Table #1, a sine wave.
    f 1 0 16384 10 1

    ; p4 = plays at a faster tempo (when p4=1).
    ; Play Instrument #1 at the normal tempo, repeat 3 times.
    r3
    i 1 00.00 00.25 0
    i 1 00.25 00.25 0
    i 1 00.50 00.25 0
    i 1 00.75 00.25 0
    s

    ; Play Instrument #1 at a faster tempo, repeat 3 times.
    r3
    i 1 00.00 00.25 1
    i 1 00.25 00.25 0
    i 1 00.50 00.25 0
    i 1 00.75 00.25 0
    s

e

</CsScore>
</CsoundSynthesizer>
```

See Also

tempoval

Credits

Example written by Kevin Conder.

tempoval

tempoval — Reads the current value of the tempo.

Description

Reads the current value of the tempo.

Syntax

```
kres tempoval
```

Performance

kres -- the value of the tempo. If you use a positive value with the *-t* command-line flag, *tempoval* returns the percentage increase/decrease from the original tempo of 60 beats per minute. If you don't, its value will be 60 (for 60 beats per minute).

Examples

Here is an example of the *tempoval* opcode. Remember, it only works if you use the *-t* flag with Csound. It uses the file *tempoval.csd* [examples/tempoval.csd].

Example 506. Example of the tempoval opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o tempoval.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Adjust the tempo to 120 beats per minute.
tempo 120, 60

; Get the tempo value.
kval tempoval

printks "kval = %f\\n", 0.1, kval
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
```

```
</CsoundSynthesizer>
```

Since 120 beats per minute is a 50% increase over the original 60 beats per minute, its output should include lines like:

```
kval = 0.500000
```

See Also

tempo and *miditempo*

Credits

Example written by Kevin Conder.

New in version 4.15

December 2002. Thanks to Drake Wilson for pointing out unclear documentation.

tigoto

tigoto — Transfer control at i-time when a new note is being tied onto a previously held note

Description

Similar to *igoto* but effective only during an i-time pass at which a new note is being “tied” onto a previously held note. (See *i Statement*) It does not work when a tie has not taken place. Allows an instrument to skip initialization of units according to whether a proposed tie was in fact successful. (See also *tival*, *delay*).

Syntax

```
tigoto label
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

See Also

cigoto, *goto*, *if*, *igoto*, *kgoto*, *timeout*

Credits

Added a note by Jim Aikin.

timedseq

timedseq — Time Variant Sequencer

Description

An event-sequencer in which time can be controlled by a time-pointer. Sequence data are stored into a table.

Syntax

```
ktrig timedseq ktimpnt, ifn, kp1 [,kp2, kp3, ...,kpN]
```

Initialization

ifn -- number of table containing sequence data.

Performance

ktri -- output trigger signal

ktimpnt -- time pointer into sequence file, in seconds.

kp1,...,kpN -- output p-fields of notes. kp2 meaning is relative action time and kp3 is the duration of notes in seconds.

timedseq is a sequencer that allows to schedule notes starting from a user sequence, and depending from an external timing given by a time-pointer value (*ktimpnt* argument). User should fill table *ifn* with a list of notes, that can be provided in an external text file by using GEN23, or by typing it directly in the orchestra (or score) file with GEN02. The format of the text file containing the sequence is made up simply by rows containing several numbers separated by space (similarly to normal Csound score). The first value of each row must be a positive or null value, except for a special case that will be explained below. This first value is normally used to define the instrument number corresponding to that particular note (like normal score). The second value of each row must contain the action time of corresponding note and the third value its duration. This is an example:

```
0 0      0.25 1  93
0 0.25  0.25 2  63
0 0.5   0.25 3  91
0 0.75  0.25 4  70
0 1     0.25 5  83
0 1.25  0.25 6  75
0 1.5   0.25 7  78
0 1.75  0.25 8  78
0 2     0.25 9  83
0 2.25  0.25 10 70
0 2.5   0.25 11 54
0 2.75  0.25 12 80
-1 3    -1   -1 -1 ;; last row of the sequence
```

In this example, the first value of each row is always zero (it is a dummy value, but this p-field can be used, for example, to express a MIDI channel or an instrument number), except the last row, that begins with -1. This value (-1) is a special value, that indicates the end of sequence. It has itself an action time, because sequences can be looped. So the previous sequence has a default duration of 3 seconds, being value 3 the last action time of the sequence.

It is important that ALL lines contains the same number of values (in the example all rows contains exactly 5 values). The number of values contained by each row, MUST be the number of kpXX output arguments (notice that, even if kp1, kp2 etc. are placed at the right of the opcode, they are output arguments, not input arguments).

ktimpnt argument provide the real temporization of the sequence. Actually the passage of time through sequence is specified by ktimpnt itself, which represents the time in seconds. ktimpnt must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the sequence file, in the same way of pvoc or lpread. When ktimpnt crosses the action time of a note, a trigger signal is sent to ktrig output argument, and kp1, kp2,...kpN arguments are updated with the values of that note. This information can then be used with schedk or schedkwhen to actually activate note events. Notice that kp1,...kpn data can be further processed (for example delayed with delayk, transposed, etc.) before feeding schedk or schedkwhen.

timepoint can be controlled by linear signal, for example:

```
ktimpnt line      0,p3,3 ; orignal sequence duration was 3 secs
ktrig   timedseq ktimpnt,1,kp1,kp2,kp3,kp4,kp5
        schedk   ktrig, 105, 2, 0, kp3,kp4,kp5
```

in this case the complete sequence (with orignal duration of 3 seconds) will be played in p3 seconds.

You can loop a sequence by contolling it with a phasor:

```
kphs     phasor    1/3
ktimpnt  =         kphs * 3
ktrig    timedseq ktimpnt,1,kp1,kp2,kp3,kp4,kp5
        schedk    ktrig, 105, 2, 0, kp3,kp4,kp5
```

Obviously you can play only a fragment of the sequence, read it backward, and non-linearly access sequence data in the same way of pvoc and lpread opcodes.

With timedseq opcode you can do almost all things of a normal score, except you have the following limitations: 1. You can't have two notes exactly starting with the same action time; actually at least a k-cycle should separate timing of two notes (otherwise the schedk mechanism eats one of them). 2. all notes of the sequence must have the same number of p-fields (even if they activate different instruments). You can remedy this limitation by filling with dummy values notes that belongs to instruments with less p-fields than other ones.

See Also

GEN02, GEN23, seqtime, seqtime2, trigseq

Credits

Author: Gabriel Maldonado

timeinstk

timeinstk — Read absolute time in k-rate cycles.

Description

Read absolute time, in k-rate cycles, since the start of an instance of an instrument. Called at both i-time as well as k-time.

Syntax

```
kres timeinstk
```

```
kres timeinsts
```

Performance

timeinstk is for time in k-rate cycles. So with:

```
sr      = 44100  
kr      = 6300  
ksmps   = 7
```

then after half a second, the *timek* opcode would report 3150. It will always report an integer.

timeinstk produces a k-rate variable for output. There are no input parameters.

timeinstk is similar to *timek* except it returns the time since the start of this instance of the instrument.

Examples

Here is an example of the *timeinstk* opcode. It uses the file *timeinstk.csd* [examples/timeinstk.csd].

Example 507. Example of the *timeinstk* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out   Audio in  
-odac         -iadc      ;;RT audio I/O  
; For Non-realtime output leave only the line below:  
; -o timeinstk.wav -W ;; for file output any platform  
</CsOptions>  
<CsInstruments>  
  
; Initialize the global variables.  
sr = 44100  
kr = 4410  
ksmps = 10  
nchnls = 1  
  
; Instrument #1.  
instr 1  
; Print out the value from timeinstk every half-second.
```

```
k1 timeinstk
printks "k1 = %f samples\\n", 0.5, k1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1 = 1.000000 samples
k1 = 2205.000000 samples
k1 = 4410.000000 samples
k1 = 6615.000000 samples
k1 = 8820.000000 samples
```

See Also

timeinsts, timek, times

Credits

Author: Robin Whittle
Australia
May 1997

Example written by Kevin Conder.

timeinsts

timeinsts — Read absolute time in seconds.

Description

Read absolute time, in seconds, since the start of an instance of an instrument.

Syntax

```
kres timeinsts
```

Performance

Time in seconds is available with *timeinsts*. This would return 0.5 after half a second.

timeinsts produces a k-rate variable for output. There are no input parameters.

timeinsts is similar to *times* except it returns the time since the start of this instance of the instrument.

Examples

Here is an example of the timeinsts opcode. It uses the file *timeinsts.csd* [examples/timeinsts.csd].

Example 508. Example of the timeinsts opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o timeinsts.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the value from timeinsts every half-second.
k1 timeinsts
printks "k1 = %f seconds\\n", 0.5, k1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```


Its output should include lines like this:

```
k1 = 0.000227 seconds  
k1 = 0.500000 seconds  
k1 = 1.000000 seconds  
k1 = 1.500000 seconds  
k1 = 2.000000 seconds
```

See Also

timeinstk, timek, times

Credits

Author: Robin Whittle
Australia
May 1997

Example written by Kevin Conder.

timek

timek — Read absolute time in k-rate cycles.

Description

Read absolute time, in k-rate cycles, since the start of the performance.

Syntax

```
ires timek
```

```
kres timek
```

Performance

timek is for time in k-rate cycles. So with:

```
sr      = 44100
kr      = 6300
ksmps  = 7
```

then after half a second, the *timek* opcode would report 3150. It will always report an integer.

timek can produce a k-rate variable for output. There are no input parameters.

timek can also operate only at the start of the instance of the instrument. It produces an i-rate variable (starting with *i* or *gi*) as its output.

Examples

Here is an example of the *timek* opcode. It uses the file *timek.csd* [examples/timek.csd].

Example 509. Example of the *timek* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o timek.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the value from timek every half-second.
k1 timek
printks "k1 = %f samples\\n", 0.5, k1
```

```
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1 = 1.000000 samples
k1 = 2205.000000 samples
k1 = 4410.000000 samples
k1 = 6615.000000 samples
k1 = 8820.000000 samples
```

See Also

timeinstk, timensts, times

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.47

Example written by Kevin Conder.

times

times — Read absolute time in seconds.

Description

Read absolute time, in seconds, since the start of the performance.

Syntax

```
ires times
```

```
kres times
```

Performance

Time in seconds is available with *times*. This would return 0.5 after half a second.

times can both produce a k-rate variable for output. There are no input parameters.

times can also operate at the start of the instance of the instrument. It produces an i-rate variable (starting with *i* or *gi*) as its output.

Examples

Here is an example of the times opcode. It uses the file *times.csd* [examples/times.csd].

Example 510. Example of the times opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o times.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print out the value from times every half-second.
k1 times
printks "k1 = %f seconds\\n", 0.5, k1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

Its output should include lines like this:

```
k1 = 0.000227 seconds  
k1 = 0.500000 seconds  
k1 = 1.000000 seconds  
k1 = 1.500000 seconds  
k1 = 2.000000 seconds
```

See Also

timeinstk, timeinsts, timek

Credits

Author: Robin Whittle
Australia
May 1997

Example written by Kevin Conder.

timeout

timeout — Conditional branch during p-time depending on elapsed note time.

Description

Conditional branch during p-time depending on elapsed note time. *istrt* and *idur* specify time in seconds. The branch to *label* will become effective at time *istrt*, and will remain so for just *idur* seconds. Note that *timeout* can be reinitialized for multiple activation within a single note (see example under *reinit*).

Syntax

```
timeout istrt, idur, label
```

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (<, =, <=, ==, !=) (and = for convenience, see also under *Conditional Values*).

See Also

goto, if, igoto, kgoto, tigoto

Credits

Added a note by Jim Aikin.

tival

tival — Puts the value of the instrument's internal “tie-in” flag into the named i-rate variable.

Syntax

```
ir tival
```

Description

Puts the value of the instrument's internal “tie-in” flag into the named i-rate variable.

Initialization

Puts the value of the instrument's internal “tie-in” flag into the named i-rate variable. Assigns 1 if this note has been “tied” onto a previously held note (see *i statement*); assigns 0 if no tie actually took place. (See also *tigoto*.)

See Also

`=`, *divz*, *init*

tlineto

tlineto — Generate glissandos starting from a control signal.

Description

Generate glissandos starting from a control signal with a trigger.

Syntax

```
kres tlineto ksig, ktime, ktrig
```

Performance

kres -- Output signal.

ksig -- Input signal.

ktime -- Time length of glissando in seconds.

ktrig -- Trigger signal.

tlineto is similar to *lineto* but can be applied to any kind of signal (not only stepped signals) without producing discontinuities. Last value of each segment is sampled and held from input signal each time *ktrig* value is set to a nonzero value. Normally *ktrig* signal consists of a sequence of zeroes (see *trigger opcode*).

The effect of glissando is quite different from *port*. Since in these cases, the lines are straight. Also the context of useage is different.

See Also

lineto

Credits

Author: Gabriel Maldonado

New in Version 4.13

tone

tone — A first-order recursive low-pass filter with variable frequency response.

Description

A first-order recursive low-pass filter with variable frequency response.

tone is a 1 term IIR filter. Its formula is:

$$y_n = c1 * x_n + c2 * y_{n-1}$$

where

- $b = 2 - \cos(2 \# \text{hp/sr})$;
- $c2 = b - \sqrt{b^2 - 1.0}$
- $c1 = 1 - c2$

Syntax

```
ares tone asig, khp [, iskip]
```

Initialization

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

ares -- the output audio signal.

asig -- the input audio signal.

khp -- the response curve's half-power point, in Hertz. Half power is defined as peak power / root 2.

tone implements a first-order recursive low-pass filter in which the variable *khp* (in Hz) determines the response curve's half-power point. Half power is defined as peak power / root 2.

See Also

areson, *aresonk*, *atone*, *atonek*, *port*, *portk*, *reson*, *resonk*, *tonek*

tonek

tonek — A first-order recursive low-pass filter with variable frequency response.

Description

A first-order recursive low-pass filter with variable frequency response.

Syntax

```
kres tonek ksig, khp [, iskip]
```

Initialization

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

kres -- the output signal at control-rate.

ksig -- the input signal at control-rate.

khp -- the response curve's half-power point, in Hertz. Half power is defined as peak power / root 2.

tonek is like *tone* except its output is at control-rate rather than audio rate.

See Also

areson, aresonk, atone, atonek, port, portk, reson, resonk, tone

Credits

Author: Robin Whittle
Australia
May 1997

tonex

tonex — Emulates a stack of filters using the tone opcode.

Description

tonex is equivalent to a filter consisting of more layers of *tone* with the same arguments, serially connected. Using a stack of a larger number of filters allows a sharper cutoff. They are faster than using a larger number instances in a Csound orchestra of the old opcodes, because only one initialization and k- cycle are needed at time and the audio loop falls entirely inside the cache memory of processor.

Syntax

```
ares tonex asig, khp [, inumlayer] [, iskip]
```

Initialization

inumlayer (optional) -- number of elements in the filter stack. Default value is 4.

iskip (optional, default=0) -- initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

Performance

asig -- input signal

khp -- the response curve's half-power point. Half power is defined as peak power / root 2.

See Also

atonex, *resonx*

Credits

Author: Gabriel Maldonado (adapted by John ffitch)
Italy

New in Csound version 3.49

trandom

trandom — Generates a controlled pseudo-random number series between min and max values according to a trigger.

Description

Generates a controlled pseudo-random number series between min and max values at k-rate whenever the trigger parameter is different to 0.

Syntax

```
kout trandom ktrig, kmin, kmax
```

Performance

ktrig -- trigger (opcode produces a new random number whenever this value is not 0).

kmin -- minimum range limit

kmax -- maximum range limit

trandom is almost identical to *random* opcode, except *trandom* updates its output with a new random value only when the *ktrig* argument is triggered (i.e. whenever it is not zero).

See also

random

Credits

Written by Gabriel Maldonado.

New in Csound 5.06

tradsyn

tradsyn — Streaming partial track additive synthesis

Description

The `tradsyn` opcode takes an input containing a TRACKS pv streaming signal (as generated, for instance by `partials`), as described in Lazzarini et al, "Time-stretching using the Instantaneous Frequency Distribution and Partial Tracking", Proc.of ICMC05, Barcelona. It resynthesises the signal using linear amplitude and frequency interpolation to drive a bank of interpolating oscillators with amplitude and pitch scaling controls.

Syntax

```
asig tradsyn fin, kscal, kpitch, kmaxtracks, ifn
```

Performance

asig -- output audio rate signal

fin -- input pv stream in TRACKS format

kscal -- amplitude scaling

kpitch -- pitch scaling

kmaxtracks -- max number of tracks in resynthesis. Limiting this will cause a non-linear filtering effect, by discarding newer and higher-frequency tracks (tracks are ordered by start time and ascending frequency, respectively)

ifn -- function table containing one cycle of a sinusoid (sine or cosine)

Examples

Example 511. Example

```
ain inch 1 ; input signal
fsl,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fsl,fsi2,.003,1,3,500 ; partial tracking
aout tradsyn fst, 1, 1.5, 500, 1 ; resynthesis (up a 5th)
out aout
```

The example above shows partial tracking of an ifd-analysis signal and linear additive resynthesis with pitch shifting.

Credits

Author: Victor Lazzarini;
June 2005

New plugin in version 5

November 2004.

transeg

transeg — Constructs a user-definable envelope.

Description

Constructs a user-definable envelope.

Syntax

```
ares transeg ia, idur, itype, ib [, idur2] [, itype] [, ic] ...
```

```
kres transeg ia, idur, itype, ib [, idur2] [, itype] [, ic] ...
```

Initialization

ia -- starting value.

ib, *ic*, etc. -- value after *idur* seconds.

idur -- duration in seconds of first segment. A zero or negative value will cause all initialization to be skipped.

idur2,... *idurx* etc. -- duration in seconds of segment

itype, *itype2*, etc. -- if 0, a straight line is produced. If non-zero, then *transeg* creates the following curve, for *n* steps:

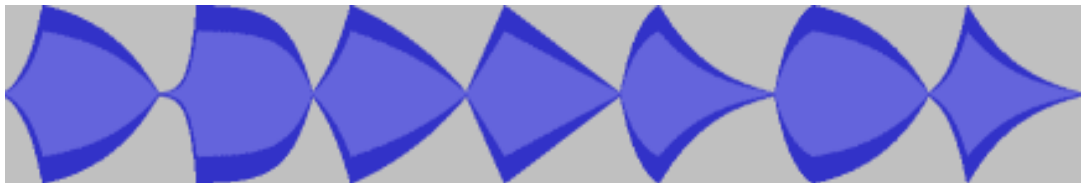
$$ibeg + (ivalue - ibeg) * (1 - \exp(i*itype/(n-1))) / (1 - \exp(itype))$$

Performance

If *itype* > 0, there is a slowly rising (concave) or slowly decaying (convex) curve, while if *itype* < 0, the curve is fast rising (convex) or fast decaying (concave). See also *GEN16*.

Examples

Here is an example of the transeg opcode. It uses the file *transeg.csd* [examples/transeg.csd]. The example produces the following output:



Output of the transeg example.

Example 512. Example of the transeg opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o transeg.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 128
nchnls = 2

0dbfs = 1

instr 1
;p4 and p5 determine the type of curve for each
;section of the envelope
kenv transeg 0.01, p3*0.25, p4, 1, p3*0.75, p5, 0.01
a1 oscil kenv, 440, 1
outs a1, a1
endin

</CsInstruments>
<CsScore>
; Table #1, a sine wave.
f 1 0 16384 10 1

i 1 0 2 2 2
i 1 + . 5 5
i 1 + . 1 1
i 1 + . 0 0
i 1 + . -2 -2
i 1 + . -2 2
i 1 + . 2 -2
e
</CsScore>
</CsoundSynthesizer>
```

See Also

expsega, expsegr, linseg, linsegr

Credits

Author: John ffitch
University of Bath, Codemist. Ltd.
Bath, UK
October 2000

New in Csound version 4.09

Thanks goes to Matt Gerassimoff for pointing out the correct command syntax.

trcross

trcross — Streaming partial track cross-synthesis.

Description

The trcross opcode takes two inputs containing TRACKS pv streaming signals (as generated, for instance by partials) and cross-synthesises them into a single TRACKS stream. Two different modes of operation are used: mode 0, cross-synthesis by multiplication of the amplitudes of the two inputs and mode 1, cross-synthesis by the substitution of the amplitudes of input 1 by the input 2. Frequencies and phases of input 1 are preserved in the output. The cross-synthesis is done by matching tracks between the two inputs using a 'search interval'. The matching algorithm will look for tracks in the second input that are within the search interval around each track in the first input. This interval can be changed at the control rate. Wider search intervals will find more matches.

Syntax

```
fsig trcross fin1, fin2, ksearch,kdepth[,kmode]
```

Performance

fsig -- output pv stream in TRACKS format

fin1 -- first input pv stream in TRACKS format.

fin2 -- second input pv stream in TRACKS format

ksearch -- search interval ratio, defining a 'search area' around each track of 1st input for matching purposes.

kdepth -- depth of effect (0-1).

kmode -- mode of cross-synthesis. 0, multiplication of amplitudes (filtering), 1, substitution of amplitudes of input 1 by input 2 (akin to vocoding). Defaults to 0.

Examples

Example 513. Example

```
ain inch 1 ; input signals
ain inch 2
fsl,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fsl,fsi2,.003,1,3,500 ; partial tracking
fsl1,fsi12 pvsifd ain,2048,512,1 ; ifd analysis (second input)
fst1 partials fsl1,fsi12,.003,1,3,500 ; partial tracking \ (second input
fcr trcross fst,fst1, 1.05, 1 ; cross-synthesis (mode 0)
aout tradsyn fcr, 1, 1, 500, 1 ; resynthesis of tracks
out aout
```

The example above shows partial tracking of two ifd-analysis signals, cross-synthesis, followed by the remix of the two parts of the spectrum and resynthesis.

Credits

Author: Victor Lazzarini;
February 2006

New in Csound5.01

trfilter

trfilter — Streaming partial track filtering.

Description

The trfilter opcode takes an input containing a TRACKS pv streaming signal (as generated, for instance by partials) and filters it using an amplitude response curve stored in a function table. The function table can have any size (no restriction to powers-of-two). The table lookup is done by linear-interpolation. It is possible to create time-varying filter curves by updating the amplitude response table with a table-writing opcode.

Syntax

```
fsig trfilter fin, kamnt, ifn
```

Performance

f`sig` -- output pv stream in TRACKS format

fin -- input pv stream in TRACKS format

kamnt -- amount of filtering (0-1)

ifn -- function table number. This will contain an amplitude response curve, from 0 Hz to the Nyquist (table indexes 0 to N). Any size is allowed. Larger tables will provide a smoother amplitude response curve. Table reading uses linear interpolation.

Examples

Example 514. Example

```
gifn ftgen 2, 0, -22050, 5 1 1000 1 4000 0.000001 17050 0.000001 ; low-pass filter curve of 22050 p
instr 1
ain inch 1 ; input signal
fsl,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fsl,fsi2,.003,1,3,500 ; partial tracking
fsc1 trfilter fst, 1, gifn ; filtering using function table 2
aout tradsyn fsc1, 1, 1, 500, 1 ; resynthesis
out aout
endin
```

The example above shows partial tracking of an ifd-analysis signal and linear additive resynthesis with low-pass filtering.

Credits

Author: Victor Lazzarini;
February 2006

New in Csound5.01

trhighest

trhighest — Extracts the highest-frequency track from a streaming track input signal.

Description

The trhighest opcode takes an input containg TRACKS pv streaming signals (as generated, for instance by partials) and outputs only the highest track. In addition it outputs two k-rate signals, corresponding to the frequency and amplitude of the highest track signal.

Syntax

```
fsig, kfr,kamp trhighest finl, kscal
```

Performance

fsig -- output pv stream in TRACKS format

kfr -- frequency (in Hz) of the highest-frequency track

kamp -- amplitude of the highest-frequency track

fin -- input pv stream in TRACKS format.

kscal -- amplitude scaling of output.

Examples

Example 515. Example

```
ain inch 1 ; input signal
fsl,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fsl,fsi2,.003,1,3,500 ; partial tracking
fhi,kfr,kamp trhighest fst,1 ; highest freq-track
aout tradsyn fhi, 1, 1, 1, 1 ; resynthesis of highest frequency
out aout
```

The example above shows partial tracking of an ifd-analysis signal, extraction of the highest frequency and resynthesis.

Credits

Author: Victor Lazzarini;
February 2006

New in Csound5.01

trigger

trigger — Informs when a krate signal crosses a threshold.

Description

Informs when a krate signal crosses a threshold.

Syntax

kout **trigger** ksig, kthreshold, kmode

Performance

ksig -- input signal

kthreshold -- trigger threshold

kmode -- can be 0, 1 or 2

Normally *trigger* outputs zeroes: only each time *ksig* crosses *kthreshold* *trigger* outputs a 1. There are three modes of using *ktrig*:

- *kmode* = 0 - (down-up) *ktrig* outputs a 1 when current value of *ksig* is higher than *kthreshold*, while old value of *ksig* was equal to or lower than *kthreshold*.
- *kmode* = 1 - (up-down) *ktrig* outputs a 1 when current value of *ksig* is lower than *kthreshold* while old value of *ksig* was equal or higher than *kthreshold*.
- *kmode* = 2 - (both) *ktrig* outputs a 1 in both the two previous cases.

Examples

Here is an example of the trigger opcode. It uses the file *trigger.csd* [examples/trigger.csd].

Example 516. Example of the trigger opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o trigger.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a square-wave low frequency oscillator as the trigger.
```

```
klf lfo 1, 10, 3
ktr trigger klf, 1, 2

; When the value of the trigger isn't equal to 0, print it out.
if (ktr == 0) kgoto contin
; Print the value of the trigger and the time it occurred.
ktm times
printks "time = %f seconds, trigger = %f\\n", 0, ktm, ktr

contin:
; Continue with processing.
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include lines like this:

```
time = 0.050340 seconds, trigger = 1.000000
time = 0.150340 seconds, trigger = 1.000000
time = 0.250340 seconds, trigger = 1.000000
time = 0.350340 seconds, trigger = 1.000000
time = 0.450340 seconds, trigger = 1.000000
time = 0.550340 seconds, trigger = 1.000000
time = 0.650340 seconds, trigger = 1.000000
time = 0.750340 seconds, trigger = 1.000000
time = 0.850340 seconds, trigger = 1.000000
time = 0.950340 seconds, trigger = 1.000000
```

Credits

Author: Gabriel Maldonado
Italy

Example written by Kevin Conder.

New in Csound version 3.49

trigseq

trigseq — Accepts a trigger signal as input and outputs a group of values.

Description

Accepts a trigger signal as input and outputs a group of values.

Syntax

```
trigseq ktrig_in, kstart, kloop, kinitndx, kfn_values, kout1 [, kout2] [...]
```

Performance

ktrig_in -- input trigger signal

kstart -- start index of looped section

kloop -- end index of looped section

kinitndx -- initial index



Note

Although *kinitndx* is listed as k-rate, it is in fact accessed only at init-time. So if you are using a k-rate argument, it must be assigned with *init*.

kfn_values -- number of a table containing a sequence of groups of values

kout1 -- output values

kout2, ... (optional) -- more output values

This opcode handles timed-sequences of groups of values stored into a table.

trigseq accepts a trigger signal (*ktrig_in*) as input and outputs group of values (contained in the *kfn_values* table) each time *ktrig_in* assumes a non-zero value. Each time a group of values is triggered, table pointer is advanced of a number of positions corresponding to the number of group-elements, in order to point to the next group of values. The number of elements of groups is determined by the number of *koutX* arguments.

It is possible to start the sequence from a value different than the first, by assigning to *initndx* an index different than zero (which corresponds to the first value of the table). Normally the sequence is looped, and the start and end of loop can be adjusted by modifying *kstart* and *kloop* arguments. User must be sure that values of these arguments (as well as *kinitndx*) correspond to valid table numbers, otherwise Csound will crash because no range-checking is implemented.

It is possible to disable loop (one-shot mode) by assigning the same value both to *kstart* and *kloop* arguments. In this case, the last read element will be the one corresponding to the value of such arguments. Table can be read backward by assigning a negative *kloop* value.

trigseq is designed to be used together with *seqtime* or *trigger* opcodes.

See Also

seqtime, *trigger*

Credits

Author: Gabriel Maldonado

November 2002. Added a note about the *kinitndx* parameter, thanks to Rasmus Ekman.

January 2003. Thanks to a note from Oeyvind Brandtsegg, I corrected the credits.

New in version 4.06

trirand

trirand — Triangular distribution random number generator

Description

Triangular distribution random number generator. This is an x-class noise generator.

Syntax

```
ares trirand krange
```

```
ires trirand krange
```

```
kres trirand krange
```

Performance

krange -- the range of the random numbers (*-krange* to *+krange*).

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the trirand opcode. It uses the file *trirand.csd* [examples/trirand.csd].

Example 517. Example of the trirand opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac        -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o trirand.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number between -1 and 1.
; krange = 1

il trirand 1
```

```
    print i1
  endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 7506.261
```

See Also

betarand, bexprnd, cauchy, exprand, gauss, linrand, pcauchy, poisson, unirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder.

trlowest

trlowest — Extracts the lowest-frequency track from a streaming track input signal.

Description

The trlowest opcode takes an input containing TRACKS pv streaming signals (as generated, for instance by partials) and outputs only the lowest track. In addition it outputs two k-rate signals, corresponding to the frequency and amplitude of the lowest track signal.

Syntax

```
fsig, kfr,kamp trlowest finl, kscal
```

Performance

fsig -- output pv stream in TRACKS format

kfr -- frequency (in Hz) of the lowest-frequency track

kamp -- amplitude of the lowest-frequency track

fin -- input pv stream in TRACKS format.

kscal -- amplitude scaling of output.

Examples

Example 518. Example

```
ain inch 1 ; input signal
fsl,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fsl,fsi2,.003,1,3,500 ; partial tracking
flow,kfr,kamp trlowest fst,1 ; lowest freq-track
aout tradsyn flow, 1, 1, 1, 1 ; resynthesis of lowest frequency
out aout
```

The example above shows partial tracking of an ifd-analysis signal, extraction of the lowest frequency and resynthesis.

Credits

Author: Victor Lazzarini;
February 2006

New in Csound5.01

trmix

trmix — Streaming partial track mixing.

Description

The trmix opcode takes two inputs containg TRACKS pv streaming signals (as generated, for instance by partials) and mixes them into a single TRACKS stream. Tracks will be mixed up to the available space (defined by the original number of FFT bins in the analysed signals). If the sum of the input tracks exceeds this space, the higher-ordered tracks in the second input will be pruned.

Syntax

```
fsig trmix fin1, fin2
```

Performance

f_{sig} -- output pv stream in TRACKS format

fin1 -- first input pv stream in TRACKS format.

fin2 -- second input pv stream in TRACKS format

Examples

Example 519. Example

```
ain inch 1 ; input signal
fs1,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fs1,fsi2,.003,1,3,500 ; partial tracking
fslo,fshi trsplit fst, 1500 ; split partial tracks at 1500 Hz
fscl trscale fshi, 1.15 ; shift the upper tracks
fmix trmix fslo,fscl ; mix the shifted and unshifted tracks
aout trdsyn fmix, 1, 1, 500, 1 ; resynthesis of tracks
out aout
```

The example above shows partial tracking of an ifd-analysis signal, frequency splitting and pitch shifting of the upper part of the spectrum, followed by the remix of the two parts of the spectrum and resynthesis.

Credits

Author: Victor Lazzarini;
February 2006

New in Csound5.01

trscale

trscale — Streaming partial track frequency scaling.

Description

The trscale opcode takes an input containing a TRACKS pv streaming signal (as generated, for instance by partials) and scales all frequencies by a k-rate amount. It can also, optionally, scale the gain of the signal by a k-rate amount (default 1). The result is pitch shifting of the input tracks.

Syntax

```
fsig trscale fin, kpitch[, kgain]
```

Performance

fsig -- output pv stream in TRACKS format

fin -- input pv stream in TRACKS format

kpitch -- frequency scaling

kgain -- amplitude scaling (default 1)

Examples

Example 520. Example

```
ain inch 1 ; input signal
fsl,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fsl,fsi2,.003,1,3,500 ; partial tracking
fsc1 trscale fst, 1.5 ; frequency scale (up a 5th)
      aout tradsyn fsc1, 1, 1, 500, 1 ; resynthesis
out aout
```

The example above shows partial tracking of an ifd-analysis signal and linear additive resynthesis with pitch shifting.

Credits

Author: Victor Lazzarini;
February 2006

New in Csound5.01

trshift

trshift — Streaming partial track frequency scaling.

Description

The trshift opcode takes an input containing a TRACKS pv streaming signal (as generated, for instance by partials) and shifts all frequencies by a k-rate frequency. It can also, optionally, scale the gain of the signal by a k-rate amount (default 1). The result is frequency shifting of the input tracks.

Syntax

```
fsig trshift fin, kpsift[, kgain]
```

Performance

fsig -- output pv stream in TRACKS format

fin -- input pv stream in TRACKS format

kshift -- frequency shift in Hz

kgain -- amplitude scaling (default 1)

Examples

Example 521. Example

```
ain inch 1 ; input signal
fs1,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fs1,fsi2,.003,1,3,500 ; partial tracking
fscl trshift fst, 150 ; frequency shift (adds 150Hz to all tracks)
aout tradsyn fscl, 1, 1, 500, 1 ; resynthesis
out aout
```

The example above shows partial tracking of an ifd-analysis signal and linear additive resynthesis with frequency shifting.

Credits

Author: Victor Lazzarini;
February 2006

New in Csound5.01

trsplit

trsplit — Streaming partial track frequency splitting.

Description

The trsplit opcode takes an input containing a TRACKS pv streaming signal (as generated, for instance by partials) and splits it into two signals according to a k-rate frequency 'split point'. The first output will contain all tracks up from 0Hz to the split frequency and the second will contain the tracks from the split frequency up to the Nyquist. It can also, optionally, scale the gain of the output signals by a k-rate amount (default 1). The result is two output signals containing only part of the original spectrum.

Syntax

```
fsiglow, fsighi trsplit fin, ksplit[, kgainlow, kgainhigh]
```

Performance

fsiglow -- output pv stream in TRACKS format containing the tracks below the split point.

fsighi -- output pv stream in TRACKS format containing the tracks above and including the split point.

fin -- input pv stream in TRACKS format

ksplit -- frequency split point in Hz

kgainlow, kgainhig -- amplitude scaling of each one of the outputs (default 1).

Examples

Example 522. Example

```
ain inch 1 ; input signal
fsl,fsi2 pvsifd ain,2048,512,1 ; ifd analysis
fst partials fsl,fsi2,.003,1,3,500 ; partial tracking
fslo,fshi trsplit fst, 1500 ; split partial tracks at 1500 Hz
aout tradsyn fshi, 1, 1, 500, 1 ; resynthesis of tracks above 1500Hz
out aout
```

The example above shows partial tracking of an ifd-analysis signal and linear additive resynthesis of the upper part of the spectrum (from 1500Hz).

Credits

Author: Victor Lazzarini;
February 2006

New in Csound5.01

turnoff

turnoff — Enables an instrument to turn itself off.

Description

Enables an instrument to turn itself off.

Syntax

turnoff

Performance

turnoff -- this p-time statement enables an instrument to turn itself off. Whether of finite duration or “held”, the note currently being performed by this instrument is immediately removed from the active note list. No other notes are affected.

Examples

The following example uses the turnoff opcode. It will cause a note to terminate when a control signal passes a certain threshold (here the Nyquist frequency). It uses the file *turnoff.csd* [examples/turnoff.csd].

Example 523. Example of the turnoff opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o turnoff.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  k1 expon 440, p3/10,880      ; begin gliss and continue
  if k1 < sr/2 kgoto contin    ; until Nyquist detected
  turnoff ; then quit

contin:
  a1 oscil 10000, k1, 1
  out a1
endin

</CsInstruments>
<CsScore>

; Table #1: an ordinary sine wave.
f 1 0 32768 10 1

; Play Instrument #1 for 4 seconds.
i 1 0 4
```


e

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

ihold

turnoff2

turnoff2 — Turn off instance(s) of other instruments at performance time.

Description

Turn off instance(s) of other instruments at performance time.

Syntax

```
turnoff2 kinsno, kmode, krelease
```

Performance

kinsno -- instrument to be turned off (can be fractional) if zero or negative, no instrument is turned off

kmode -- sum of the following values:

- 0, 1, or 2: turn off all instances (0), oldest only (1), or newest only (2)
- 4: only turn off notes with exactly matching (fractional) instrument number, rather than ignoring fractional part
- 8: only turn off notes with indefinite duration ($p3 < 0$ or MIDI)

krelease -- if non-zero, the turned off instances are allowed to release, otherwise are deactivated immediately (possibly resulting in clicks)

See Also

turnoff

Credits

Author: Istvan Varga
2005

New in Csound 5.00

turnon

turnon — Activate an instrument for an indefinite time.

Description

Activate an instrument for an indefinite time.

Syntax

```
turnon insnum [, itime]
```

Initialization

insnum -- instrument number to be activated

itime (optional, default=0) -- delay, in seconds, after which instrument *insnum* will be activated. Default is 0.

Performance

turnon activates instrument *insnum* after a delay of *itime* seconds, or immediately if *itime* is not specified. Instrument is active until explicitly turned off. (See *turnoff*.)

unirand

unirand — Uniform distribution random number generator (positive values only).

Description

Uniform distribution random number generator (positive values only). This is an x-class noise generator.

Syntax

ares **unirand** krange

ires **unirand** krange

kres **unirand** krange

Performance

krange -- the range of the random numbers (0 - *krange*).

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the unirand opcode. It uses the file *unirand.csd* [examples/unirand.csd].

Example 524. Example of the unirand opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o unirand.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Generate a random number between 0 and 1.
; krange = 1

il unirand 1
```

```
    print i1
  endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 0.840
```

See Also

seed, betarand, bexprnd, cauchy, exprand, gauss, linrand, pcauchy, poisson, trirand, weibull

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder.

upsamp

upsamp — Modify a signal by up-sampling.

Description

Modify a signal by up-sampling.

Syntax

```
ares upsamp ksig
```

Performance

upsamp converts a control signal to an audio signal. It does it by simple repetition of the kval. *upsamp* is a slightly more efficient form of the assignment, *asig = ksig*.

Examples

```
asrc  buzz      10000,440,20, 1    ; band-limited pulse train
adif  diff      asrc              ; emphasize the highs
anew  balance   adif, asrc         ; but retain the power
agate  reson    asrc,0,440        ; use a lowpass of the original
asamp  samphold anew, agate        ; to gate the new audiosig
aout  tone      asamp,100         ; smooth out the rough edges
```

See Also

diff, downsamp, integ, interp, samphold

urd

urd — A discrete user-defined-distribution random generator that can be used as a function.

Description

A discrete user-defined-distribution random generator that can be used as a function.

Syntax

```
aout = urd(ktableNum)
```

```
iout = urd(itableNum)
```

```
kout = urd(ktableNum)
```

Initialization

itableNum -- number of table containing the random-distribution function. Such table is generated by the user. See GEN40, GEN41, and GEN42. The table length does not need to be a power of 2

Performance

ktableNum -- number of table containing the random-distribution function. Such table is generated by the user. See GEN40, GEN41, and GEN42. The table length does not need to be a power of 2

urd is the same opcode as *duserrnd*, but can be used in function fashion.

For a tutorial about random distribution histograms and functions see:

- D. Lorrain. "A panoply of stochastic cannons". In C. Roads, ed. 1989. Music machine. Cambridge, Massachusetts: MIT press, pp. 351 - 379.

See Also

cuserrnd, *duserrnd*

Credits

Author: Gabriel Maldonado

New in Version 4.16

vadd

vadd — Adds a scalar value to a vector in a table.

Description

Adds a scalar value to a vector in a table.

Syntax

```
vadd ifn, kval, kelements [, kdstoffset] [, kverbose]
```

Initialization

ifn - number of the table hosting the vectorial signal to be processed

Performance

kval - scalar value to be added

kelements - number of elements of the vector

kdstoffset - index offset for the destination table (Optional, default = 0)

kverbose - Selects whether or not warnings are printed (Default=0)

vadd adds the value of *kval* to each element of the vector contained in the table *ifn*, starting from table index *kdstoffset*. This enables you to process a specific section of a table by specifying the offset and the number of elements to be processed. Offset is counted starting from 0, so if no offset is specified (or set to 0), the table will be modified from the beginning.

Note that this opcode runs at k-rate so the value of *kval* is added every control period. Use with care or you will end up with very large numbers (or use *vadd_i*).

These opcodes (*vadd*, *vmult*, *vpow* and *vexp*) perform numeric operations between a vectorial control signal (hosted by the table *ifn*), and a scalar signal (*kval*). Result is a new vector that overrides old values of *ifn*. All these opcodes work at k-rate.

Negative values for *kdstoffset* are valid. Elements from the vector that are outside the table, will be discarded, and they will not wrap around the table.

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.

In all these opcodes, the resulting vectors are stored in *ifn*, overriding the initial vectors. If you want to keep initial vector, use *vcopy* or *vcopy_i* to copy it in another table. All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2*, etc. They can also be useful in conjunction with the spectral opcodes *pvsftw* and *pvsftr*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
```



```
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

Examples

Here is an example of the vadd opcode. It uses the file *vadd.csd* [examples/vadd.csd].

Example 525. Example of the vadd opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ival = p5
ielements = p6
idstoffset = p7
kval init 25
vadd ifn1, ival, ielements, idstoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17

i2 0.0 0.2 1
i1 0.4 0.01 1 5 3 4
i2 0.8 0.2 1
i1 1.0 0.01 1 8 5 -3
i2 1.2 0.2 1
i1 1.4 0.01 1 1 10 12
i2 1.6 0.2 1
e

</CsScore>
</CsoundSynthesizer>
```

See also

vadd_i, *vmult*, *vpow* and *vexp*.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vadd_i

vadd_i — Adds a scalar value to a vector in a table.

Description

Adds a scalar value to a vector in a table.

Syntax

```
vadd_i ifn, ival, ielements [, idstoffset]
```

Initialization

ifn - number of the table hosting the vectorial signal to be processed

ielements - number of elements of the vector

ival - scalar value to be added

idstoffset - index offset for the destination table

Performance

vadd_i adds the value of *ival* to each element of the vector contained in the table *ifn*, starting from table index *idstoffset*. This enables you to process a specific section of a table by specifying the offset and the number of elements to be processed. Offset is counted starting from 0, so if no offset is specified (or set to 0), the table will be modified from the beginning.

This opcode runs only on initialization, there is a k-rate version of this opcode called *vadd*.

Negative values for *idstoffset* are valid. Elements from the vector that are outside the table, will be discarded, and they will not wrap around the table.

In all these opcodes, the resulting vectors are stored in *ifn*, overriding the initial vectors. If you want to keep initial vector, use *vcopy* or *vcopy_i* to copy it in another table. All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2*, etc. They can also be useful in conjunction with the spectral opcodes *pvsftw* and *pvsftr*.

Examples

Here is an example of the *vadd_i* opcode. It uses the file *vadd_i.csd* [examples/vadd_i.csd].

Example 526. Example of the vadd_i opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac        -iadc        ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
```

```
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ival = p5
ielements = p6
idstoffset = p7
kval init 25
vadd_i ifn1, ival, ielements, dstoffset
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17

i2 0.0 0.2 1
i1 0.4 0.01 1 2 3 4
i2 0.8 0.2 1
i1 1.0 0.01 1 0.5 5 -3
i2 1.2 0.2 1
i1 1.4 0.01 1 1.5 10 12
i2 1.6 0.2 1
e

</CsScore>
</CsSoundSynthesizer>
```

See also

vadd, *vmult_i*, *vpow_i* and *vexp_i*.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vaddv

vaddv — Performs addition between two vectorial control signals

Description

Performs addition between two vectorial control signals

Syntax

```
vaddv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

Performance

kelements - number of elements of the two vectors

kdstoffset - index offset for the destination (ifn1) table (Default=0)

ksrcoffset - index offset for the source (ifn2) table (Default=0)

kverbose - Selects whether or not warnings are printed (Default=0)

vaddv adds two vectorial control signals, that is, each element of the first vector is processed (only) with the corresponding element of the other vector. Each vectorial signal is hosted by a table (ifn1 and ifn2). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of ifn1. If you want to keep the old ifn1 vector, use *vcopy_iopcode* to copy it in another table. You can use *kdstoffset* and *ksrcoffset* to specify vectors in any location of the tables.

Negative values for *kdstoffset* and *ksrcoffset* are acceptable. If *kdstoffset* is negative, the out of range section of the vector will be discarded. If *ksrcoffset* is negative, the out of range elements will be assumed to be 0 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 0 (i.e. the destination vector will not be changed for these elements).



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

Please note that using the same table as source and destination table, might produce unexpected behavior so use with care.

This opcode works at k-rate (this means that every k-pass the vectors are added). There's an i-rate version of this opcode called *vaddv_i*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

All these operators (*vaddv*, *vsubv*, *vmultv*, *vdivv*, *vpowv*, *vexpv*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Examples

Here is an example of the *vaddv* opcode. It uses the file *vaddv.csd* [examples/vaddv.csd].

Example 527. Example of the *vaddv* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ifn2 = p5
ielements = p6
idstoffset = p7
isrcoffset = p8
kval init 25
vaddv ifn1, ifn2, ielements, idstoffset, isrcoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 15 16
f 2 0 16 -7 1 15 2
```

```
i2 0.0 0.2 1
i2 0.2 0.2 2
i1 0.4 0.01 1 2 5 3 8
i2 0.8 0.2 1
i1 1.0 0.01 1 2 5 10 -2
i2 1.2 0.2 1
i1 1.4 0.01 1 2 8 14 0
i2 1.6 0.2 1
i1 1.8 0.01 1 2 8 0 14
i2 2.0 0.2 1
i1 2.2 0.002 1 1 8 5 2
i2 2.4 0.2 1
e
```

```
</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vaddv_i

vaddv_i — Performs addition between two vectorial control signals at init time.

Description

Performs addition between two vectorial control signals at init time.

Syntax

```
vaddv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

ielements - number of elements of the two vectors

idstoffset - index offset for the destination (*ifn1*) table (Default=0)

isrcoffset - index offset for the source (*ifn2*) table (Default=0)

Performance

vaddv_i adds two vectorial control signals, that is, each element of the first vector is processed (only) with the corresponding element of the other vector. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use *vcopy_i* opcode to copy it in another table. You can use *idstoffset* and *isrcoffset* to specify vectors in any location of the tables.

Negative values for *idstoffset* and *isrcoffset* are acceptable. If *idstoffset* is negative, the out of range section of the vector will be discarded. If *isrcoffset* is negative, the out of range elements will be assumed to be 0 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 0 (i.e. the destination vector will not be changed for these elements).



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at init time. There's an k-rate version of this opcode called *vaddv*.

All these operators (*vaddv_i*, *vsubv_i*, *vmultv_i*, *vdivv_i*, *vpowv_i*, *vexpv_i*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vaget

vaget — Access values of the current buffer of an a-rate variable by indexing.

Description

Access values of the current buffer of an a-rate variable by indexing. Useful for doing sample-by-sample manipulation at k-rate without using setksmps 1.



Note

Because this opcode does not do any bounds checking, the user must be careful not to try to read values past ksmps (the size of a buffer for an a-rate variable) by using index values greater than ksmps.

Syntax

```
kval vaget kndx, avar
```

Performance

kval - value read from avar

kndx - index of the sample to read from the current buffer of the given avar variable

avar - a-rate variable to read from

Examples

Here is an example of the vaget opcode. It uses the file *vaget.csd* [examples/vaget.csd].

Example 528. Example of the vaget opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o avarget.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr=44100
ksmps=16
nchnls=2

instr 1 ; Sqrt Signal
ifreq = (p4 > 15 ? p4 : cpspch(p4))
iamp = ampdB(p5)

aout init 0
ksampnum init 0

kenv linseg 0, p3 * .5, 1, p3 * .5, 0

aout1 vco2 1, ifreq
aout2 vco2 .5, ifreq * 2
aout3 vco2 .2, ifreq * 4
```

```
aout sum          aout1, aout2, aout3

;Take Sqrt of signal, checking for negatives
kcount = 0

loopStart:

    kval vaget kcount, aout

    if (kval > .0) then
        kval = sqrt(kval)
    elseif (kval < 0) then
        kval = sqrt(-kval) * -1
    else
        kval = 0
    endif

    vaset kval, kcount, aout

loop_lt kcount, 1, ksmpls, loopStart

aout = aout * kenv
aout moogladder aout, 8000, .1
aout = aout * iamp

outs aout, aout
    endin

</CsInstruments>
<CsScore>

i1 0.0 2 440 80
e

</CsScore>
</CsoundSynthesizer>
```

See Also

vaset

Credits

Author: Steven Yi

New in version 5.04

September 2006.

valpass

valpass — Variably reverberates an input signal with a flat frequency response.

Description

Variably reverberates an input signal with a flat frequency response.

Syntax

```
ares valpass asig, krvt, xlpt, imaxlpt [, iskip] [, insmps]
```

Initialization

imaxlpt -- maximum loop time for *klpt*

iskip (optional, default=0) -- initial disposition of delay-loop data space (cf. *reson*). The default value is 0.

insmps (optional, default=0) -- delay amount, as a number of samples.

Performance

krvt -- the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude).

xlpt -- variable loop time in seconds, same as *ilpt* in *comb*. Loop time can be as large as *imaxlpt*.

This filter reiterates input with an echo density determined by loop time *ilpt*. The attenuation rate is independent and is determined by *krvt*, the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude). Its output will begin to appear immediately.

See Also

alpass, *comb*, *reverb*, *vcomb*

Credits

Author: William “Pete” Moss
University of Texas at Austin
Austin, Texas USA
January 2002

vaset

vaset — Write value of into the current buffer of an a-rate variable by index.

Description

Write values into the current buffer of an a-rate variable at the given index. Useful for doing sample-by-sample manipulation at k-rate without using setksmps 1.



Note

Because this opcode does not do any bounds checking, the user must be careful not to try to write values past ksmps (the size of a buffer for an a-rate variable) by using index values greater than ksmps.

Syntax

```
vaset kval, kndx, avar
```

Performance

kval - value to write into avar

kndx - index of the sample to write to the current buffer of the given avar variable

avar - a-rate variable to write to

Examples

Here is an example of the vaset opcode. It uses the file *vaset.csd* [examples/vaset.csd].

Example 529. Example of the vaset opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o avarset.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr=44100
ksmps=1
nchnls=2

instr 1 ; Sine Wave
ifreq = (p4 > 15 ? p4 : cpspch(p4))
iamp = ampdB(p5)

kenv adsr 0.1, 0.05, .9, 0.2

aout init 0
ksampnum init 0

kcount = 0

iperiod = sr / ifreq
```

```
i2pi = 3.14159 * 2
loopStart:
kphase = (ksampnum % iperiod) / iperiod
knewval = sin(kphase * i2pi)
    vaset knewval, kcount, aout
    ksampnum = ksampnum + 1
loop_lt kcount, 1, ksmps, loopStart
aout = aout * iamp * kenv
outs aout, aout
    endin

</CsInstruments>
<CsScore>

i1 0.0 2 440 80
e

</CsScore>
</CsoundSynthesizer>
```

See Also

vaget

Credits

Author: Steven Yi

New in version 5.04

September 2006.

vbap16

vbap16 — Distributes an audio signal among 16 channels.

Description

Distributes an audio signal among 16 channels.

Syntax

```
ar1, ..., ar16 vbap16 asig, kazim [, kelev] [, kspread]
```

Performance

asig -- audio signal to be panned

kazim -- azimuth angle of the virtual source

kelev (optional) -- elevation angle of the virtual source

kspread (optional) -- spreading of the virtual source (range 0 - 100). If value is zero, conventional amplitude panning is used. When *kspread* is increased, the number of loudspeakers used in panning increases. If value is 100, the sound is applied to all loudspeakers.

vbap16 takes an input signal, *asig*, and distribute it among 16 outputs, according to the controls *kazim* and *kelev*, and the configured loudspeaker placement. If *idim* = 2, *kelev* is set to zero. The distribution is performed using Vector Base Amplitude Panning (VBAP - See reference). VBAP distributes the signal using loudspeaker data configured with *vbaplsinit*. The signal is applied to, at most, two loudspeakers in 2-D loudspeaker configurations, and three loudspeakers in 3-D loudspeaker configurations. If the virtual source is panned outside the region spanned by loudspeakers, the nearest loudspeakers are used in panning.



Warning

Please note that all *vbap* panning opcodes require the *vbap* system to be initialized using *vbaplsinit*.

Examples

See the entry for *vbap8* for an example of usage of the *vbap* opcodes.

Reference

Ville Pulkki: “Virtual Sound Source Positioning Using Vector Base Amplitude Panning” *Journal of the Audio Engineering Society*, 1997 June, Vol. 45/6, p. 456.

See Also

vbap16move, *vbap4*, *vbap4move*, *vbap8*, *vbap8move*, *vbaplsinit*, *vbapz*, *vbapzmove*

Credits

Author: Ville Pulkki

Sibelius Academy Computer Music Studio
Laboratory of Acoustics and Audio Signal Processing
Helsinki University of Technology
Helsinki, Finland
May 2000

New in Csound Version 4.07. Input parameters accept k-rate since Csund 5.09.

vbap16move

vbap16move — Distribute an audio signal among 16 channels with moving virtual sources.

Description

Distribute an audio signal among 16 channels with moving virtual sources.

Syntax

```
ar1, ..., ar16 vbap16move asig, idur, ispread, ifldnum, ifld1 \
[, ifld2] [...]
```

Initialization

idur -- the duration over which the movement takes place.

ispread -- spreading of the virtual source (range 0 - 100). If value is zero, conventional amplitude panning is used. When *ispread* is increased, the number of loudspeakers used in panning increases. If value is 100, the sound is applied to all loudspeakers.

ifldnum -- number of fields (absolute value must be 2 or larger). If *ifldnum* is positive, the virtual source movement is a polyline specified by given directions. Each transition is performed in an equal time interval. If *ifldnum* is negative, specified angular velocities are applied to the virtual source during specified relative time intervals (see below).

ifld1, ifld2, ... -- azimuth angles or angular velocities, and relative durations of movement phases.

Performance

asig -- audio signal to be panned

vbap16move allows the use of moving virtual sources. If *ifldnum* is positive, the fields represent directions of virtual sources and equal times, *iazi1*, [*iele1*,] *iazi2*, [*iele2*,], etc. The position of the virtual source is interpolated between directions starting from the first direction and ending at the last. Each interval is interpolated in time that is fraction $\text{total_time} / \text{number_of_intervals}$ of the duration of the sound event.

If *ifldnum* is negative, the fields represent angular velocities and equal times. The first field is, however, the starting direction, *iazi1*, [*iele1*,] *iazi_vel1*, [*iele_vel1*,] *iazi_vel2*, [*iele_vel2*,] Each velocity is applied to the note that is fraction $\text{total_time} / \text{number_of_velocities}$ of the duration of the sound event. If the elevation of the virtual source becomes greater than 90 degrees or less than 0 degrees, the polarity of angular velocity is changed. Thus the elevational angular velocity produces a virtual source that moves up and down between 0 and 90 degrees.



Warning

Please note that all *vbap* panning opcodes require the *vbap* system to be initialized using *vbaplsinit*.

Examples

See the entry for *vbap8move* for an example of usage of the *vbapXmove* opcodes.

Reference

Ville Pulkki: “Virtual Sound Source Positioning Using Vector Base Amplitude Panning” *Journal of the Audio Engineering Society*, 1997 June, Vol. 45/6, p. 456.

See Also

vbap16, vbap4, vbap4move, vbap8, vbap8move, vbaplsinit, vbapz, vbapzmove, vbapzmove

Credits

Author: Ville Pulkki
Sibelius Academy Computer Music Studio
Laboratory of Acoustics and Audio Signal Processing
Helsinki University of Technology
Helsinki, Finland
May 2000

New in Csound Version 4.07

vbap4

vbap4 — Distributes an audio signal among 4 channels.

Description

Distributes an audio signal among 4 channels.

Syntax

```
ar1, ar2, ar3, ar4 vbap4 asig, kazim [, kelev] [, kspread]
```

Performance

asig -- audio signal to be panned

kazim -- azimuth angle of the virtual source

kelev (optional) -- elevation angle of the virtual source

kspread (optional) -- spreading of the virtual source (range 0 - 100). If value is zero, conventional amplitude panning is used. When *kspread* is increased, the number of loudspeakers used in panning increases. If value is 100, the sound is applied to all loudspeakers.

vbap4 takes an input signal, *asig* and distributes it among 4 outputs, according to the controls *kazim* and *kelev*, and the configured loudspeaker placement. If *idim* = 2, *kelev* is set to zero. The distribution is performed using Vector Base Amplitude Panning (VBAP - See reference). VBAP distributes the signal using loudspeaker data configured with *vbaplsinit*. The signal is applied to, at most, two loudspeakers in 2-D loudspeaker configurations, and three loudspeakers in 3-D loudspeaker configurations. If the virtual source is panned outside the region spanned by loudspeakers, the nearest loudspeakers are used in panning.



Warning

Please note that all *vbap* panning opcodes require the *vbap* system to be initialized using *vbaplsinit*.

Examples

See the entry for *vbap8* for an example of usage of the *vbap* opcodes.

Reference

Ville Pulkki: “Virtual Sound Source Positioning Using Vector Base Amplitude Panning” *Journal of the Audio Engineering Society*, 1997 June, Vol. 45/6, p. 456.

See Also

vbap16, *vbap16move*, *vbap4move*, *vbap8*, *vbap8move*, *vbaplsinit*, *vbapz*, *vbapzmove*

Credits

Author: Ville Pulkki

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Helsinki University of Technology
Helsinki, Finland
May 2000

New in Csound Version 4.06. Input parameters accept k-rate since Csund 5.09.

vbap4move

vbap4move — Distributes an audio signal among 4 channels with moving virtual sources.

Description

Distributes an audio signal among 4 channels with moving virtual sources.

Syntax

```
ar1, ar2, ar3, ar4 vbap4move asig, idur, ispread, ifldnum, ifld1 \
[, ifld2] [...]
```

Initialization

idur -- the duration over which the movement takes place.

ispread -- spreading of the virtual source (range 0 - 100). If value is zero, conventional amplitude panning is used. When *ispread* is increased, the number of loudspeakers used in panning increases. If value is 100, the sound is applied to all loudspeakers.

ifldnum -- number of fields (absolute value must be 2 or larger). If *ifldnum* is positive, the virtual source movement is a polyline specified by given directions. Each transition is performed in an equal time interval. If *ifldnum* is negative, specified angular velocities are applied to the virtual source during specified relative time intervals (see below).

ifld1, *ifld2*, ... -- azimuth angles or angular velocities, and relative durations of movement phases (see below).

Performance

asig -- audio signal to be panned

vbap4move allows the use of moving virtual sources. If *ifldnum* is positive, the fields represent directions of virtual sources and equal times, *iazi1*, [*iele1*,] *iazi2*, [*iele2*,], etc. The position of the virtual source is interpolated between directions starting from the first direction and ending at the last. Each interval is interpolated in time that is fraction $\text{total_time} / \text{number_of_intervals}$ of the duration of the sound event.

If *ifldnum* is negative, the fields represent angular velocities and equal times. The first field is, however, the starting direction, *iazi1*, [*iele1*,] *iazi_vel1*, [*iele_vel1*,] *iazi_vel2*, [*iele_vel2*,] Each velocity is applied to the note that is fraction $\text{total_time} / \text{number_of_velocities}$ of the duration of the sound event. If the elevation of the virtual source becomes greater than 90 degrees or less than 0 degrees, the polarity of angular velocity is changed. Thus the elevational angular velocity produces a virtual source that moves up and down between 0 and 90 degrees.



Warning

Please note that all *vbap* panning opcodes require the *vbap* system to be initialized using *vbaplsinit*.

Examples

See the entry for *vbap8move* for an example of usage of the *vbap* opcodes.

Reference

Ville Pulkki: “Virtual Sound Source Positioning Using Vector Base Amplitude Panning” *Journal of the Audio Engineering Society*, 1997 June, Vol. 45/6, p. 456.

See Also

vbap16, vbap16move, vbap4, vbap8, vbap8move, vbaplsinit, vbapz, vbapzmove

Credits

Author: Ville Pulkki
Sibelius Academy Computer Music Studio
Laboratory of Acoustics and Audio Signal Processing
Helsinki University of Technology
Helsinki, Finland
May 2000

New in Csound Version 4.07

vbap8

vbap8 — Distributes an audio signal among 8 channels.

Description

Distributes an audio signal among 8 channels.

Syntax

```
ar1, ..., ar8 vbap8 asig, kazim [, kelev] [, kspread]
```

Performance

asig -- audio signal to be panned

kazim -- azimuth angle of the virtual source

kelev (optional) -- elevation angle of the virtual source

kspread (optional) -- spreading of the virtual source (range 0 - 100). If value is zero, conventional amplitude panning is used. When *kspread* is increased, the number of loudspeakers used in panning increases. If value is 100, the sound is applied to all loudspeakers.

vbap8 takes an input signal, *asig*, and distributes it among 8 outputs, according to the controls *kazim* and *kelev*, and the configured loudspeaker placement. If *idim* = 2, *kelev* is set to zero. The distribution is performed using Vector Base Amplitude Panning (VBAP - See reference). VBAP distributes the signal using loudspeaker data configured with *vbaplsinit*. The signal is applied to, at most, two loudspeakers in 2-D loudspeaker configurations, and three loudspeakers in 3-D loudspeaker configurations. If the virtual source is panned outside the region spanned by loudspeakers, the nearest loudspeakers are used in panning.



Warning

Please note that all *vbap* panning opcodes require the *vbap* system to be initialized using *vbaplsinit*.

Example

Here is a simple example of the *vbap8* opcode. It uses the file *vbap8.csd* [examples/vbap8.csd].

Example 530. Example of the vbap8 opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
;-odac        -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
-o vbap8.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr      =      41000
```

```
kr      =      441
ksmps  =      100
nchnls =      4
vbaplsinit      2, 8,  0, 45, 90, 135, 200, 245, 290, 315

instr 1
asig    oscil      20000, 440, 1
a1,a2,a3,a4,a5,a6,a7,a8    vbap8    asig, p4, 0, 20 ;p4 = azimuth

;render twice with alternate outq statements
; to obtain two 4 channel .wav files:

outq      a1,a2,a3,a4
outq      a5,a6,a7,a8
; or use an 8-channel output for realtime output (set nchnls to 8):
outo a1,a2,a3,a4,a5,a6,a7,a8
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
,      azimuth
i 1 0 1      20
i 1 + .      40
i 1 + .      60
i 1 + .      80
i 1 + .     100
i 1 + .     120
i 1 + .     140
i 1 + .     160
e

</CsScore>
</CsoundSynthesizer>
```

Reference

Ville Pulkki: “Virtual Sound Source Positioning Using Vector Base Amplitude Panning” *Journal of the Audio Engineering Society*, 1997 June, Vol. 45/6, p. 456.

See Also

vbap16, vbap16move, vbap4, vbap4move, vbap8move, vbaplsinit, vbapz, vbapzmove

Credits

Author: Ville Pulkki
Sibelius Academy Computer Music Studio
Laboratory of Acoustics and Audio Signal Processing
Helsinki University of Technology
Helsinki, Finland
May 2000

New in Csound Version 4.07. Input parameters accept k-rate since Csound 5.09.

vbap8move

vbap8move — Distributes an audio signal among 8 channels with moving virtual sources.

Description

Distributes an audio signal among 8 channels with moving virtual sources.

Syntax

```
ar1, ..., ar8 vbap8move asig, idur, ispread, ifldnum, ifld1 \
[, ifld2] [...]
```

Initialization

idur -- the duration over which the movement takes place.

ispread -- spreading of the virtual source (range 0 - 100). If value is zero, conventional amplitude panning is used. When *ispread* is increased, the number of loudspeakers used in panning increases. If value is 100, the sound is applied to all loudspeakers.

ifldnum -- number of fields (absolute value must be 2 or larger). If *ifldnum* is positive, the virtual source movement is a polyline specified by given directions. Each transition is performed in an equal time interval. If *ifldnum* is negative, specified angular velocities are applied to the virtual source during specified relative time intervals (see below).

ifld1, *ifld2*, ... -- azimuth angles or angular velocities, and relative durations of movement phases (see below).

Performance

asig -- audio signal to be panned

vbap8move allows the use of moving virtual sources. If *ifldnum* is positive, the fields represent directions of virtual sources and equal times, *iazi1*, [*iele1*,] *iazi2*, [*iele2*,], etc. The position of the virtual source is interpolated between directions starting from the first direction and ending at the last. Each interval is interpolated in time that is fraction $\text{total_time} / \text{number_of_intervals}$ of the duration of the sound event.

If *ifldnum* is negative, the fields represent angular velocities and equal times. The first field is, however, the starting direction, *iazi1*, [*iele1*,] *iazi_vel1*, [*iele_vel1*,] *iazi_vel2*, [*iele_vel2*,] Each velocity is applied to the note that is fraction $\text{total_time} / \text{number_of_velocities}$ of the duration of the sound event. If the elevation of the virtual source becomes greater than 90 degrees or less than 0 degrees, the polarity of angular velocity is changed. Thus the elevational angular velocity produces a virtual source that moves up and down between 0 and 90 degrees.



Warning

Please note that all *vbap* panning opcodes require the *vbap* system to be initialized using *vbaplsinit*.

Example

Here is a simple example of the *vbap8move* opcode. It uses the file *vbap8move.csd* [examples/vbap8move.csd].

Example 531. Example of the vbap8move opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o vbap4move.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 48000
ksmps = 10
nchnls = 8

;Example by Hector Centeno 2007

vbaplsinit      2, 8, 15, 65, 115, 165, 195, 245, 295, 345

    instr 1
ifldnum = 9
ispread = 30
idur = p3

;; Generate a sound source
kenv loopseg 10, 0, 0, 0, 0.5, 1, 10, 0
a1 pinkish 3000*kenv

;; Move circling around once all the speakers
aout1, aout2, aout3, aout4, aout5, aout6, aout7, aout8 vbap8move a1, idur, ispread, ifldnum, 15, 65

;; Speaker mapping
aFL = aout8 ; Front Left
aFR = aout1 ; Front Right
aMFL = aout7 ; Mid Front Left
aMFR = aout2 ; Mid Front Right
aMBL = aout6 ; Mid Back Left
aMBR = aout3 ; Mid Back Right
aBL = aout5 ; Back Left
aBR = aout4 ; Back Right

outo aFL, aFR, aMFL, aMFR, aMBL, aMBR, aBL, aBR

    endin

</CsInstruments>
<CsScore>
i1 0 30
e
</CsScore>
</CsoundSynthesizer>
```

Reference

Ville Pulkki: “Virtual Sound Source Positioning Using Vector Base Amplitude Panning” *Journal of the Audio Engineering Society*, 1997 June, Vol. 45/6, p. 456.

See Also

vbap16, vbap16move, vbap4, vbap4move, vbap8, vbaplsinit, vbapz, vbapzmove

Credits

Author: Ville Pulkki
Sibelius Academy Computer Music Studio
Laboratory of Acoustics and Audio Signal Processing

Helsinki University of Technology
Helsinki, Finland
May 2000

New in Csound Version 4.07

vbaplsinit

vbaplsinit — Configures VBAP output according to loudspeaker parameters.

Description

Configures VBAP output according to loudspeaker parameters.

Syntax

```
vbaplsinit idim, ilsnum [, idir1] [, idir2] [...] [, idir32]
```

Initialization

idim -- dimensionality of loudspeaker array. Either 2 or 3.

ilsnum -- number of loudspeakers. In two dimensions, the number can vary from 2 to 16. In three dimensions, the number can vary from 3 and 16.

idir1, *idir2*, ..., *idir32* -- directions of loudspeakers. Number of directions must be less than or equal to 16. In two-dimensional loudspeaker positioning, *idirn* is the azimuth angle respective to *n*th channel. In three-dimensional loudspeaker positioning, fields are the azimuth and elevation angles of each loudspeaker consequently (*azi1*, *ele1*, *azi2*, *ele2*, etc.).

Performance

VBAP distributes the signal using loudspeaker data configured with *vbaplsinit*. The signal is applied to, at most, two loudspeakers in 2-D loudspeaker configurations, and three loudspeakers in 3-D loudspeaker configurations. If the virtual source is panned outside the region spanned by loudspeakers, the nearest loudspeakers are used in panning.

Examples

See the entry for *vbap16move* and *vbap8* for examples of usage of the *vbap* opcodes.

Reference

Ville Pulkki: “Virtual Sound Source Positioning Using Vector Base Amplitude Panning” *Journal of the Audio Engineering Society*, 1997 June, Vol. 45/6, p. 456.

See Also

vbap16, *vbap16move*, *vbap4*, *vbap4move*, *vbap8*, *vbap8move*, *vbapz*, *vbapzmove*

Credits

Author: Ville Pulkki
Sibelius Academy Computer Music Studio
Laboratory of Acoustics and Audio Signal Processing
Helsinki University of Technology
Helsinki, Finland
May 2000

New in Csound Version 4.07

vbapz

vbapz — Writes a multi-channel audio signal to a ZAK array.

Description

Writes a multi-channel audio signal to a ZAK array.

Syntax

```
vbapz inumchnls, istartndx, asig, kazim [, kelev] [, kspread]
```

Initialization

inumchnls -- number of channels to write to the ZA array. Must be in the range 2 - 256.

istartndx -- first index or position in the ZA array to use

Performance

asig -- audio signal to be panned

kazim -- azimuth angle of the virtual source

kelev (optional) -- elevation angle of the virtual source

kspread (optional) -- spreading of the virtual source (range 0 - 100). If value is zero, conventional amplitude panning is used. When *kspread* is increased, the number of loudspeakers used in panning increases. If value is 100, the sound is applied to all loudspeakers.

The opcode *vbapz* is the multiple channel analog of the opcodes like *vbap4*, working on *inumchnls* and using a ZAK array for output.



Warning

Please note that all *vbap* panning opcodes require the *vbap* system to be initialized using *vbaplsinit*.

Examples

See the entry for *vbap8* for an example of usage of the *vbap* opcodes.

Reference

Ville Pulkki: “Virtual Sound Source Positioning Using Vector Base Amplitude Panning” *Journal of the Audio Engineering Society*, 1997 June, Vol. 45/6, p. 456.

See Also

vbap16, *vbap16move*, *vbap4*, *vbap4move*, *vbap8*, *vbap8move*, *vbaplsinit*, *vbapzmove*

Credits

John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07. Input parameters accept k-rate since Csound 5.09.

vbapzmove

vbapzmove — Writes a multi-channel audio signal to a ZAK array with moving virtual sources.

Description

Writes a multi-channel audio signal to a ZAK array with moving virtual sources.

Syntax

```
vbapzmove inumchnls, istartndx, asig, idur, ispread, ifldnum, ifld1, \
            ifld2, [...]
```

Initialization

inumchnls -- number of channels to write to the ZA array. Must be in the range 2 - 256.

istartndx -- first index or position in the ZA array to use

idur -- the duration over which the movement takes place.

ispread -- spreading of the virtual source (range 0 - 100). If value is zero, conventional amplitude panning is used. When *ispread* is increased, the number of loudspeakers used in panning increases. If value is 100, the sound is applied to all loudspeakers.

ifldnum -- number of fields (absolute value must be 2 or larger). If *ifldnum* is positive, the virtual source movement is a polyline specified by given directions. Each transition is performed in an equal time interval. If *ifldnum* is negative, specified angular velocities are applied to the virtual source during specified relative time intervals (see below).

ifld1, *ifld2*, ... -- azimuth angles or angular velocities, and relative durations of movement phases (see below).

Performance

asig -- audio signal to be panned

The opcode *vbapzmove* is the multiple channel analog of the opcodes like *vbap4move*, working on *inumchnls* and using a ZAK array for output.



Warning

Please note that all *vbap* panning opcodes require the *vbap* system to be initialized using *vbaplsinit*.

Examples

See the entry for *vbap8move* for an example of usage of the *vbap* opcodes.

Reference

Ville Pulkki: “Virtual Sound Source Positioning Using Vector Base Amplitude Panning” *Journal of the Audio Engineering Society*, 1997 June, Vol. 45/6, p. 456.

See Also

vbap16, vbap16move, vbap4, vbap4move, vbap8, vbap8move, vbaplsinit, vbapz,

Credits

John ffitch
University of Bath/Codemist Ltd.
Bath, UK
May 2000

New in Csound Version 4.07

vcella

vcella — Cellular Automata

Description

Unidimensional Cellular Automata applied to Csound vectors

Syntax

```
vcella ktrig, kreinit, ioutFunc, initStateFunc, \  
         iRuleFunc, ielements, irulelen [, iradius]
```

Initialization

ioutFunc - number of the table where the state of each cell is stored

initStateFunc - number of a table containig the initial states of each cell

iRuleFunc - number of a lookup table containing the rules

ielements - total number of cells

irulelen - total number of rules

iradius (optional) - radius of Cellular Automata. At present time CA radius can be 1 or 2 (1 is the default)

Performance

ktrig - trigger signal. Each time it is non-zero, a new generation of cells is evaluated

kreinit - trigger signal. Each time it is non-zero, state of all cells is forced to be that of *initStateFunc*.

vcella supports unidimensional cellular automata, where the state of each cell is stored in *ioutFunc*. So *ioutFunc* is a vector containing current state of each cell. This variant vector can be used together with any other vector-based opcode, such as *adsynt*, *vmap*, *vpowv* etc.

initStateFunc is an input vector containing the initial value of the row of cells, while *iRuleFunc* is an input vector containing the rules in the form of a lookup table. Notice that *initStateFunc* and *iRuleFunc* can be updated during the performance by means of other vector-based opcodes (for example *vcopy*) in order to force to change rules and status at performance time.

A new generation of cells is evaluated each time *ktrig* contains a non-zero value. Also the status of all cells can be forced to assume the status corresponding to the contents of *initStateFunc* each time *kreinit* contains a non-zero value.

Radius of CA algorithm can be 1 or 2 (optional *iradius* arguement).

Examples

Here is an example of the *vcella* opcode. It uses the file *vcella.csd* [examples/vcella.csd].

The following example uses *vcella*

Example 532. Example of the vcella opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o vcella.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
; vcella.csd
; by Anthony Kozar

; This file demonstrates some of the new opcodes available in
; Csound 5 that come from Gabriel Maldonado's CsoundAV.

sr          = 44100
kr          = 4410
ksmps      = 10
nchnls     = 1

; Cellular automata-driven oscillator bank using vcella and adsynt
instr 1
  idur      = p3
  iCarate   = p4                                ; number of times per second the CA calculates new

  ; f-tables for CA parameters
  iCAinit   = p5                                ; CA initial states
  iCARule    = p6                                ; CA rule values
  ; The rule is used as follows:
  ; the states (values) of each cell are summed with their neighboring cells within
  ; the specied radius (+/- 1 or 2 cells). Each sum is used as an index to read a
  ; value from the rule table which becomes the new state value for its cell.
  ; All new states are calculated first, then the new values are all applied
  ; simultaneously.

  ielements = ftlen(iCAinit)
  inumrules  = ftlen(iCARule)
  iradius    = 1

  ; create some needed tables
  iCAstate   ftgen 0, 0, ielements, -2, 0      ; will hold the current CA states
  ifreqs     ftgen 0, 0, ielements, -2, 0      ; will hold the oscillator frequency for each cell
  iamps      ftgen 0, 0, ielements, -2, 0      ; will hold the amplitude for each cell

  ; calculate cellular automata state
  ktrig      metro iCarate                      ; trigger the CA to update iCarate times per second
              vcella ktrig, 0, iCAstate, iCAinit, iCARule, ielements, inumrules, iradius

  ; scale CA state for use as amplitudes of the oscillator bank
              vcopy iamps, iCAstate, ielements
              vmult iamps, (1/3), ielements    ; divide by 3 since state values are 0-3

              vport iamps, .01, ielements      ; need to smooth the amplitude changes for adsynt
  ; we could use adsynt2 instead of adsynt, but it does not seem to be working

  ; i-time loop for calculating frequencies
  index      = 0
  inew       = 1
  iratio     = 1.125                            ; just major second (creating a whole tone scale)
loop1:
              tableiw inew, index, ifreqs, 0    ; 0 indicates integer indices
  inew       = inew * iratio
  index      = index + 1
  if (index < ielements) igoto loop1

  ; create sound with additive oscillator bank
  ifreqbase  = 64
  iwavefn    = 1
  iphs       = 2                                ; random oscillator phases

  kenv       linseg 0.0, 0.5, 1.0, idur - 1.0, 1.0, 0.5, 0.0
  aosc       adsynt kenv, ifreqbase, iwavefn, ifreqs, iamps, ielements, iphs

              out aosc * ampdb(68)

endin

</CsInstruments>
<CsScore>
f1 0 16384 10 1

; This example uses a 4-state cellular automata
; Possible state values are 0, 1, 2, and 3
```

```
; CA initial state  
; We have 16 cells in our CA, so the initial state table is size 16  
f10 0 16 -2 0 1 0 0 1 0 0 2 2 0 0 1 0 0 1 0  
  
; CA rule  
; The maximum sum with radius 1 (3 cells) is 9, so we need 10 values in the rule (0-9)  
f11 0 16 -2 1 0 3 2 1 0 0 2 1 0  
  
; Here is our one and only note!  
i1 0 20 4 10 11  
  
e  
  
</CsScore>  
</CsoundSynthesizer>
```

Credits

Written by: Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

Example by: Anthony Kozar

VCO

vco — Implementation of a band limited, analog modeled oscillator.

Description

Implementation of a band limited, analog modeled oscillator, based on integration of band limited impulses. *vco* can be used to simulate a variety of analog wave forms.

Syntax

```
ares vco xamp, xcps, iwave, kpw [, ifn] [, imaxd] [, ileak] [, inyx] \  
      [, iphs] [, iskip]
```

Initialization

iwave -- determines the waveform:

- *iwave* = 1 - sawtooth
- *iwave* = 2 - square/PWM
- *iwave* = 3 - triangle/saw/ramp

ifn (optional, default = 1) -- should be the table number of a of a stored sine wave. Must point to a valid table which contains a sine wave. Csound will report an error if this parameter is not set and table 1 doesn't exist.

imaxd (optional, default = 1) -- is the maximum delay time. A time of $1/lfqc$ may be required for the PWM and triangle waveform. To bend the pitch down this value must be as large as $1/(\text{minimum frequency})$.

ileak (optional, default = 0) -- if *ileak* is between zero and one ($0 < ileak < 1$) then *ileak* is used as the leaky integrator value. Otherwise a leaky integrator value of .999 is used for the saw and square waves and .995 is used for the triangle wave. This can be used to “flatten” the square wave or “straighten” the saw wave at low frequencies by setting *ileak* to .99999 or a similar value. This should give a hollow sounding square wave.

inyx (optional, default = .5) -- this is used to determine the number of harmonics in the band limited pulse. All overtones up to $sr * inyx$ will be used. The default gives $sr * .5$ ($sr/2$). For $sr/4$ use *inyx* = .25. This can generate a “fatter” sound in some cases.

iphs (optional, default = 0) -- this is a phase value. There is an artifact (bug-like feature) in *vco* which occurs during the first half cycle of the square wave which causes the waveform to be greater in magnitude than all others. The value of *iphs* has an effect on this artifact. In particular setting *iphs* to .5 will cause the first half cycle of the square wave to resemble a small triangle wave. This may be more desirable than the large wave artifact which is the current default.

iskip (optional, default = 0) -- if non zero skip the initialisation of the filter. (New in Csound version 4.23f13 and 5.0)

Performance

kpw -- determines either the pulse width (if *iwave* is 2) or the saw/ramp character (if *iwave* is 3) The value of *kpw* should be greater than 0 and less than 1. A value of 0.5 will generate either a square wave (if *iwave* is 2) or a triangle wave (if *iwave* is 3).

xamp -- determines the amplitude

xcps -- is the frequency of the wave in cycles per second.

Examples

Here is an example of the vco opcode. It uses the file *vco.csd* [examples/vco.csd].

Example 533. Example of the vco opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o vco.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 44100
ksmps = 1
nchnls = 1

; Instrument #1
instr 1
; Set the amplitude.
kamp = p4

; Set the frequency.
kcps = cpspch(p5)

; Select the wave form.
iwave = p6

; Set the pulse-width/saw-ramp character.
kpw init 0.5

; Use Table #1.
ifn = 1

; Generate the waveform.
asig vco kamp, kcps, iwave, kpw, ifn

; Output and amplification.
out asig
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 65536 10 1

; Define the score.
; p4 = raw amplitude (0-32767)
; p5 = frequency, in pitch-class notation.
; p6 = the waveform (1=Saw, 2=Square/PWM, 3=Tri/Saw-Ramp-Mod)
i 1 00 02 20000 05.00 1
i 1 02 02 20000 05.00 2
i 1 04 02 20000 05.00 3

i 1 06 02 20000 07.00 1
i 1 08 02 20000 07.00 2
i 1 10 02 20000 07.00 3

i 1 12 02 20000 09.00 1
i 1 14 02 20000 09.00 2
i 1 16 02 20000 09.00 3

i 1 18 02 20000 11.00 1
i 1 20 02 20000 11.00 2
```

```
i 1 22 02 20000 11.00 3  
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

See Also

vco2

Credits

Author: Hans Mikelson
December 1998

New in Csound version 3.50

November 2002. Corrected the documentation for the *kpw* parameter thanks to Luis Jure and Hans Mikelson.

vco2

vco2 — Implementation of a band-limited oscillator using pre-calculated tables.

Description

vco2 is similar to *vco*. But the implementation uses pre-calculated tables of band-limited waveforms (see also *GEN30*) rather than integrating impulses. This opcode can be faster than *vco* (especially if a low control-rate is used) and also allows better sound quality. Additionally, there are more waveforms and oscillator phase can be modulated at k-rate. The disadvantage is increased memory usage. For more details about *vco2* tables, see also *vco2init* and *vco2ft*.

Syntax

```
ares vco2 kamp, kcps [, imode] [, kpw] [, kphs] [, inyx]
```

Initialization

imode (optional, default=0) -- a sum of values representing the waveform and its control values.

One may use any of the following values for *imode*:

- 16: enable k-rate phase control (if set, *kphs* is a required k-rate parameter that allows phase modulation)
- 1: skip initialization

One may use exactly one of these *imode* values to select the waveform to be generated:

- 14: user defined waveform -1 (requires using the *vco2init* opcode)
- 12: triangle (no ramp, faster)
- 10: square wave (no PWM, faster)
- 8: $4 * x * (1 - x)$ (i.e. integrated sawtooth)
- 6: pulse (not normalized)
- 4: sawtooth / triangle / ramp
- 2: square / PWM
- 0: sawtooth

The default value for *imode* is zero, which means a sawtooth wave with no k-rate phase control.

inyx (optional, default=0.5) -- bandwidth of the generated waveform, as percentage (0 to 1) of the sample rate. The expected range is 0 to 0.5 (i.e. up to $sr/2$), other values are limited to the allowed range.

Setting *inyx* to 0.25 ($sr/4$), or 0.3333 ($sr/3$) can produce a “fatter” sound in some cases, although it is more likely to reduce quality.

Performance

ares -- the output audio signal.

kamp -- amplitude scale. In the case of a *imode* waveform value of 6 (a pulse waveform), the actual output level can be a lot higher than this value.

kcps -- frequency in Hz (should be in the range $-sr/2$ to $sr/2$).

kpw (optional) -- the pulse width of the square wave (*imode* waveform=2) or the ramp characteristics of the triangle wave (*imode* waveform=4). It is required only by these waveforms and ignored in all other cases. The expected range is 0 to 1, any other value is wrapped to the allowed range.



Warning

kpw must not be an exact integer value (e.g. 0 or 1) if a sawtooth / triangle / ramp (*imode* waveform=4) is generated. In this case, the recommended range is about 0.01 to 0.99. There is no such limitation for a square/PWM waveform.

kphs (optional) -- oscillator phase (depending on *imode*, this can be either an optional i-rate parameter that defaults to zero or required k-rate). Similarly to *kpw*, the expected range is 0 to 1.



Note

When a low control-rate is used, pulse width (*kpw*) and phase (*kphs*) modulation is internally converted to frequency modulation. This allows for faster processing and reduced artifacts. But in the case of very long notes and continuous fast changes in *kpw* or *kphs*, the phase may drift away from the requested value. In most cases, the phase error is at most 0.037 per hour (assuming a sample rate of 44100 Hz).

This is a problem mainly in the case of pulse width (*kpw*), where it may result in various artifacts. While future releases of *vco2* may fix such errors, the following workarounds may also be of some help:

- Use *kpw* values only in the range 0.05 to 0.95. (There are more artifacts around integer values)
- Try to avoid modulating *kpw* by asymmetrical waveforms like a sawtooth wave. Relatively slow (≤ 20 Hz) symmetrical modulation (e.g. sine or triangle), random splines (also slow), or a fixed pulse width is a lot less likely to cause synchronization problems.
- In some cases, adding random jitter (for example: random splines with an amplitude of about 0.01) to *kpw* may also fix the problem.

Examples

Here is an example of the *vco2* opcode. It uses the file *vco2.csd* [examples/vco2.csd].

Example 534. Example of the *vco2* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
```



```
; Audio out   Audio in   No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o vco2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr      = 44100
ksmps   = 10
nchnls  = 1

; user defined waveform -1: trapezoid wave with default parameters (can be
; accessed at ftables starting from 10000)
itmp    ftgen 1, 0, 16384, 7, 0, 2048, 1, 4096, 1, 4096, -1, 4096, -1, 2048, 0
ift     vco2init -1, 10000, 0, 0, 0, 1
; user defined waveform -2: fixed table size (4096), number of partials
; multiplier is 1.02 (~238 tables)
itmp    ftgen 2, 0, 16384, 7, 1, 4095, 1, 1, -1, 4095, -1, 1, 0, 8192, 0
ift     vco2init -2, ift, 1.02, 4096, 4096, 2

instr 1
kcps    expon p4, p3, p5          ; instr 1: basic vco2 example
a1      vco2 12000, kcps          ; (sawtooth wave with default
out a1                                ; parameters)
endin

instr 2
kcps    expon p4, p3, p5          ; instr 2:
kpw     linseg 0.1, p3/2, 0.9, p3/2, 0.1 ; PWM example
a1      vco2 10000, kcps, 2, kpw
out a1
endin

instr 3
kcps    expon p4, p3, p5          ; instr 3: vco2 with user
a1      vco2 14000, kcps, 14      ; defined waveform (-1)
aenv    linseg 1, p3 - 0.1, 1, 0.1, 0 ; de-click envelope
out a1 * aenv
endin

instr 4
kcps    expon p4, p3, p5          ; instr 4: vco2ft example,
kfn     vco2ft kcps, -2, 0.25    ; with user defined waveform
a1      oscilikt 12000, kcps, kfn ; (-2), and sr/4 bandwidth
out a1
endin

</CsInstruments>
<CsScore>

i 1 0 3 20 2000
i 2 4 2 200 400
i 3 7 3 400 20
i 4 11 2 100 200

f 0 14

e

</CsScore>
</CsoundSynthesizer>
```

See Also

vco, *vco2ft*, *vco2ift*, and *vco2init*.

Credits

Author: Istvan Varga

New in version 4.22

vco2ft

vco2ft — Returns a table number at k-time for a given oscillator frequency and waveform.

Description

vco2ft returns the function table number to be used for generating the specified waveform at a given frequency. This function table number can be used by any Csound opcode that generates a signal by reading function tables (like *oscilikt*). The tables must be calculated by *vco2init* before *vco2ft* is called and shared as Csound ftables (*ibasfn*).

Syntax

```
kfn vco2ft kcps, iwave [, inyx]
```

Initialization

iwave -- the waveform for which table number is to be selected. Allowed values are:

- 0: sawtooth
- 1: $4 * x * (1 - x)$ (integrated sawtooth)
- 2: pulse (not normalized)
- 3: square wave
- 4: triangle

Additionally, negative *iwave* values select user defined waveforms (see also *vco2init*).

inyx (optional, default=0.5) -- bandwidth of the generated waveform, as percentage (0 to 1) of the sample rate. The expected range is 0 to 0.5 (i.e. up to $sr/2$), other values are limited to the allowed range.

Setting *inyx* to 0.25 ($sr/4$), or 0.3333 ($sr/3$) can produce a “fatter” sound in some cases, although it is more likely to reduce quality.

Performance

kfn -- the ftable number, returned at k-rate.

kcps -- frequency in Hz, returned at k-rate. Zero and negative values are allowed. However, if the absolute value exceeds $sr/2$ (or $sr * inyx$), the selected table will contain silence.

Examples

See the example for the *vco2* opcode.

See Also

vco2ift, *vco2init*, and *vco2*.

Credits

Author: Istvan Varga

New in version 4.22

vco2ift

vco2ift — Returns a table number at i-time for a given oscillator frequency and waveform.

Description

vco2ift is the same as *vco2ft*, but works at i-time. It is suitable for use with opcodes that expect an i-rate table number (for example, *oscili*).

Syntax

```
ifn vco2ift icps, iwave [, inyx]
```

Initialization

ifn -- the ftable number.

icps -- frequency in Hz. Zero and negative values are allowed. However, if the absolute value exceeds $sr/2$ (or $sr * inyx$), the selected table will contain silence.

iwave -- the waveform for which table number is to be selected. Allowed values are:

- 0: sawtooth
- 1: $4 * x * (1 - x)$ (integrated sawtooth)
- 2: pulse (not normalized)
- 3: square wave
- 4: triangle

Additionally, negative *iwave* values select user defined waveforms (see also *vco2init*).

inyx (optional, default=0.5) -- bandwidth of the generated waveform, as percentage (0 to 1) of the sample rate. The expected range is 0 to 0.5 (i.e. up to $sr/2$), other values are limited to the allowed range.

Setting *inyx* to 0.25 ($sr/4$), or 0.3333 ($sr/3$) can produce a “fatter” sound in some cases, although it is more likely to reduce quality.

See Also

vco2ft, *vco2init*, and *vco2*.

Credits

Author: Istvan Varga

New in version 4.22

vco2init

vco2init — Calculates tables for use by vco2 opcode.

Description

vco2init calculates tables for use by *vco2* opcode. Optionally, it is also possible to access these tables as standard Csound function tables. In this case, *vco2ft* can be used to find the correct table number for a given oscillator frequency.

In most cases, this opcode is called from the orchestra header. Using *vco2init* in instruments is possible but not recommended. This is because replacing tables during performance can result in a Csound crash if other opcodes are accessing the tables at the same time.

Note that *vco2init* is not required for *vco2* to work (tables are automatically allocated by the first *vco2* call, if not done yet), however it can be useful in some cases:

- Pre-calculate tables at orchestra load time. This is useful to avoid generating the tables during performance, which could interrupt real-time processing.
- Share the tables as Csound ftables. By default, the tables can be accessed only by *vco2*.
- Change the default parameters of tables (e.g. size) or use an user-defined waveform specified in a function table.

Syntax

```
ifn vco2init iwave [, ibasfn] [, ipmul] [, iminsiz] [, imaxsiz] [, isrcft]
```

Initialization

ifn -- the first free ftable number after the allocated tables. If *ibasfn* was not specified, -1 is returned.

iwave -- sum of the following values selecting which waveforms are to be calculated:

- 16: triangle
- 8: square wave
- 4: pulse (not normalized)
- 2: $4 * x * (1 - x)$ (integrated sawtooth)
- 1: sawtooth

Alternatively, *iwave* can be set to a negative integer that selects an user-defined waveform. This also requires the *isrcft* parameter to be specified. *vco2* can access waveform number -1. However, other user-defined waveforms are usable only with *vco2ft* or *vco2ift*.

ibasfn (optional, default=-1) -- ftable number from which the table set(s) can be accessed by opcodes other than *vco2*. This is required by user defined waveforms, with the exception of -1. If this value is less than 1, it is not possible to access the tables calculated by *vco2init* as Csound function tables.

ipmul (optional, default=1.05) -- multiplier value for number of harmonic partials. If one table has n

partials, the next one will have $n * ipmul$ (at least $n + 1$). The allowed range for *ipmul* is 1.01 to 2. Zero or negative values select the default (1.05).

iminsiz (optional, default=-1) -- minimum table size.

imaxsiz (optional, default=-1) -- maximum table size.

The actual table size is calculated by multiplying the square root of the number of harmonic partials by *iminsiz*, rounding up the result to the next power of two, and limiting this not to be greater than *imaxsiz*.

Both parameters, *iminsiz* and *imaxsiz*, must be power of two, and in the allowed range. The allowed range is 16 to 262144 for *iminsiz* to up to 16777216 for *imaxsiz*. Zero or negative values select the default settings:

- The minimum size is 128 for all waveforms except pulse (*iwave*=4). Its minimum size is 256.
- The default maximum size is usually the minimum size multiplied by 64, but not more than 16384 if possible. It is always at least the minimum size.

isrcft (optional, default=-1) -- source ftable number for user-defined waveforms (if *iwave* < 0). *isrcft* should point to a function table containing the waveform to be used for generating the table array. The table size is recommended to be at least *imaxsiz* points. If *iwave* is not negative (built-in waveforms are used), *isrcft* is ignored.



Warning

The number and size of tables is not fixed. Orchestras should not depend on these parameters, as they are subject to changes between releases.

If the selected table set already exists, it is replaced. If any opcode is accessing the tables at the same time, it is very likely that a crash will occur. This is why it is recommended to use *vco2init* only in the orchestra header.

These tables should not be replaced/overwritten by GEN routines or the *ftgen* opcode. Otherwise, unpredictable behavior or a Csound crash may occur if *vco2* is used. The first free ftable after the table array(s) is returned in *ifn*.

Examples

See the example for the *vco2* opcode.

See Also

vco2ft, *vco2ift*, and *vco2*.

Credits

Author: Istvan Varga

New in version 4.22

vcomb

vcomb — Variably reverberates an input signal with a “colored” frequency response.

Description

Variably reverberates an input signal with a “colored” frequency response.

Syntax

```
ares vcomb asig, krvt, xlpt, imaxlpt [, iskip] [, insmps]
```

Initialization

imaxlpt -- maximum loop time for *klpt*

iskip (optional, default=0) -- initial disposition of delay-loop data space (cf. *reson*). The default value is 0.

insmps (optional, default=0) -- delay amount, as a number of samples.

Performance

krvt -- the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude).

xlpt -- variable loop time in seconds, same as *ilpt* in *comb*. Loop time can be as large as *imaxlpt*.

This filter reiterates input with an echo density determined by loop time *ilpt*. The attenuation rate is independent and is determined by *krvt*, the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude). Output will appear only after *ilpt* seconds.

Examples

Here is an example of the vcomb opcode. It uses the file *vcomb.csd* [examples/vcomb.csd].

Example 535. Example of the vcomb opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          -M0 ;;RT audio I/O with MIDI in
</CsOptions>
<CsInstruments>

; Example by Jonathan Murphy and Charles Gran 2007
sr      = 44100
ksmps   = 10
nchnls  = 2

; new, and important. Make sure that midi note events are only
; received by instruments that actually need them.

; turn default midi routing off
```

```
    massign      0, 0
    ; route note events on channel 1 to instr 1
    massign      1, 1

; Define your midi controllers
#define C1 #21#
#define C2 #22#
#define C3 #23#

; Initialize MIDI controllers
    initc7       1, $C1, 0.5           ;delay send
    initc7       1, $C2, 0.5           ;delay: time to zero
    initc7       1, $C3, 0.5           ;delay: rate

    gaosc        init      0

; Define an opcode to "smooth" the MIDI controller signal
    opcode smooth, k, k
    kin          xin
    kport        linseg    0, 0.0001, 0.01, 1, 0.01
    kin          portk     kin, kport
    xout         xout      kin
    endop

instr 1
; Generate a sine wave at the frequency of the MIDI note that triggered the instrument
    ifgc         cpsmidi
    iamp         ampmidi    10000
    aenv         linenr     iamp, .01, .1, .01           ;envelope
    al          oscil      aenv, ifgc, 1
; All sound goes to the global variable gaosc
    gaosc        = gaosc + al
    endin

    instr        198 ; ECHO
    kcbsnd       ctrl17    1, $C1, 0, 1                 ;delay send
    ktime        ctrl17    1, $C2, 0.01, 6              ;time loop fades out
    kloop        ctrl17    1, $C3, 0.01, 1              ;loop speed
; Receive MIDI controller values and then smooth them
    kcbsnd       smooth    kcbsnd
    ktime        smooth    ktime
    kloop        smooth    kloop
    imaxlpt      = 1                                     ;max loop time
; Create a variable reverberation (delay) of the gaosc signal
    acomb        vcomb     gaosc, ktime, kloop, imaxlpt, 1
    aout         = (acomb * kcbsnd) + gaosc * (1 - kcbsnd)
    outs         outs      aout, aout
    gaosc        = 0
    endin

</CsInstruments>
<CsScore>
f1 0 16384 10 1
i198 0 10000
e
</CsScore>
</CsoundSynthesizer>
```

See Also

alpass, comb, reverb, valpass

Credits

Author: William “Pete” Moss
University of Texas at Austin
Austin, Texas USA
January 2002

vcopy

vcopy — Copies between two vectorial control signals

Description

Copies between two vectorial control signals

Syntax

```
vcopy ifn, ifn2, kelements [, kdstoffset] [, ksrcoffset] [, kverbose]
```

Initialization

ifn1 - number of the table where the vectorial signal will be copied (destination)

ifn2 - number of the table hosting the vectorial signal to be copied (source)

Performance

kelements - number of elements of the vector

kdstoffset - index offset for the destination (*ifn1*) table (Default=0)

ksrcoffset - index offset for the source (*ifn2*) table (Default=0)

kverbose - Selects whether or not warnings are printed (Default=0)

vcopy copies *kelements* elements from *ifn2* (starting from position *ksrcoffset*) to *ifn1* (starting from position *kdstoffset*). Useful to keep old vector values, by storing them in another table.

Negative values for *kdstoffset* and *ksrcoffset* are acceptable. If *kdstoffset* is negative, the out of range section of the vector will be discarded. If *kdstoffset* is negative, the out of range elements will be assumed to be 1 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 1 (i.e. the destination vector will not be changed for these elements).

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at k-rate (this means that every k-pass the vectors are copied). There's an i-rate version of this opcode called *vcopy_i*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
```

```
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

All these operators (*vaddv*, *vsubv*, *vmultv*, *vdivv*, *vpowv*, *vexp*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Note: *bmscan* not yet available on Canonical Csound

Examples

Here is an example of the *vcopy* opcode. It uses the file *vcopy.csd* [examples/vcopy.csd].

Example 536. Example of the *vcopy* opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o vcopy.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
kr=4410
ksmps=10
nchnls=2

instr 1 ;table playback
ar lposcil 1, 1, 0, 262144, 1
outs ar,ar
endin

instr 2
vcopy 2, 1, 20000 ;copy vector from sample to empty table
vmult 5, 20000, 262144 ;scale noise to make it audible
vcopy 1, 5, 20000 ;put noise into sample
turnoff
endin

instr 3
vcopy 1, 2, 20000 ;put original information back in
turnoff
endin

</CsInstruments>
<CsScore>
f1 0 262144 -1 "beats.wav" 0 4 0
f2 0 262144 2 0

f5 0 262144 21 3 30000

i1 0 4
i2 3 1

s
i1 0 4
i3 3 1
s

i1 0 4

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vcopy_i

vcopy_i — Copies a vector from one table to another.

Description

Copies a vector from one table to another.

Syntax

```
vcopy_i ifn, ifn2, ielements [,idstoffset, isrcoffset]
```

Initialization

ifn - number of the table where the vectorial signal will be copied

ifn - number of the table hosting the vectorial signal to be copied

ielements - number of elements of the vector

idstoffset - index offset for destination table

isrcoffset - index offset for source table

Performance

vcopy copies *ielements* elements from *ifn2* (starting from position *isrcoffset*) to *ifn1* (starting from position *idstoffset*). Useful to keep old vector values, by storing them in another table. This opcode is exactly the same as *vcopy* but performs all the copying on the initialization pass only.

Negative values for *idstoffset* and *isrcoffset* are acceptable. If *idstoffset* is negative, the out of range section of the vector will be discarded. If *isrcoffset* is negative, the out of range elements will be assumed to be 0 (i.e. the destination elements will be set to 0). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 0 (i.e. the destination vector elements will be 0).



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

All these operators (*vaddv*, *vsubv*, *vmultv*, *vdivv*, *vpowv*, *vexp*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Note: *bmscan* not yet available on Canonical Csound

Examples

See *vcopy* for an example.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vdelay

vdelay — An interpolating variable time delay.

Description

This is an interpolating variable time delay, it is not very different from the existing implementation (*deltapi*), it is only easier to use.

Syntax

```
ares vdelay asig, adel, imaxdel [, iskip]
```

Initialization

imaxdel -- Maximum value of delay in milliseconds. If *adel* gains a value greater than *imaxdel* it is folded around *imaxdel*. This should not happen.

iskip -- Skip initialization if present and non-zero

Performance

With this unit generator it is possible to do Doppler effects or chorusing and flanging.

asig -- Input signal.

adel -- Current value of delay in milliseconds. Note that linear functions have no pitch change effects. Fast changing values of *adel* will cause discontinuities in the waveform resulting noise.

Examples

```
f1 0 8192 10 1
ims      =      100          ; Maximum delay time in msec
a1      oscil      10000, 1737, 1 ; Make a signal
a2      oscil      ims/2, 1/p3, 1 ; Make an LFO
a2      =      a2 + ims/2      ; Offset the LFO so that it is positive
a3      vdelay     a1, a2, ims    ; Use the LFO to control delay time
out      out      a3
```

Two important points here. First, the delay time must be always positive. And second, even though the delay time can be controlled in k-rate, it is not advised to do so, since sudden time changes will create clicks.

See Also

vdelay3

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

vdelay3

vdelay3 — An variable time delay with cubic interpolation.

Description

vdelay3 is experimental. It is the same as *vdelay* except that it uses cubic interpolation. (New in Version 3.50.)

Syntax

```
ares vdelay3 asig, adel, imaxdel [, iskip]
```

Initialization

imaxdel -- Maximum value of delay in milliseconds. If *adel* gains a value greater than *imaxdel* it is folded around *imaxdel*. This should not happen.

iskip (optional) -- Skip initialization if present and non-zero.

Performance

With this unit generator it is possible to do Doppler effects or chorusing and flanging.

asig -- Input signal.

adel -- Current value of delay in milliseconds. Note that linear functions have no pitch change effects. Fast changing values of *adel* will cause discontinuities in the waveform resulting noise.

Examples

```
f1 0 8192 10 1
ims      =      100          ; Maximum delay time in msec
a1      oscil      10000, 1737, 1 ; Make a signal
a2      oscil      ims/2, 1/p3, 1 ; Make an LFO
a2      =      a2 + ims/2      ; Offset the LFO so that it is positive
a3      vdelay     a1, a2, ims    ; Use the LFO to control delay time
out      a3
```

Two important points here. First, the delay time must be always positive. And second, even though the delay time can be controlled in k-rate, it is not advised to do so, since sudden time changes will create clicks.

See Also

vdelay

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

vdelayx

vdelayx — A variable delay opcode with high quality interpolation.

Description

A variable delay opcode with high quality interpolation.

Syntax

```
aout vdelayx ain, adl, imd, iws [, ist]
```

Initialization

aout -- output audio signal

ain -- input audio signal

adl -- delay time in seconds

imd -- max. delay time (seconds)

iws -- interpolation window size (see below)

ist (optional) -- skip initialization if not zero

Performance

This opcode uses high quality (and slow) interpolation, that is much more accurate than the currently available linear and cubic interpolation. The *iws* parameter sets the number of input samples used for calculating one output sample (allowed values are any integer multiply of 4 in the range 4 - 1024); higher values mean better quality and slower speed.



Notes

- Delay time is measured in seconds (unlike in *vdelay* and *vdelay3*), and must be a-rate.
- The minimum allowed delay is $iws/2$ samples.
- Using the same variables as input and output is allowed in these opcodes.
- In *vdelayxw**, changing the delay time has some effects on output volume:
$$a = 1 / (1 + dt)$$
where *a* is the output gain, and *dt* is the change of delay time per seconds.
- These opcodes are best used in the double-precision version of Csound.

See Also

vdelayxq, *vdelayxs*, *vdelayxw*, *vdelayxwq*, *vdelayxws*

vdelayxq

vdelayxq — A 4-channel variable delay opcode with high quality interpolation.

Description

A 4-channel variable delay opcode with high quality interpolation.

Syntax

```
aout1, aout2, aout3, aout4 vdelayxq ain1, ain2, ain3, ain4, adl, imd, iws [, ist]
```

Initialization

aout1, aout2, aout3, aout4 -- output audio signals.

ain1, ain2, ain3, ain4 -- input audio signals.

adl -- delay time in seconds

imd -- max. delay time (seconds)

iws -- interpolation window size (see below)

ist (optional) -- skip initialization if not zero

Performance

This opcode uses high quality (and slow) interpolation, that is much more accurate than the currently available linear and cubic interpolation. The *iws* parameter sets the number of input samples used for calculating one output sample (allowed values are any integer multiply of 4 in the range 4 - 1024); higher values mean better quality and slower speed.

The multichannel opcodes (eg. *vdelayxq*) allow delaying 2 or 4 variables at once (stereo or quad signals); this is much more efficient than using separate opcodes for each channel.



Notes

- Delay time is measured in seconds (unlike in *vdelay* and *vdelay3*), and must be a-rate.
- The minimum allowed delay is $iws/2$ samples.
- Using the same variables as input and output is allowed in these opcodes.
- In *vdelayxw**, changing the delay time has some effects on output volume:
$$a = 1 / (1 + dt)$$
where *a* is the output gain, and *dt* is the change of delay time per seconds.
- These opcodes are best used in the double-precision version of Csound.

See Also

vdelayx, vdelayxs, vdelayxw, vdelayxwq, vdelayxws

vdelayxs

vdelayxs — A stereo variable delay opcode with high quality interpolation.

Description

A stereo variable delay opcode with high quality interpolation.

Syntax

```
aout1, aout2 vdelayxs ain1, ain2, adl, imd, iws [, ist]
```

Initialization

aout1, *aout2* -- output audio signals

ain1, *ain2* -- input audio signals

adl -- delay time in seconds

imd -- max. delay time (seconds)

iws -- interpolation window size (see below)

ist -- skip initialization if not zero

Performance

This opcode uses high quality (and slow) interpolation, that is much more accurate than the currently available linear and cubic interpolation. The *iws* parameter sets the number of input samples used for calculating one output sample (allowed values are any integer multiply of 4 in the range 4 - 1024); higher values mean better quality and slower speed.

The multichannel opcodes (eg. *vdelayxq*) allow delaying 2 or 4 variables at once (stereo or quad signals); this is much more efficient than using separate opcodes for each channel.



Notes

- Delay time is measured in seconds (unlike in *vdelay* and *vdelay3*), and must be a-rate.
- The minimum allowed delay is $iws/2$ samples.
- Using the same variables as input and output is allowed in these opcodes.
- In *vdelayxw**, changing the delay time has some effects on output volume:
$$a = 1 / (1 + dt)$$

where *a* is the output gain, and *dt* is the change of delay time per seconds.
- These opcodes are best used in the double-precision version of Csound.

See Also

vdelayx, vdelayxq, vdelayxw, vdelayxwq, vdelayxws

vdelayxw

vdelayxw — Variable delay opcodes with high quality interpolation.

Description

Variable delay opcodes with high quality interpolation.

Syntax

```
aout vdelayxw ain, adl, imd, iws [, ist]
```

Initialization

aout -- output audio signal

ain -- input audio signal

adl -- delay time in seconds

imd -- max. delay time (seconds)

iws -- interpolation window size (see below)

ist -- skip initialization if not zero

Performance

These opcodes use high quality (and slow) interpolation, that is much more accurate than the currently available linear and cubic interpolation. The *iws* parameter sets the number of input samples used for calculating one output sample (allowed values are any integer multiply of 4 in the range 4 - 1024); higher values mean better quality and slower speed.

The *vdelayxw* opcodes change the position of the write tap in the delay line (unlike all other delay ugens that move the read tap), and are most useful for implementing Doppler effects where the position of the listener is fixed, and the sound source is moving.



Notes

- Delay time is measured in seconds (unlike in *vdelay* and *vdelay3*), and must be a-rate.
- The minimum allowed delay is $iws/2$ samples.
- Using the same variables as input and output is allowed in these opcodes.
- In *vdelayxw**, changing the delay time has some effects on output volume:
$$a = 1 / (1 + dt)$$
where *a* is the output gain, and *dt* is the change of delay time per seconds.
- These opcodes are best used in the double-precision version of Csound.

See Also

vdelayx, vdelayxq, vdelayxs, vdelayxwq, vdelayxws

vdelayxwq

vdelayxwq — Variable delay opcodes with high quality interpolation.

Description

Variable delay opcodes with high quality interpolation.

Syntax

```
aout1, aout2, aout3, aout4 vdelayxwq ain1, ain2, ain3, ain4, adl, \  
imd, iws [, ist]
```

Initialization

ain1, ain2, ain3, ain4 -- input audio signals

aout1, aout2, aout3, aout4 -- output audio signals

adl -- delay time in seconds

imd -- max. delay time (seconds)

iws -- interpolation window size (see below)

ist -- skip initialization if not zero

Performance

These opcodes use high quality (and slow) interpolation, that is much more accurate than the currently available linear and cubic interpolation. The *iws* parameter sets the number of input samples used for calculating one output sample (allowed values are any integer multiply of 4 in the range 4 - 1024); higher values mean better quality and slower speed.

The *vdelayxw* opcodes change the position of the write tap in the delay line (unlike all other delay ugens that move the read tap), and are most useful for implementing Doppler effects where the position of the listener is fixed, and the sound source is moving.

The multichannel opcodes (eg. *vdelayxq*) allow delaying 2 or 4 variables at once (stereo or quad signals); this is much more efficient than using separate opcodes for each channel.



Notes

- Delay time is measured in seconds (unlike in *vdelay* and *vdelay3*), and must be a-rate.
- The minimum allowed delay is $iws/2$ samples.
- Using the same variables as input and output is allowed in these opcodes.
- In *vdelayxw**, changing the delay time has some effects on output volume:

$$a = 1 / (1 + dt)$$

where *a* is the output gain, and *dt* is the change of delay time per seconds.

- These opcodes are best used in the double-precision version of Csound.

See Also

vdelayx, vdelayxq, vdelayxs, vdelayxw, vdelayxws

vdelayxws

vdelayxws — Variable delay opcodes with high quality interpolation.

Description

Variable delay opcodes with high quality interpolation.

Syntax

```
aout1, aout2 vdelayxws ain1, ain2, adl, imd, iws [, ist]
```

Initialization

ain1, ain2 -- input audio signals

aout1, aout2 -- output audio signals

adl -- delay time in seconds

imd -- max. delay time (seconds)

iws -- interpolation window size (see below)

ist -- skip initialization if not zero

Performance

These opcodes use high quality (and slow) interpolation, that is much more accurate than the currently available linear and cubic interpolation. The *iws* parameter sets the number of input samples used for calculating one output sample (allowed values are any integer multiply of 4 in the range 4 - 1024); higher values mean better quality and slower speed.

The *vdelayxw* opcodes change the position of the write tap in the delay line (unlike all other delay ugens that move the read tap), and are most useful for implementing Doppler effects where the position of the listener is fixed, and the sound source is moving.

The multichannel opcodes (eg. *vdelayx*) allow delaying 2 or 4 variables at once (stereo or quad signals); this is much more efficient than using separate opcodes for each channel.



Notes

- Delay time is measured in seconds (unlike in *vdelay* and *vdelay3*), and must be a-rate.
- The minimum allowed delay is $iws/2$ samples.
- Using the same variables as input and output is allowed in these opcodes.
- In *vdelayxw**, changing the delay time has some effects on output volume:
$$a = 1 / (1 + dt)$$
where a is the output gain, and dt is the change of delay time per seconds.
- These opcodes are best used in the double-precision version of Csound.

See Also

vdelayx, vdelayxq, vdelayxs, vdelayxw, vdelayxwq

vdivv

vdivv — Performs division between two vectorial control signals

Description

Performs division between two vectorial control signals

Syntax

```
vdivv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

Performance

kelements - number of elements of the two vectors

kdstoffset - index offset for the destination (ifn1) table (Default=0)

ksrcoffset - index offset for the source (ifn2) table (Default=0)

kverbose - Selects whether or not warnings are printed (Default=0)

vdivv divides two vectorial control signals, that is, each element of *ifn1* is divided by the corresponding element of *ifn2*. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use *vcopy_i* opcode to copy it in another table. You can use *kdstoffset* and *ksrcoffset* to specify vectors in any location of the tables.

Negative values for *kdstoffset* and *ksrcoffset* are acceptable. If *kdstoffset* is negative, the out of range section of the vector will be discarded. If *ksrcoffset* is negative, the out of range elements will be assumed to be 0 (i.e. the destination elements will be set to 0). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 0 (i.e. the destination elements will be set to 0).

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at k-rate (this means that every k-pass the vectors are divided). There's an i-rate version of this opcode called *vdivv_i*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

All these operators (*vaddv*, *vsubv*, *vmultv*, *vdivv*, *vpowv*, *vexpv*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Examples

Here is an example of the *vdivv* opcode. It uses the file *vdivv.csd* [examples/vdivv.csd].

Example 537. Example of the *vdivv* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ifn2 = p5
ielements = p6
idstoffset = p7
isrcoffset = p8
kval init 25
vdivv ifn1, ifn2, ielements, idstoffset, isrcoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 15 16
f 2 0 16 -7 1 15 2
```

```
i2 0.0 0.2 1
i2 0.2 0.2 2
i1 0.4 0.01 1 2 5 3 8
i2 0.8 0.2 1
i1 1.0 0.01 1 2 5 10 -2
i2 1.2 0.2 1
i1 1.4 0.01 1 2 8 14 0
i2 1.6 0.2 1
i1 1.8 0.01 1 2 8 0 14
i2 2.0 0.2 1
i1 2.2 0.002 1 1 8 5 2
i2 2.4 0.2 1
e
```

```
</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vdivv_i

`vdivv_i` — Performs division between two vectorial control signals at init time.

Description

Performs division between two vectorial control signals at init time.

Syntax

```
vdivv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

ielements - number of elements of the two vectors

idstoffset - index offset for the destination (*ifn1*) table (Default=0)

isrcoffset - index offset for the source (*ifn2*) table (Default=0)

Performance

`vdivv_i` divides two vectorial control signals, that is, each element of *ifn1* is divided by the corresponding element of *ifn2*. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use `vcopy_i` opcode to copy it in another table. You can use *idstoffset* and *isrcoffset* to specify vectors in any location of the tables.

Negative values for *idstoffset* and *isrcoffset* are acceptable. If *idstoffset* is negative, the out of range section of the vector will be discarded. If *isrcoffset* is negative, the out of range elements will be assumed to be 1 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 1 (i.e. the destination vector will not be changed for these elements).



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at init time. There's an k-rate version of this opcode called `vdivv`.

All these operators (`vaddv_i`, `vsubv_i`, `vmultv_i`, `vdivv_i`, `vpowv_i`, `vexpv_i`, `vcopy` and `vmap`) are designed to be used together with other opcodes that operate with vectorial signals such as `bmscan`, `vcella`, `adsynt`, `adsynt2` etc.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vdelayk

vdelayk — k-rate variable time delay.

Description

Variable delay applied to a k-rate signal

Syntax

```
kout vdelayk  iksig, kdel, imaxdel [, iskip, imode]
```

Initialization

imaxdel - maximum value of delay in seconds.

iskip (optional) - Skip initialization if present and non zero.

imode (optional) - if non-zero it suppresses linear interpolation. While, normally, interpolation increases the quality of a signal, it should be suppressed if using vdelay with discrete control signals, such as, for example, trigger signals.

Performance

kout - delayed output signal

ksig - input signal

kdel - delay time in seconds can be varied at k-rate

vdelayk is similar to *vdelay*, but works at k-rate. It is designed to delay control signals, to be used, for example, in algorithmic composition.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vecdelay

vecdelay — Vectorial Control-rate Delay Paths

Description

Generate a sort of 'vectorial' delay

Syntax

```
vecdelay ifn, ifnIn, ifnDel, ielements, imaxdel [, iskip]
```

Initialization

ifn - number of the table containing the output vector

ifnIn - number of the table containing the input vector

ifnDel - number of the table containing a vector whose elements contain delay values in seconds

ielements - number of elements of the two vectors

imaxdel - Maximum value of delay in seconds.

iskip (optional) - initial disposition of delay-loop data space (see reson). The default value is 0.

Performance

vecdelay is similar to *vdelay*, but it works at k-rate and, instead of delaying a single signal, it delays a vector. *ifnIn* is the input vector of signals, *ifn* is the output vector of signals, and *ifnDel* is a vector containing delay times for each element, expressed in seconds. Elements of *ifnDel* can be updated at k-rate. Each single delay can be different from that of the other elements, and can vary at k-rate. *imaxdel* sets the maximum delay allowed for all elements of *ifnDel*.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

veloc

veloc — Get the velocity from a MIDI event.

Description

Get the velocity from a MIDI event.

Syntax

```
ival veloc [ilow] [, ihigh]
```

Initialization

ilow, ihigh -- low and hi ranges for mapping

Performance

Get the MIDI byte value (0 - 127) denoting the velocity of the current event.

Examples

Here is an example of the veloc opcode. It uses the file *veloc.csd* [examples/veloc.csd].

Example 538. Example of the veloc opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages  MIDI in
-odac        -iadc      -d         -M0   ;;RT audio I/O with MIDI in
; For Non-realtime ouput leave only the line below:
; -o veloc.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  il veloc

  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 12 seconds.
i 1 0 12
e

</CsScore>
</CsoundSynthesizer>
```

See Also

aftouch, ampmidi, cpsmidi, cpsmidib, midictrl, notnum, octmidi, octmidib, pchbend, pchmidi, pchmidib

Credits

Author: Barry L. Vercoe - Mike Berry
MIT - Mills
May 1997

Example written by Kevin Conder.

vexp

vexp — Performs power-of operations between a vector and a scalar

Description

Performs power-of operations between a vector and a scalar

Syntax

```
vexp ifn, kval, kelements [, kdstoffset] [, kverbose]
```

Initialization

ifn - number of the table hosting the vectorial signal to be processed

Performance

kval - scalar operand to be processed

kelements - number of elements of the vector

kdstoffset - index offset for the destination table (Optional, default = 0)

kverbose - Selects whether or not warnings are printed (Default=0)

vexp rises *kval* to each element contained in a vector from table *ifn*, starting from table index *kdstoffset*. This enables you to process a specific section of a table by specifying the offset and the number of elements to be processed. Offset is counted starting from 0, so if no offset is specified (or set to 0), the table will be modified from the beginning.

Note that this opcode runs at k-rate so the value of *kval* is processed every control period. Use with care or you will end up with very large (or small) numbers (or use *vexp_i*).

These opcodes (*vadd*, *vmult*, *vpow* and *vexp*) perform numeric operations between a vectorial control signal (hosted by the table *ifn*), and a scalar signal (*kval*). Result is a new vector that overrides old values of *ifn*. All these opcodes work at k-rate.

Negative values for *kdstoffset* are valid. Elements from the vector that are outside the table, will be discarded, and they will not wrap around the table.

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.

In all these opcodes, the resulting vectors are stored in *ifn*, overriding the initial vectors. If you want to keep initial vector, use *vcopy* or *vcopy_i* to copy it in another table. All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2*, etc. They can also be useful in conjunction with the spectral opcodes *pvsftw* and *pvsftr*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
```

```
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

Examples

Here is an example of the vexp opcode. It uses the file *vexp.csd* [examples/vexp.csd].

Example 539. Example of the vexp opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ival = p5
ielements = p6
idstoffset = p7
kval init 25
vexp ifn1, ival, ielements, idstoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17

i2 0.0 0.2 1
i1 0.4 0.01 1 2 3 4
i2 0.8 0.2 1
i1 1.0 0.01 1 0.5 5 -3
i2 1.2 0.2 1
i1 1.4 0.01 1 1.5 10 12
i2 1.6 0.2 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vexp_i

vexp_i — Performs power-of operations between a vector and a scalar

Description

Performs power-of operations between a vector and a scalar

Syntax

```
vexp_i ifn, ival, ielements[, idstoffset]
```

Initialization

ifn - number of the table hosting the vectorial signal to be processed

ielements - number of elements of the vector

ival - scalar value to be added

idstoffset - index offset for the destination table

Performance

vexp_i rises *kval* to each element contained in a vector from table *ifn*, starting from table index *idstoffset*. This enables you to process a specific section of a table by specifying the offset and the number of elements to be processed. Offset is counted starting from 0, so if no offset is specified (or set to 0), the table will be modified from the beginning.

Negative values for *idstoffset* are valid. Elements from the vector that are outside the table, will be discarded, and they will not wrap around the table.

This opcode runs only on initialization, there is a k-rate version of this opcode called *vexp*.

In all these opcodes, the resulting vectors are stored in *ifn*, overriding the initial vectors. If you want to keep initial vector, use *vcopy* or *vcopy_i* to copy it in another table. All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2*, etc. They can also be useful in conjunction with the spectral opcodes *pvsftw* and *pvsftr*.

Examples

Here is an example of the *vexp_i* opcode. It uses the file *vexp_i.csd* [examples/vexp_i.csd].

Example 540. Example of the vexp_i opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac        -iadc        ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
```

```
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ival = p5
ielements = p6
idstoffset = p7
kval init 25
vexp_i ifn1, ival, ielements, dstoffset
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17

i2 0.0 0.2 1
i1 0.4 0.01 1 2 3 4
i2 0.8 0.2 1
i1 1.0 0.01 1 0.5 5 -3
i2 1.2 0.2 1
i1 1.4 0.01 1 1.5 10 12
i2 1.6 0.2 1
e

</CsScore>
</CsSoundSynthesizer>
```

See also

vadd, *vmult_i*, *vpow_i* and *vexp_i*.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vexpseg

vexpseg — Vectorial envelope generator

Description

Generate exponential vectorial segments

Syntax

```
vexpseg ifnout, ielements, ifn1, idur1, ifn2 [, idur2, ifn3 [...]]
```

Initialization

ifnout - number of table hosting output vectorial signal

ifn1 - starting vector

ifn2, ifn3, etc. - vector after idurx seconds

idur1 - duration in seconds of first segment.

dur2, idur3, etc. - duration in seconds of subsequent segments.

ielements - number of elements of vectors.

Performance

These opcodes are similar to linseg and expseg, but operate with vectorial signals instead of with scalar signals.

Output is a vectorial control signal hosted by ifnout (that must be previously allocated), while each break-point of the envelope is actually a vector of values. All break-points must contain the same number of elements (ielements).

All these operators are designed to be used together with other opcodes that operate with vectorial signals such as bmscan, vcella, adsynt, adsynt2 etc.

Example

Here is an example of the vexpseg opcode. It uses the files *vexpseg.csd* [examples/vexpseg.csd].

Example 541. Example of the vexpseg opcode.

```
<CsoundSynthesizer>
<CsOptions>
-odac -B441 -b441
</CsOptions>
<CsInstruments>

sr=44100
ksmps=10
nchnls=2

gilen init 32

gitable1 ftgen 0, 0, gilen, 10, 1
gitable2 ftgen 0, 0, gilen, 10, 1
```

```
gitable3 ftgen 0, 0, gilen, -7, 30, gilen, 35
gitable4 ftgen 0, 0, gilen, -7, 400, gilen, 450
gitable5 ftgen 0, 0, gilen, -7, 5000, gilen, 5500

instr 1
vcopy gitable2, gitable1, gilen
turnoff
endin

instr 2
vexpseg gitable2, 16, gitable3, 2, gitable4, 2, gitable5
endin

instr 3
kcount init 0
if kcount < 16 then
    kval table kcount, gitable2
    printk 0,kval
    kcount = kcount +1
else
turnoff
endif

endin

</CsInstruments>
<CsScore>
i1 0 1
s
i2 0 10
i3 0 1
i3 1 1
i3 1.5 1
i3 2 1
i3 2.5 1
i3 3 1
i3 3.5 1
i3 4 1
i3 4.5 1

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado.

Example by Andres Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

vexpv

vexpv — Performs exponential operations between two vectorial control signals

Description

Performs exponential operations between two vectorial control signals

Syntax

```
vexpv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

Performance

kelements - number of elements of the two vectors

kdstoffset - index offset for the destination (*ifn1*) table (Default=0)

ksrcoffset - index offset for the source (*ifn2*) table (Default=0)

kverbose - Selects whether or not warnings are printed (Default=0)

vexpv elevates each element of *ifn2* to the corresponding element of *ifn1*. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use *vcopy_i* opcode to copy it in another table. You can use *kdstoffset* and *ksrcoffset* to specify vectors in any location of the tables.

Negative values for *kdstoffset* and *ksrcoffset* are acceptable. If *kdstoffset* is negative, the out of range section of the vector will be discarded. If *ksrcoffset* is negative, the out of range elements will be assumed to be 0 (i.e. the destination elements will be set to 1). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 0 (i.e. the destination elements will be set to 1).

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at k-rate (this means that every k-pass the vectors are processed). There's an i-rate version of this opcode called *vexpv_i*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

All these operators (*vaddv*, *vsubv*, *vmultv*, *vdivv*, *vpowv*, *vexpv*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Examples

Here is an example of the *vexpv* opcode. It uses the file *vexpv.csd* [examples/vexpv.csd].

Example 542. Example of the *vexpv* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ifn2 = p5
ielements = p6
idstoffset = p7
isrcoffset = p8
kval init 25
vexpv ifn1, ifn2, ielements, idstoffset, isrcoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17
f 2 0 16 -7 0 16 1
```

```
i2 0.0 0.2 1
i2 0.2 0.2 2
i1 0.4 0.01 1 2 5 3 8
i2 0.8 0.2 1
i1 1.0 0.01 1 2 5 10 -2
i2 1.2 0.2 1
i1 1.4 0.01 1 2 8 14 0
i2 1.6 0.2 1
i1 1.8 0.002 1 2 8 0 14
i2 2.0 0.2 1
i1 2.2 0.002 1 1 8 5 2
i2 2.4 0.2 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vexpv_i

vexpv_i — Performs exponential operations between two vectorial control signals at init time.

Description

Performs exponential operations between two vectorial control signals at init time.

Syntax

```
vexpv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

ielements - number of elements of the two vectors

idstoffset - index offset for the destination (*ifn1*) table (Default=0)

isrcoffset - index offset for the source (*ifn2*) table (Default=0)

Performance

vexpv_i elevates each element of *ifn2* to the corresponding element of *ifn1*. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use *vcopy_i* opcode to copy it in another table. You can use *idstoffset* and *isrcoffset* to specify vectors in any location of the tables.

Negative values for *idstoffset* and *isrcoffset* are acceptable. If *idstoffset* is negative, the out of range section of the vector will be discarded. If *isrcoffset* is negative, the out of range elements will be assumed to be 1 (i.e. the destination elements will be set to 1). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 1 (i.e. the destination vector elements will be set to 1).



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at init time. There's an k-rate version of this opcode called *vexpv*.

All these operators (*vaddv_i*, *vsubv_i*, *vmultv_i*, *vdivv_i*, *vpowv_i*, *vexpv_i*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vibes

vibes — Physical model related to the striking of a metal block.

Description

Audio output is a tone related to the striking of a metal block as found in a vibraphone. The method is a physical model developed from Perry Cook, but re-coded for Csound.

Syntax

```
ares vibes kamp, kfreq, ihrd, ipos, imp, kvibf, kvamp, ivibfn, idec
```

Initialization

ihrd -- the hardness of the stick used in the strike. A range of 0 to 1 is used. 0.5 is a suitable value.

ipos -- where the block is hit, in the range 0 to 1.

imp -- a table of the strike impulses. The file *marmstk1.wav* [examples/marmstk1.wav] is a suitable function from measurements and can be loaded with a *GEN01* table. It is also available at <ftp://ftp.cs.bath.ac.uk/pub/dream/documentation/sounds/modelling/>.

ivfn -- shape of vibrato, usually a sine table, created by a function

idec -- time before end of note when damping is introduced

idoubles (optional) -- percentage of double strikes. Default is 40%.

itriples (optional) -- percentage of triple strikes. Default is 20%.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kvibf -- frequency of vibrato in Hertz. Suggested range is 0 to 12

kvamp -- amplitude of the vibrato

Examples

Here is an example of the vibes opcode. It uses the file *vibes.csd* [examples/vibes.csd], and *marmstk1.wav* [examples/marmstk1.wav].

Example 543. Example of the vibes opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc     -d      ;;RT audio I/O
```

```
; For Non-realtime ouput leave only the line below:
; -o vibes.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; kamp = 20000
; kfreq = 440
; ihrd = 0.5
; ipos = 0.561
; imp = 1
; kvibf = 6.0
; kvamp = 0.05
; ivibfn = 2
; idec = 0.1

a1 vibes 20000, 440, 0.5, 0.561, 1, 6.0, 0.05, 2, 0.1

out a1
endin

</CsInstruments>
<CsScore>

; Table #1, the "marmstk1.wav" audio file.
f 1 0 256 1 "marmstk1.wav" 0 0 0
; Table #2, a sine wave for the vibrato.
f 2 0 128 10 1

; Play Instrument #1 for four seconds.
i 1 0 4
e

</CsScore>
</CsoundSynthesizer>
```

See Also

marimba

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

vibr

vibr — Easier-to-use user-controllable vibrato.

Description

Easier-to-use user-controllable vibrato.

Syntax

```
kout vibr kAverageAmp, kAverageFreq, ifn
```

Initialization

ifn -- Number of vibrato table. It normally contains a sine or a triangle wave.

Performance

kAverageAmp -- Average amplitude value of vibrato

kAverageFreq -- Average frequency value of vibrato (in cps)

vibr is an easier-to-use version of *vibrato*. It has the same generation-engine of *vibrato*, but the parameters corresponding to missing input arguments are hard-coded to default values.

Examples

Here is an example of the vibr opcode. It uses the file *vibr.csd* [examples/vibr.csd].

Example 544. Example of the vibr opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o vibr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create a vibrato waveform.
kaverageamp init 7500
kaveragefreq init 5
ifn = 1
kvamp vibr kaverageamp, kaveragefreq, ifn

; Generate a tone including the vibrato.
a1 oscili 10000+kvamp, 440, 2

out a1
```

```
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave for the vibrato.
f 1 0 256 10 1
; Table #1, a sine wave for the oscillator.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

jitter, jitter2, vibrato

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

New in Version 4.15

vibrato

vibrato — Generates a natural-sounding user-controllable vibrato.

Description

Generates a natural-sounding user-controllable vibrato.

Syntax

```
kout vibrato kAverageAmp, kAverageFreq, kRandAmountAmp, \  
          kRandAmountFreq, kAmpMinRate, kAmpMaxRate, kcpsMinRate, \  
          kcpsMaxRate, ifn [, iphs]
```

Initialization

ifn -- Number of vibrato table. It normally contains a sine or a triangle wave.

iphs -- (optional) Initial phase of table, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is 0.

Performance

kAverageAmp -- Average amplitude value of vibrato

kAverageFreq -- Average frequency value of vibrato (in cps)

kRandAmountAmp -- Amount of random amplitude deviation

kRandAmountFreq -- Amount of random frequency deviation

kAmpMinRate -- Minimum frequency of random amplitude deviation segments (in cps)

kAmpMaxRate -- Maximum frequency of random amplitude deviation segments (in cps)

kcpsMinRate -- Minimum frequency of random frequency deviation segments (in cps)

kcpsMaxRate -- Maximum frequency of random frequency deviation segments (in cps)

vibrato outputs a natural-sounding user-controllable vibrato. The concept is to randomly vary both frequency and amplitude of the oscillator generating the vibrato, in order to simulate the irregularities of a real vibrato.

In order to have a total control of these random variations, several input arguments are present. Random variations are obtained by two separated segmented lines, the first controlling amplitude deviations, the second the frequency deviations. Average duration of each segment of each line can be shortened or enlarged by the arguments *kAmpMinRate*, *kAmpMaxRate*, *kcpsMinRate*, *kcpsMaxRate*, and the deviation from the average amplitude and frequency values can be independently adjusted by means of *kRandAmountAmp* and *kRandAmountFreq*.

Examples

Here is an example of the vibrato opcode. It uses the file *vibrato.csd* [examples/vibrato.csd].

Example 545. Example of the vibrato opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o vibrato.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create a vibrato waveform.
kaverageamp init 2500
kaveragefreq init 6
krandamountamp init 0.3
krandamountfreq init 0.5
kampminrate init 3
kampmaxrate init 5
kcpsminrate init 3
kcpsmaxrate init 5
ifn = 1
kvamp vibrato kaverageamp, kaveragefreq, krandamountamp, \
             krandamountfreq, kampminrate, kampmaxrate, \
             kcpsminrate, kcpsmaxrate, ifn

; Generate a tone including the vibrato.
a1 oscili 10000+kvamp, 440, 2

out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave for the vibrato.
f 1 0 256 10 1
; Table #1, a sine wave for the oscillator.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

jitter, jitter2, vibr

Credits

Author: Gabriel Maldonado

Example written by Kevin Conder.

New in Version 4.15

vincr

vincr — Accumulates audio signals.

Description

vincr increments one audio variable with another signal, i.e. it accumulates output.

Syntax

```
vincr accum, aincr
```

Performance

accum -- audio-rate accumulator variable to be incremented

aincr -- incrementing signal

vincr (variable increment) and *clear* are intended to be used together. *vincr* stores the result of the sum of two audio variables into the first variable itself (which is intended to be used as an accumulator in polyphony). The accumulator is typically a global variable that is used to combine signals from several sources (different instruments or instrument instances) for further processing (for example, via a global effect that reads the accumulator) or for outputting the combined signal by some means other than one of the *out* opcodes (eg. via the *fout* opcode). After the accumulator is used, the accumulator variable should be set to zero by means of the *clear* opcode (or it will explode).

Examples

See the *fout* opcode for an example.

See Also

clear

Credits

Author: Gabriel Maldonado
Italy
1999

New in Csound version 3.56

vlimit

vlimit — Limiting and Wrapping Vectorial Signals

Description

Limits elements of vectorial control signals.

Syntax

```
vlimit ifn, kmin, kmax, ielements
```

Initialization

ifn - number of the table hosting the vector to be processed

ielements - number of elements of the vector

Performance

kmin - minimum threshold value

kmax - maximum threshold value

vlimit set lower and upper limits on each element of the vector they process.

These opcodes are similar to *limit*, *wrap* and *mirror*, but operate with a vectorial signal instead of with a scalar signal.

Result overrides old values of *ifn1*, if these are out of min/max interval. If you want to keep input vector, use *vcopy* opcode to copy it in another table.

All these opcodes are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Note: *bmscan* not yet available on Canonical Csound

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vlinseg

vlinseg — Vectorial envelope generator

Description

Generate linear vectorial segments

Syntax

```
vlinseg ifnout, ielements, ifn1, idur1, ifn2 [, idur2, ifn3 [...]]
```

Initialization

ifnout - number of table hosting output vectorial signal

ifn1 - starting vector

ifn2, ifn3, etc. - vector after idurx seconds

idur1 - duration in seconds of first segment.

dur2, idur3, etc. - duration in seconds of subsequent segments.

ielements - number of elements of vectors.

Performance

These opcodes are similar to linseg and expseg, but operate with vectorial signals instead of with scalar signals.

Output is a vectorial control signal hosted by ifnout (that must be previously allocated), while each break-point of the envelope is actually a vector of values. All break-points must contain the same number of elements (ielements).

All these operators are designed to be used together with other opcodes that operate with vectorial signals such as bmscan, vcella, adsynt, adsynt2 etc.

Example

Here is an example of the vlinseg opcode. It uses the files *vlinseg.csd* [examples/vlinseg.csd].

Example 546. Example of the vlinseg opcode.

```
<CsoundSynthesizer>
<CsOptions>
-odac -B441 -b441
</CsOptions>
<CsInstruments>

sr=44100
ksmps=10
nchnls=2

gilen init 32

gitable1 ftgen 0, 0, gilen, 10, 1
gitable2 ftgen 0, 0, gilen, 10, 1
```

```
gitable3 ftgen 0, 0, gilen, -7, 30, gilen, 35
gitable4 ftgen 0, 0, gilen, -7, 400, gilen, 450
gitable5 ftgen 0, 0, gilen, -7, 5000, gilen, 5500

instr 1
vcopy gitable2, gitable1, gilen
turnoff
endin

instr 2
vlinseg gitable2, 16, gitable3, 2, gitable4, 2, gitable5
endin

instr 3
kcount init 0
if kcount < 16 then
    kval table kcount, gitable2
    printk 0,kval
    kcount = kcount +1
else
turnoff
endif

endin

</CsInstruments>
<CsScore>
i1 0 1
s
i2 0 10
i3 0 1
i3 1 1
i3 1.5 1
i3 2 1
i3 2.5 1
i3 3 1
i3 3.5 1
i3 4 1
i3 4.5 1

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado.

Example by Andres Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

vlowres

vlowres — A bank of filters in which the cutoff frequency can be separated under user control.

Description

A bank of filters in which the cutoff frequency can be separated under user control

Syntax

ares **vlowres** asig, kfco, kres, iord, ksep

Initialization

iord -- total number of filters (1 to 10)

Performance

asig -- input signal

kfco -- frequency cutoff (not in Hz)

ksep -- frequency cutoff separation for each filter

vlowres (variable resonant lowpass filter) allows a variable response curve in resonant filters. It can be thought of as a bank of lowpass resonant filters, each with the same resonance, serially connected. The frequency cutoff of each filter can vary with the *kfco* and *ksep* parameters.

Examples

Here is an example of the vlowres opcode. It uses the file *vlowres.csd* [examples/vlowres.csd], and *beats.wav* [examples/beats.wav].

Example 547. Example of the vlowres opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o vlowres.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Use a nice sawtooth waveform.
asig vco 32000, 220, 1

; Vary the cutoff frequency from 30 to 300 Hz.
kfco line 30, p3, 300
```

```
kres = 25
iord = 2
ksep = 20

; Apply the filters.
avlr vlowres asig, kfco, kres, iord, ksep

; It gets loud, so clip the output amplitude to 30,000.
al clip avlr, 1, 30000
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Gabriel Maldonado
Italy

Example written by Kevin Conder.

New in Csound version 3.49

vmap

vmap — Maps elements from a vector according to indeces contained in another vector

Description

Maps elements from a vector onto another according to the indeces of a this vector

Syntax

```
vmap ifn1, ifn2, ielements [,idstoffset, isrcoffset]
```

Initialization

ifn1 - number of the table where the vectorial signal will be copied, and which contains the mapping vector

ifn2 - number of the table hosting the vectorial signal to be copied

ielements - number of elements to process

idstoffset - index offset for destination table (*ifn1*)

isrcoffset - index offset for source table (*ifn2*)

Performance

vmap maps elements of *ifn2* according to the values of table *ifn1*. Elements of *ifn1* are treated as indexes of table *ifn2*, so element values of *ifn1* must not exceed the length of *ifn2* table otherwise a Csound will report an error. Elements of *ifn1* are treated as integers, so any fractional part will be truncated. There is no interpolation performed on this operation.

In practice, what happens is that the elements of *ifn1* are used as indeces to *ifn2*, and then are replaced by the corresponding elements from *ifn2*. *ifn1* must be different from *ifn2*, otherwise the results are unpredictable. Csound will produce an init error if they are not.

All these operators (*vaddv*, *vsubv*, *vmultv*, *vdivv*, *vpowv*, *vexpv*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Note: *bmscan* not yet available on Canonical Csound

Examples

Here is an example of the *vmap* opcode. It uses the file *vmap.csd* [examples/vmap.csd].

Example 548. Example of the *vmap* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;;RT audio I/O
```

```
; For Non-realtime output leave only the line below:
; -o vmap.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
ksmps = 256
nchnls = 2
gisize = 64

gitable ftgen 0, 0, gisize, 10, 1 ;Table to be processed
gimap1 ftgen 0, 0, gisize, -7, gisize-1, gisize-1, 0 ; Mapping function to reverse table
gimap2 ftgen 0, 0, gisize, -5, 1, gisize-1, gisize-1 ; Mapping function for PWM
gimap3 ftgen 0, 0, gisize, -7, 1, (gisize/2)-1, gisize-1, 1, 1, (gisize/2)-1, gisize-1 ; Double fr

instr 1 ;Hear an oscillator using gitable
asig oscil 10000, 440, gitable
outs asig,asig
endin

instr 2 ;Reverse the table (no sound change, except for a single click
vmap gimap1, gitable, gisize
vcopy_i gitable, gimap1, gisize
turnoff
endin

instr 3 ;Non-interpolated PWM (or phase waveshaping)
vmap gimap2, gitable, gisize
vcopy_i gitable, gimap2, gisize
turnoff
endin

instr 4 ;Double frequency
vmap gimap3, gitable, gisize
vcopy_i gitable, gimap3, gisize
turnoff
endin

</CsInstruments>
<CsScore>
i 1 0 8

i 2 2 1
i 3 4 1
i 4 6 1

e
</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vmirror

vmirror — Limiting and Wrapping Vectorial Signals

Description

'Reflects' elements of vectorial control signals on thresholds.

Syntax

```
vmirror ifn, kmin, kmax, ielements
```

Initialization

ifn - number of the table hosting the vector to be processed

ielements - number of elements of the vector

Performance

kmin - minimum threshold value

kmax - maximum threshold value

vmirror 'reflects' each element of corresponding vector if it exceeds low or high thresholds.

These opcodes are similar to *limit*, *wrap* and *mirror*, but operate with a vectorial signal instead of with a scalar signal.

Result overrides old values of *ifn1*, if these are out of min/max interval. If you want to keep input vector, use *vcopy* opcode to copy it in another table.

All these opcodes are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Note: *bmscan* not yet available on Canonical Csound

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vmult

vmult — Multiplies a vector in a table by a scalar value.

Description

Multiplies a vector in a table by a scalar value.

Syntax

```
vmult ifn, kval, kelements [, kdstoffset] [, kverbose]
```

Initialization

ifn - number of the table hosting the vectorial signal to be processed

Performance

kval - scalar value to be multiplied

kelements - number of elements of the vector

kdstoffset - index offset for the destination table (Optional, default = 0)

kverbose - Selects whether or not warnings are printed (Default=0)

vmult multiplies each element of the vector contained in the table *ifn* by *kval*, starting from table index *kdstoffset*. This enables you to process a specific section of a table by specifying the offset and the number of elements to be processed. Offset is counted starting from 0, so if no offset is specified (or set to 0), the table will be modified from the beginning.

Note that this opcode runs at k-rate so the value of *kval* is multiplied every control period. Use with care or you will end up with very large numbers (or use *vmult_i*).

These opcodes (*vadd*, *vmult*, *vpow* and *vexp*) perform numeric operations between a vectorial control signal (hosted by the table *ifn*), and a scalar signal (*kval*). Result is a new vector that overrides old values of *ifn*. All these opcodes work at k-rate.

Negative values for *kdstoffset* are valid. Elements from the vector that are outside the table, will be discarded, and they will not wrap around the table.

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.

In all these opcodes, the resulting vectors are stored in *ifn*, overriding the initial vectors. If you want to keep initial vector, use *vcopy* or *vcopy_i* to copy it in another table. All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2*, etc. They can also be useful in conjunction with the spectral opcodes *pvsftw* and *pvsftr*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
```

```
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

See also

vadd_i, *vadd*, *vmult_i*, *vpow* and *vexp*.

Example

Here is an example of the *vmult* opcode. It uses the file *vmult-2.csd* [examples/vmult-2.csd].

Example 549. Example of the *vmult* opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ival = p5
ielements = p6
idstoffset = p7
kval init 25
vmult ifn1, ival, ielements, idstoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
  turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17

i2 0.0 0.2 1
i1 0.4 0.01 1 2 3 4
i2 0.8 0.2 1
i1 1.0 0.01 1 0.5 5 -3
i2 1.2 0.2 1
i1 1.4 0.01 1 1.5 10 12
i2 1.6 0.2 1
e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the vmult opcode. It uses the file *vmult.csd* [examples/vmult.csd].

Example 550. Example of the vmult opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
kr=4410
ksmps=10
nchnls=2

        instr 1 ;table playback
ar lposcil 1, 1, 0, 262144, 1
outs ar,ar
        endin

        instr 2
vcopy 2, 1, 40000 ;copy vector from sample to empty table
vmult 5, 10000, 262144 ;scale noise to make it audible
vcopy 1, 5, 40000 ;put noise into sample
turnoff
        endin

        instr 3
vcopy 1, 2, 40000 ;put original information back in
turnoff
        endin

</CsInstruments>
<CsScore>
f1 0 262144 -1 "beats.aiff" 0 4 0
f2 0 262144 2 0

f5 0 262144 21 3 30000

i1 0 4
i2 3 1

s
i1 0 4
i3 3 1
s

i1 0 4

</CsScore>
</CsoundSynthesizer>
```

See also

vadd_i, *vmult*, *vpow* and *vexp*.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

Example by Andres Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

vmult_i

vmult_i — Multiplies a vector in a table by a scalar value.

Description

Multiplies a vector in a table by a scalar value.

Syntax

```
vmult_i ifn, ival, ielements [, idstoffset]
```

Initialization

ifn - number of the table hosting the vectorial signal to be processed

ival - scalar value to be multiplied

ielements - number of elements of the vector

idstoffset - index offset for the destination table

Performance

vmult_i multiplies each element of the vector contained in the table *ifn* by *ival*, starting from table index *idstoffset*. This enables you to process a specific section of a table by specifying the offset and the number of elements to be processed. Offset is counted starting from 0, so if no offset is specified (or set to 0), the table will be modified from the beginning.

This opcode runs only on initialization, there is a k-rate version of this opcode called *vmult*.

Negative values for *idstoffset* are valid. Elements from the vector that are outside the table, will be discarded, and they will not wrap around the table.

In all these opcodes, the resulting vectors are stored in *ifn*, overriding the initial vectors. If you want to keep initial vector, use *vcopy* or *vcopy_i* to copy it in another table. All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2*, etc. They can also be useful in conjunction with the spectral opcodes *pvsftw* and *pvsftr*.

Examples

Here is an example of the *vmult_i* opcode. It uses the file *vmult_i.csd* [examples/vmult_i.csd].

Example 551. Example of the vmult_i opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac        -iadc        ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
```

```
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

    instr 1
ifn1 = p4
ival = p5
ielements = p6
idstoffset = p7
kval init 25
vmult_i ifn1, ival, ielements, idstoffset
    endin

    instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
    turnoff
endif

kcount = kcount + 1
    endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17

i2 0.0 0.2 1
i1 0.4 0.01 1 2 3 4
i2 0.8 0.2 1
i1 1.0 0.01 1 0.5 5 -3
i2 1.2 0.2 1
i1 1.4 0.01 1 1.5 10 12
i2 1.6 0.2 1
e

</CsScore>
</CsoundSynthesizer>
```

See also

vadd, *vadd*, *vmult*, *vpow* and *vexp*.

See also

vadd_i, *vmult*, *vpow_i* and *vexp_i*.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

Example by Andres Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

vmultv

vmultv — Performs multiplication between two vectorial control signals

Description

Performs multiplication between two vectorial control signals

Syntax

```
vmultv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

Performance

kelements - number of elements of the two vectors

kdstoffset - index offset for the destination (ifn1) table (Default=0)

ksrcoffset - index offset for the source (ifn2) table (Default=0)

kverbose - Selects whether or not warnings are printed (Default=0)

vmultv multiplies two vectorial control signals, that is, each element of the first vector is processed (only) with the corresponding element of the other vector. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The Result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use *vcopy_i* opcode to copy it in another table. You can use *kdstoffset* and *ksrcoffset* to specify vectors in any location of the tables.

Negative values for *kdstoffset* and *ksrcoffset* are acceptable. If *kdstoffset* is negative, the out of range section of the vector will be discarded. If *ksrcoffset* is negative, the out of range elements will be assumed to be 1 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 1 (i.e. the destination vector will not be changed for these elements).

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at k-rate (this means that every k-pass the vectors are multiplied). There's an i-rate version of this opcode called *vmultv_i*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

All these operators (*vaddv*, *vsubv*, *vmultv*, *vdivv*, *vpowv*, *vexpv*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Examples

Here is an example of the *vmultv* opcode. It uses the file *vmultv.csd* [examples/vmultv.csd].

Example 552. Example of the *vmultv* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ifn2 = p5
ielements = p6
idstoffset = p7
isrcoffset = p8
kval init 25
vmultv ifn1, ifn2, ielements, idstoffset, isrcoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17
f 2 0 16 -7 1 16 2
```

```
i2 0.0 0.2 1
i2 0.2 0.2 2
i1 0.4 0.01 1 2 5 3 8
i2 0.8 0.2 1
i1 1.0 0.01 1 2 5 10 -2
i2 1.2 0.2 1
i1 1.4 0.01 1 2 8 14 0
i2 1.6 0.2 1
i1 1.8 0.01 1 2 8 0 14
i2 2.0 0.2 1
i1 2.2 0.002 1 1 8 5 2
i2 2.4 0.2 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vmultv_i

vmultv_i — Performs multiplication between two vectorial control signals at init time.

Description

Performs multiplication between two vectorial control signals at init time.

Syntax

```
vmultv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

ielements - number of elements of the two vectors

idstoffset - index offset for the destination (*ifn1*) table (Default=0)

isrcoffset - index offset for the source (*ifn2*) table (Default=0)

Performance

vmultv_i multiplies two vectorial control signals, that is, each element of the first vector is processed (only) with the corresponding element of the other vector. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use *vcopy_i* opcode to copy it in another table. You can use *idstoffset* and *isrcoffset* to specify vectors in any location of the tables.

Negative values for *idstoffset* and *isrcoffset* are acceptable. If *idstoffset* is negative, the out of range section of the vector will be discarded. If *isrcoffset* is negative, the out of range elements will be assumed to be 1 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 1 (i.e. the destination vector will not be changed for these elements).



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at init time. There's an k-rate version of this opcode called *vmultv*.

All these operators (*vaddv_i*, *vsubv_i*, *vmultv_i*, *vdivv_i*, *vpowv_i*, *vexpv_i*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

voice

voice — An emulation of a human voice.

Description

An emulation of a human voice.

Syntax

ares **voice** kamp, kfreq, kphoneme, kform, kvibf, kvamp, ifn, ivfn

Initialization

ifn, *ivfn* -- two table numbers containing the carrier waveform and the vibrato waveform. The files *impuls20.aiff* [examples/impuls20.aiff], *ahh.aiff* [examples/ahh.aiff], *eee.aiff* [examples/eee.aiff], or *ooo.aiff* [examples/ooo.aiff] are suitable for the first of these, and a sine wave for the second. These files are available from <ftp://ftp.cs.bath.ac.uk/pub/dream/documentation/sounds/modelling/>.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played. It can be varied in performance.

kphoneme -- an integer in the range 0 to 16, which select the formants for the sounds:

- “eee”, “ihh”, “ehh”, “aaa”,
- “ahh”, “aww”, “ohh”, “uhh”,
- “uuu”, “ooo”, “rrr”, “lll”,
- “mmm”, “nnn”, “nng”, “ngg”.

At present the phonemes

- “fff”, “sss”, “thh”, “shh”,
- “xxx”, “hee”, “hoo”, “hah”,
- “bbb”, “ddd”, “jjj”, “ggg”,
- “vvv”, “zzz”, “thz”, “zhh”

are not available (!)

kform -- Gain on the phoneme. values 0.0 to 1.2 recommended.

kvibf -- frequency of vibrato in Hertz. Suggested range is 0 to 12

kvamp -- amplitude of the vibrato

Examples

Here is an example of the voice opcode. It uses the file *voice.csd* [examples/voice.csd], and *impuls20.aiff* [examples/impuls20.aiff].

Example 553. Example of the voice opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o voice.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 22050
kr = 2205
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 3
  kfreq = 0.8
  kphoneme = 6
  kform = 0.488
  kvibf = 0.04
  kvamp = 1
  ifn = 1
  ivfn = 2

  av voice kamp, kfreq, kphoneme, kform, kvibf, kvamp, ifn, ivfn

  ; It tends to get loud, so clip voice's amplitude at 30,000.
  al clip av, 2, 30000
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, an audio file for the carrier waveform.
f 1 0 256 1 "impuls20.aiff" 0 0 0
; Table #2, a sine wave for the vibrato waveform.
f 2 0 256 10 1

; Play Instrument #1 for a half-second.
i 1 0 0.5
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

Example written by Kevin Conder.

New in Csound version 3.47

vosim

vosim — Simple vocal simulation based on glottal pulses with formant characteristics.

Description

This opcode produces a simple vocal simulation based on glottal pulses with formant characteristics. Output is a series of sound events, where each event is composed of a burst of squared sine pulses followed by silence. The VOSIM (VOcal SIMulation) synthesis method was developed by Kaegi and Tempelaars in the 1970's.

Syntax

```
ar vosim kamp, kFund, kForm, kDecay, kPulseCount, kPulseFactor, ifn [, iskip]
```

Intialization

ifn - a sound table, normally containing half a period of a sinewave, squared (see notes below).

iskip - (optional) Skip initialization, for tied notes.

Performance

ar - output signal. Note that the output is usually unipolar - positive only.

kamp - output amplitude, the peak amplitude of the first pulse in each burst.

kFund - fundamental pitch, in Herz. Each event is 1/kFund seconds long.

kForm - formant center frequency. Length of each pulse in the burst is 1/kForm seconds.

kDecay - a dampening factor from pulse to pulse. This is subtracted from amplitude on each new pulse.

kPulseCount - number of pulses in the burst part of each event.

kPulseFactor - the pulse width is multiplied by this value at each new pulse. This results in formant sweeping. If factor is < 1.0, the formant sweeps up, if > 1.0 each new pulse is longer, so the formant sweeps down. The final pitch of the formant is $kForm * \text{pow}(kPulseFactor, kPulseCount)$

The output of *vosim* is a series of sound events, where each event is composed of a burst of squared sine pulses followed by silence. The total duration of the events determines fundamental frequency. The length of each single pulse in the squared-sine bursts produce a formant frequency band. The width of the formant is determined by rate of silence to pulses (see below). The final result is also shaped by the dampening factor from pulse to pulse.

A small practical problem in using this opcode is that no GEN function will create a squared sine wave out of the box. Something like the following can be used to create the appropriate table from the score.

```
; use GEN09 to create half a sine in table 17
f 17 time size 9 0.5 1 0
; run instr 101 on table 17 for a single init-pass
i 101 0 0 17
```

It can also be done with an instrument writing to an f-table in the orchestra:

```
        ; square each point in table #p4. This should be run as init-only, just once in the perform
instr 101
    index tableng p4
    index = index - 1 ; start from last point
loop:
    ival table index, p4
    ival = ival * ival
    tableiw ival, index, p4
    index = index - 1
    if index < 0 igoto endloop
                        igoto loop
endloop:
endin
```



Parameter Limits

The count of pulses multiplied by pulse width should fit in the event length ($1/kFund$). If this is not fulfilled, the algorithm does not break, we just do not start any pulses that would outlast the event. This might introduce a silence at end of event even if none was intended. In consequence, $kForm$ should be higher than $kFund$, otherwise only silence is output.

Vosim was created to emulate voice sounds using a model of glottal pulse. Rich sounds can be created by combining several instances of *vosim* with different parameters. One drawback is that the signal is not band-limited. But as the authors point out, attenuation of high-pitch components is -60 dB at 6 times the fundamental frequency. The signal can also be changed by changing the source signal in the lookup table. The technique has historical interest, and can produce rich sound very cheaply (each sample requires only a table lookup and a single multiplication for attenuation).

As stated, formant bandwidth depends on the ratio between pulse burst and silence in an event. But this is not an independent parameter: The fundamental decides event length, and formant center defines the pulse length. It is therefore impossible to guarantee a specific burst/silence ratio, since the burst length has to be an integer multiple of pulse length. The decay of pulses can be used to smooth the transition from N to $N\pm 1$ pulses, but there will still be steps in the spectral profile of output. The example code below shows one approach to this.

All input parameters are k-rate. The input parameters are only used to set up each new event (or grain). Event amplitude is fixed for each event at initialization. In normal parameter ranges, when $kamps < 500$, the k-rate parameters are updated more often than events are created. In any case, no wide-band noise will be injected in the system due to k-rate inputs being updated less often than they are read, but some other artefacts could be created.

The opcode should behave reasonably in the face of all user inputs. Some details:

- $kFund < 0$: This is forced to positive - no point in "reversed" events.
- $kFund == 0$: This leads to "infinite" length event, ie a pulse burst followed by very long indefinite silence.
- $kForm == 0$: This leads to infinite length pulse, so no pulses are generated (i.e. silence).
- $kForm < 0$: Table is read backward. If table is symmetric, $kform$ and $-kform$ should give bit-identical outputs.
- $kPulseFactor == 0$: Second pulse onwards is zero. See (c).

f. `kPulseFactor < 0`: Pulses alternately read table forward and reversed.

With asymmetric pulse table there may be some use for negative `kForm` or negative `kPulseFactor`.

Examples

Here is an example of the `vosim` opcode. It uses the file `vosim.csd` [examples/vosim.csd].

Example 554. Example of the `vosim` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o vosim.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr          = 44100
ksmps       = 100
nchnls      = 1

#####
; By Rasmus Ekman 2008

; Square each point in table #p4. This should only be run once in the performance.
instr 10

    index tablen p4
    index = index - 1 ; start from last point
loop:
    ival table index, p4
    ival = ival * ival
    tableiw ival, index, p4
    index = index - 1
    if index < 0 igoto endloop
    igoto loop
endloop:
endin

#####

; Main vosim instrument. Sweeps from a fund1/form1 to fund2/form2,
; trying for narrowest formant bandwidth (still quite wide by the looks of it)
; p4:      amp
; p5, p6: fund beg-end
; p7, p8: form beg-end
; p9:      amp decay (ignored)
; p10:     pulse count (ignored - calc internally)
; p11:     pulse length mod
; p12:     skip (for tied events)
; p13:     don't fade out (if followed by tied note)
instr 1
    kamp    init p4
    ; freq start, end
    kfund   line p5, p3, p6
    ; formant start, end
    kform   line p7, p3, p8

    ; Try for constant ratio burst/silence, and narrowest formant bandwidth
    kPulseCount = (kform / kfund) ;init p10
    ; Attempt to smooth steps between format bandwidths,
    ; increasing decay before we are forced to a lower pulse count
    kDecay = kPulseCount/(kform % kfund) ; init p9
    if (kDecay * kPulseCount) > kamp then
        kDecay = kamp / kPulseCount
    endif
    kDecay = 0.3 * kDecay

    kPulseFactor init p11

; ar vosim kamp, kFund, kForm, kDecay, kPulseCount, kPulseFactor, ifn [, iskip]
```

```
ar1 vosim      kamp, kfund, kform, kDecay, kPulseCount, kPulseFactor, 17, p12

; scale amplitude for 16-bit files, with quick fade out
amp init 20000
if (p13 != 0) goto nofade
amp linseg 20000, p3-.02, 20000, .02, 0
nofade:
out ar1 * amp
endin

</CsInstruments>
<CsScore>

f1      0 32768 9 1 1 0 ; sine wave
f17     0 32768 9 0.5 1 0 ; half sine wave
i10 0 0 17 ; init run only, square table 17

; Vosim score

; Picking some formants from the table in Csound manual

;      p4=amp      fund      form      decay pulses pulsemod [skip] nofade
; tenor a -> e
i1 0 .5 .5 280 240 650 400 .03 5 1
i1 . . .3 . . 1080 1700 .03 5 .
i1 . . .2 . . 2650 2600 .03 5 .
i1 . . .15 . . 2900 3200 .03 5 .

; tenor a -> o
i1 0.6 .2 .5 300 210 650 400 .03 5 1 0 1
i1 . . .3 . . 1080 800 .03 5 . .
i1 . . .2 . . 2650 2600 .03 5 . .
i1 . . .15 . . 2900 2800 .03 5 . .

; tenor o -> aah
i1 .8 .3 .5 210 180 400 650 .03 5 1 1 1
i1 . . .3 . . 800 1080 .03 5 . .
i1 . . .2 . . 2600 2650 .03 5 . .
i1 . . .15 . . 2800 2900 .03 5 . .

; tenor aa -> i
i1 1.1 .2 .5 180 250 650 290 .03 5 1 1 1
i1 . . .3 . . 1080 1870 .03 5 . .
i1 . . .2 . . 2650 2800 .03 5 . .
i1 . . .15 . . 2900 3250 .03 5 . .

; tenor i -> u
i1 1.3 .3 .5 250 270 290 350 .03 5 1 1 0
i1 . . .3 . . 1870 600 .03 5 . .
i1 . . .2 . . 2800 2700 .03 5 . .
i1 . . .15 . . 3250 2900 .03 5 . .

e

</CsScore>
</CsoundSynthesizer>
```

See also

fof, fof2

Credits

Author: Rasmus Ekman
March 2008

vphaseseg

vphaseseg — Allows one-dimensional HVS (Hyper-Vectorial Synthesis).

Description

vphaseseg allows one-dimensional HVS (Hyper-Vectorial Synthesis).

Syntax

```
vphaseseg kphase, ioutab, ielems, itab1,idist1,itab2 \  
[,idist2,itab3, ... ,idistN-1,itabN]
```

Initialization

ioutab - number of output table.

ielem - number of elements to process

itab1,...,itabN - breakpoint table numbers

idist1,...,idistN-1 - distances between breakpoints in percentage values

Performance

kphase - phase pointer

vphaseseg returns the coordinates of section points of an N-dimensional space path. The coordinates of section points are stored into an output table. The number of dimensions of the N-dimensional space is determined by the *ielem* argument that is equal to N and can be set to any number. To define the path, user have to provide a set of points of the N-dimensional space, called break-points. Coordinates of each break-point must be contained by a different table. The number of coordinates to insert in each break-point table must obviously equal to *ielem* argument. There can be any number of break-point tables filled by the user.

Hyper-Vectorial Synthesis actually deals with two kinds of spaces. The first space is the N-dimensional space in which the path is defined, this space is called time-variant parameter space (or SPACE A). The path belonging to this space is covered by moving a point into the second space that normally has a number of dimensions smaller than the first. Actually, the point in motion is the projection of corresponding point of the N-dimensional space (could also be considered a section of the path). The second space is called user-pointer-motion space (or SPACE B) and, in the case of *vphaseseg* opcode, has only ONE DIMENSION. Space B is covered by means of *kphase* argument (that is a sort of path pointer), and its range is 0 to 1. The output corresponding to current pointer value is stored in *ioutab* table, whose data can be afterwards used to control any syntesis parameters.

In *vphaseseg*, each break-point is separated from the other by a distance expressed in percentage, where all the path length is equal to the sum of all distances. So distances between breakpoints can be different, differently from kinds of HVS in which space B has more than one dimension, in these cases distance between break-points MUST be THE SAME for all intervals.

See Also

hvs1, *hvs2*, *hvs3*

Credits

Author: Gabriel Maldonado

New in version 5.06

vport

vport — Vectorial Control-rate Delay Paths

Description

Generate a sort of 'vectorial' portamento

Syntax

```
vport ifn, khtime, ielements [, ifnInit]
```

Initialization

ifn - number of the table containing the output vector

ielements - number of elements of the two vectors

ifnInit (optional) - number of the table containing a vector whose elements contain initial portamento values.

Performance

vport is similar to *port*, but operates with vectorial signals, instead of with scalar signals. Each vector element is treated as an independent control signal. Input vector input and output vectors are placed in the same table and output vector overrides input vector. If you want to keep input vector, use *vcopy* opcode to copy it in another table.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vpow

vpow — Raises each element of a vector to a scalar power

Description

Raises each element of a vector to a scalar power

Syntax

```
vpow ifn, kval, kelements [, kdstoffset] [, kverbose]
```

Initialization

ifn - number of the table hosting the vectorial signal to be processed

Performance

kval - scalar value to which the elements of *ifn* will be raised

kelements - number of elements of the vector

kdstoffset - index offset for the destination table (Optional, default = 0)

kverbose - Selects whether or not warnings are printed (Default=0)

vpow raises each element of the vector contained in the table *ifn* to the power of *kval*, starting from table index *kdstoffset*. This enables you to process a specific section of a table by specifying the offset and the number of elements to be processed. Offset is counted starting from 0, so if no offset is specified (or set to 0), the table will be modified from the beginning.

Note that this opcode runs at k-rate so the value of *kval* is processed every control period. Use with care or you will end up with very large (or small) numbers (or use *vpow_i*).

These opcodes (*vadd*, *vmult*, *vpow* and *vexp*) perform numeric operations between a vectorial control signal (hosted by the table *ifn*), and a scalar signal (*kval*). Result is a new vector that overrides old values of *ifn*. All these opcodes work at k-rate.

Negative values for *kdstoffset* are valid. Elements from the vector that are outside the table, will be discarded, and they will not wrap around the table.

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.

In all these opcodes, the resulting vectors are stored in *ifn*, overriding the initial vectors. If you want to keep initial vector, use *vcopy* or *vcopy_i* to copy it in another table. All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2*, etc. They can also be useful in conjunction with the spectral opcodes *pvsftw* and *pvsftr*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
```

```
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

Examples

Here is an example of the `vpow` opcode. It uses the file `vpow.csd` [examples/vpow.csd].

Example 555. Example of the `vpow` opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ival = p5
ielements = p6
idstoffset = p7
kval init 25
vpow ifn1, ival, ielements, idstoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17

i2 0.0 0.2 1
i1 0.4 0.01 1 2 3 4
i2 0.8 0.2 1
i1 1.0 0.01 1 0.5 5 -3
i2 1.2 0.2 1
i1 1.4 0.01 1 1.5 10 12
i2 1.6 0.2 1
e

</CsScore>
</CsoundSynthesizer>
```

See also

vadd_i, *vmult*, *vpow* and *vexp*.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vpow_i

vpow_i — Raises each element of a vector to a scalar power

Description

Raises each element of a vector to a scalar power

Syntax

```
vpow_i ifn, ival, ielements [, idstoffset]
```

Initialization

ifn - number of the table hosting the vectorial signal to be processed

ielements - number of elements of the vector

ival - scalar value to which the elements of *ifn* will be raised

idstoffset - index offset for the destination table

Performance

vpow_i elevates each element of the vector contained in the table *ifn* to the power of *ival*, starting from table index *idstoffset*. This enables you to process a specific section of a table by specifying the offset and the number of elements to be processed. Offset is counted starting from 0, so if no offset is specified (or set to 0), the table will be modified from the beginning.

This opcode runs only on initialization, there is a k-rate version of this opcode called *vpow*.

Negative values for *idstoffset* are valid. Elements from the vector that are outside the table, will be discarded, and they will not wrap around the table.

In all these opcodes, the resulting vectors are stored in *ifn*, overriding the initial vectors. If you want to keep initial vector, use *vcopy* or *vcopy_i* to copy it in another table. All these operators are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2*, etc. They can also be useful in conjunction with the spectral opcodes *pvsftw* and *pvsftr*.

Examples

Here is an example of the *vpow_i* opcode. It uses the file *vpow_i.csd* [examples/vpow_i.csd].

Example 556. Example of the vpow_i opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac        -iadc        ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
```

```
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ival = p5
ielements = p6
idstoffset = p7
kval init 25
vpow_i ifn1, ival, ielements, idstoffset
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17

i2 0.0 0.2 1
i1 0.4 0.01 1 2 3 4
i2 0.8 0.2 1
i1 1.0 0.01 1 0.5 5 -3
i2 1.2 0.2 1
i1 1.4 0.01 1 1.5 10 12
i2 1.6 0.2 1
e

</CsScore>
</CsoundSynthesizer>
```

See also

vadd_i, *vmult_i*, *vpow* and *vexp_i*.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vpowv

vpowv — Performs power-of operations between two vectorial control signals

Description

Performs power-of operations between two vectorial control signals

Syntax

```
vpowv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

Performance

kelements - number of elements of the two vectors

kdstoffset - index offset for the destination (*ifn1*) table (Default=0)

ksrcoffset - index offset for the source (*ifn2*) table (Default=0)

kverbose - Selects whether or not warnings are printed (Default=0)

vpowv raises each element of *ifn1* to the corresponding element of *ifn2*. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use *vcopy_i* opcode to copy it in another table. You can use *kdstoffset* and *ksrcoffset* to specify vectors in any location of the tables.

Negative values for *kdstoffset* and *ksrcoffset* are acceptable. If *kdstoffset* is negative, the out of range section of the vector will be discarded. If *ksrcoffset* is negative, the out of range elements will be assumed to be 1 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 1 (i.e. the destination vector will not be changed for these elements).

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at k-rate (this means that every k-pass the vectors are processed). There's an i-rate version of this opcode called *vpowv_i*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-

rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

All these operators (*vaddv*, *vsubv*, *vmultv*, *vdivv*, *vpowv*, *vexpv*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Examples

Here is an example of the *vpowv* opcode. It uses the file *vpowv.csd* [examples/vpowv.csd].

Example 557. Example of the *vpowv* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ifn2 = p5
ielements = p6
idstoffset = p7
isrcoffset = p8
kval init 25
vpowv ifn1, ifn2, ielements, idstoffset, isrcoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 16 17
f 2 0 16 -7 1 16 2

i2 0.0 0.2 1
i2 0.2 0.2 2
i1 0.4 0.01 1 2 5 3 8
```

```
i2 0.8 0.2 1
i1 1.0 0.01 1 2 5 10 -2
i2 1.2 0.2 1
i1 1.4 0.01 1 2 8 14 0
i2 1.6 0.2 1
i1 1.8 0.01 1 2 8 0 14
i2 2.0 0.2 1
e
</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vpowv_i

`vpowv_i` — Performs power-of operations between two vectorial control signals at init time.

Description

Performs power-of operations between two vectorial control signals at init time.

Syntax

```
vpowv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

ielements - number of elements of the two vectors

idstoffset - index offset for the destination (*ifn1*) table

isrcoffset - index offset for the source (*ifn2*) table

Performance

`vpowv_i` raises each element of *ifn1* to the corresponding element of *ifn2*. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use `vcopy_i` opcode to copy it in another table. You can use *idstoffset* and *isrcoffset* to specify vectors in any location of the tables.

Negative values for *idstoffset* and *isrcoffset* are acceptable. If *idstoffset* is negative, the out of range section of the vector will be discarded. If *isrcoffset* is negative, the out of range elements will be assumed to be 1 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 1 (i.e. the destination vector will not be changed for these elements).



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at init time. There's an k-rate version of this opcode called `vpowv`.

All these operators (`vaddv_i`, `vsubv_i`, `vmultv_i`, `vdivv_i`, `vpowv_i`, `vexpv_i`, `vcopy` and `vmap`) are designed to be used together with other opcodes that operate with vectorial signals such as `bmscan`, `vcella`, `adsynt`, `adsynt2` etc.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vpvoc

vpvoc — Implements signal reconstruction using an fft-based phase vocoder and an extra envelope.

Description

Implements signal reconstruction using an fft-based phase vocoder and an extra envelope.

Syntax

```
ares vpvoc ktmpnt, kfmod, ifile [, ispecwp] [, ifn]
```

Initialization

ifile -- the pvoc number (n in pvoc.n) or the name in quotes of the analysis file made using pvanal. (See *pvoc*.)

ispecwp (optional, default=0) -- if non-zero, attempts to preserve the spectral envelope while its frequency content is varied by *kfmod*. The default value is zero.

ifn (optional, default=0) -- optional function table containing control information for vpvoc. If *ifn* = 0, control is derived internally from a previous *tableseg* or *tablexseg* unit. Default is 0. (New in Csound version 3.59)

Performance

ktmpnt -- The passage of time, in seconds, through the analysis file. *ktmpnt* must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the analysis file.

kfmod -- a control-rate transposition factor: a value of 1 incurs no transposition, 1.5 transposes up a perfect fifth, and .5 down an octave.

This implementation of *pvoc* was originally written by Dan Ellis. It is based in part on the system of Mark Dolson, but the pre-analysis concept is new. The spectral extraction and amplitude gating (new in Csound version 3.56) were added by Richard Karpen based on functions in SoundHack by Tom Erbe.

vpvoc is identical to *pvoc* except that it takes the result of a previous *tableseg* or *tablexseg* and uses the resulting function table (passed internally to the *vpvoc*), as an envelope over the magnitudes of the analysis data channels. Optionally, a table specified by *ifn* may be used.

The result is spectral enveloping. The function size used in the *tableseg* should be *framesize/2*, where *framesize* is the number of bins in the phase vocoder analysis file that is being used by the *vpvoc*. Each location in the table will be used to scale a single analysis bin. By using different functions for *ifn1*, *ifn2*, etc.. in the *tableseg*, the spectral envelope becomes a dynamically changing one. See also *tableseg* and *tablexseg*.

Examples

The following example, using *vpvoc*, shows the use of functions such as

```
f 1 0 256 5 .001 128 1 128 .001
f 2 0 256 5 1 128 .001 128 1
f 3 0 256 7 1 256 1
```

to scale the amplitudes of the separate analysis bins.

```
ktime  line      0, p3,3 ; time pointer, in seconds, into file
        tablexseg 1, p3*.5, 2, p3*.5, 3
apv     vpvoc     ktime,1, "pvoc.file"
```

The result would be a time-varying “spectral envelope” applied to the phase vocoder analysis data. Since this amplifies or attenuates the amount of signal at the frequencies that are paired with the amplitudes which are scaled by these functions, it has the effect of applying very accurate filters to the signal. In this example the first table would have the effect of a band-pass filter, gradually be band-rejected over half the note's duration, and then go towards no modification of the magnitudes over the second half.

See Also

pvoc

Credits

Authors: Dan Ellis and Richard Karpen
Seattle, WA USA
1997

New in version 3.44

vrandh

vrandh — Generates a vector of random numbers stored into a table, holding the values for a period of time.

Description

Generates a vector of random numbers stored into a table, holding the values for a period of time. Generates a sort of 'vectorial band-limited noise'.

Syntax

```
vrandh ifn, krange, kcps, ielements [, idstoffset] [, iseed]  
        [, isize] [, ioffset]
```

Initialization

ifn - number of the table where the vectorial signal will be generated

ielements - number of elements of the vector

idstoffset - (optional, default=0) -- index offset for the destination table

iseed (optional, default=0.5) -- seed value for the recursive pseudo-random formula. A value between 0 and +1 will produce an initial output of *kamp* * *iseed*. A negative value will cause seed re-initialization to be skipped. A value greater than 1 will seed from system time, this is the best option to generate a different random sequence for each run.

isize (optional, default=0) -- if zero, a 16 bit number is generated. If non-zero, a 31-bit random number is generated. Default is 0.

ioffset - (optional, default=0) -- a base value added to the random result.

Performance

krange - range of random elements (from -krange to krange)

kcps - rate of generated elements in cycles per seconds

This opcode is similar to *randh*, but operates on vectors instead of with scalar values.

Though the argument *isize* defaults to 0, thus using a 16-bit random number generator, using the newer 31-bit algorithm is recommended, as this will produce a random sequence with a longer period (more random numbers before the sequence starts repeating).

The output is a vector contained in *ifn* (that must be previously allocated).

All these operators are designed to be used together with other opocdes that operate with vector such as *bmscan*, *adsynt* etc.

Note: *bmscan* not yet available on Canonical Csound

Examples

Here is an example of the *vrandh* opcode. It uses the file *vrandh.csd* [examples/vrandh.csd].

Example 558. Example of the vrandh opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o vrandh.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
;Example by Andres Cabrera

sr=44100
ksmps=128
nchnls=2

gitab ftgen 0, 0, 16, -7, 0, 128, 0

instr 1
  krange init p4
  kcps init p5
  ioffset init p6

  kav1 init 0
  kav2 init 0
  kcount init 0

  ;      table  krange  kcps  ielements  idstoffset  iseed  isize ioffset
  vrandh  gitab,  krange, kcps,      3,          3,          2,  0,  ioffset

  kfreq1 table 3, gitab
  kfreq2 table 4, gitab
  kfreq3 table 5, gitab

  ;Change the frequency of three oscillators according to the random values
  aosc1 oscili 4000, kfreq1, 1
  aosc2 oscili 2000, kfreq2, 1
  aosc3 oscili 4000, kfreq3, 1

  outs aosc1+aosc2, aosc3+aosc2
endin

</CsInstruments>
<CsScore>
f 1 0 1024 10 1
;      krange  kcps      ioffset
i 1 0          5 100 1 300
i 1 5          5 300 1 400
i 1 10         5 100 2 1000
i 1 15         5 400 4 1000
i 1 20         5 1000 8 2000
i 1 25         5 250 16 300
e

</CsScore>
</CsoundSynthesizer>
```

See also

vrandi, *randh*

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vrandi

vrandi — Generate a sort of 'vectorial band-limited noise'

Description

Generate a sort of 'vectorial band-limited noise'

Syntax

```
vrandi ifn, krange, kcps, ielements [, idstoffset] [, iseed]  
        [, isize] [, ioffset]
```

Initialization

ifn - number of the table where the vectorial signal will be generated

ielements - number of elements to process

idstoffset - (optional, default=0) -- index offset for the destination table

iseed (optional, default=0.5) -- seed value for the recursive pseudo-random formula. A value between 0 and +1 will produce an initial output of *kamp* * *iseed*. A negative value will cause seed re-initialization to be skipped. A value greater than 1 will seed from system time, this is the best option to generate a different random sequence for each run.

isize (optional, default=0) -- if zero, a 16 bit number is generated. If non-zero, a 31-bit random number is generated. Default is 0.

ioffset - (optional, default=0) -- a base value added to the random result.

Performance

krange - range of random elements (from -krange to krange)

kcps - rate of generated elements in cycles per seconds

This opcode is similar to *randi*, but operates on vectors instead of with scalar values.

Though argument *isize* defaults to 0, thus using a 16-bit random number generator, using the newer 31-bit algorithm is recommended, as this will produce a random sequence with a longer period (more random numbers before the sequence starts repeating).

The output is a vector contained in *ifn* (that must be previously allocated).

All these operators are designed to be used together with other opocdes that operate with vector such as *bmscan*, *adsynt* etc.

Note: *bmscan* not yet available on Canonical Csound

Examples

Here is an example of the *vrandi* opcode. It uses the file *vrandi.csd* [examples/vrandi.csd].

Example 559. Example of the vrandi opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o vrandi.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

;Example by Andres Cabrera

gitab ftgen 0, 0, 16, -7, 0, 128, 0

instr 1
  krange init p4
  kcps init p5
  ioffset init p6
  ;      table  krange  kcps  ielements  idstoffset  iseed  isize  ioffset
  vrandi gitab, krange, kcps,      3,          3,          2,  1,  ioffset

  kfreq1 table 3, gitab
  kfreq2 table 4, gitab
  kfreq3 table 5, gitab

  ;Change the frequency of three oscillators according to the random values
  aosc1 oscili 4000, kfreq1, 1
  aosc2 oscili 2000, kfreq2, 1
  aosc3 oscili 4000, kfreq3, 1

  outs aosc1+aosc2, aosc3+aosc2
endin

</CsInstruments>
<CsScore>

f 1 0 2048 10 1

;      krange  kcps      ioffset
i 1 0          5 100 1 300
i 1 5          5 5 1 400
i 1 10         5 100 2 1000
i 1 15         5 400 4 1000
i 1 20         5 1000 8 2000
i 1 20         5 300 32 350

e

</CsScore>
</CsoundSynthesizer>
```

See also

vrandh, *randi*

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vstaudio, vstaudiog

vstaudio — VST audio output.

Syntax

```
aout1,aout2 vstaudio instance, [ain1, ain2]
```

```
aout1,aout2 vstaudiog instance, [ain1, ain2]
```

Description

vstaudio and *vstaudiog* are used for sending and receiving audio from a VST plugin.

vstaudio is used within an instrument definition that contains a *vstmidiout* or *vstnote* opcode. It outputs audio for only that one instrument. Any audio remaining in the plugin after the end of the note, for example a reverb tail, will be cut off and should be dealt with using a damping envelope.

vstaudiog (*vstaudio* global) is used in a separate instrument to process audio from any number of VST notes or MIDI events that share the same VST plugin instance (*instance*). The *vstaudiog* instrument must be numbered higher than all the instruments receiving notes or MIDI data, and the note controlling the *vstplug* instrument must have an indefinite duration, or at least a duration as long as the VST plugin is active.

Initialization

instance - the number which identifies the plugin, to be passed to other *vst4cs* opcodes.

Performance

aout1, *aout2* - the audio output received from the plugin.

ain1, *ain2* - the audio input sent to the plugin.

Examples

See *vstmidiout* and *vstparamset* for examples.

Credits

By: Andrés Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's *vst~* object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

vstbankload

vstbankload — Loads parameter banks to a VST plugin.

Syntax

```
vstbankload instance, ipath
```

Description

vstbankload is used for loading parameter banks to a VST plugin.

Initialization

instance - the number which identifies the plugin, to be passed to other vst4cs opcodes.

ipath - the full pathname of the parameter bank (. fxb file).

Examples

Example 560. Example for vstbankload

```
/* orc */
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
gihandle1 vstinit "c:/vstplugins/cheeze/cheeze machine.dll",1
instr 4
vstbankload gihandle1,"c:/vstplugins/cheeze/chengo'scheese.fxb"
vstinfo gihandle1
endin

/* sco */
i 3 0 21
i4 1 1 57 32
```

Credits

By: Andrés Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's vst~ object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

vstedit

vstedit — Opens the GUI editor widow for a VST plugin.

Syntax

```
vstedit instance
```

Description

vstedit opens the custom GUI editor widow for a VST plugin. Note that not all VST plugins have custom GUI editors.

Initialization

instance - the number which identifies the plugin, to be passed to other vst4cs opcodes.

Credits

By: Andrés Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's vst~ object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

vstinit

vstinit — Load a VST plugin into memory for use with the other vst4cs opcodes.

Syntax

```
instance vstinit ilibrarypath [,iverbose]
```

Description

vstinit is used to load a VST plugin into memory for use with the other vst4cs opcodes. Both VST effects and instruments (synthesizers) can be used.

Initialization

instance - the number which identifies the plugin, to be passed to other vst4cs opcodes.

ilibrarypath - the full path to the vst plugin shared library (dll, on Windows). Remember to use '/' instead of '\' as separator.

iverbose - show plugin information and parameters when loading.

Examples

Example 561. Loading a VST Plugin

```
/* orc */
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
gihandle1 vstinit "c:/vstplugins/cheeze/cheeze machine.dll",1
instr 1
    gihandle2 vstinit "c:/vstplugins/crazy diamonds.dll",1
endin

/* sco */
i 1 0 1
e
```

Credits

By: Andrés Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's vst~ object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

vstinfo

vstinfo — Displays the parameters and the programs of a VST plugin.

Syntax

```
vstinfo instance
```

Description

vstinfo displays the parameters and the programs of a VST plugin.

Note: The *verbose* flag in *vstinit* gives the same information as *vstinfo*. *vstinfo* is useful after loading parameter banks, or when the plugin changes parameters dynamically.

Initialization

instance - the number which identifies the plugin, to be passed to other vst4cs opcodes.

Examples

Example 562. Example for vstinfo

```
/* orc */
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
gihandle1 vstinit "c:/vstplugins/cheeze/cheeze machine.dll",1
instr 4
vstbankload gihandle1,"c:/vstplugins/cheeze/chengo'scheese.fxb"
vstinfo gihandle1
endin

/* sco */
i 3 0 21
i4 1 1 57 32
e
```

Credits

By: Andrés Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's vst~ object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

vstmidiout

vstmidiout — Sends MIDI information to a VST plugin.

Syntax

```
vstmidiout instance, kstatus, kchan, kdata1, kdata2
```

Description

vstmidiout is used for sending MIDI information to a VST plugin.

Initialization

instance - the number which identifies the plugin, to be passed to other vst4cs opcodes.

Performance

kstatus - the type of midi message to be sent. Currently noteon (144), note off (128), Control Change (176), Program change (192), Aftertouch (208) and Pitch Bend (224) are supported.

kchan - the MIDI channel transmitted on.

kdata1, *kdata2* - the MIDI data pair, which varies depending on *kstatus*. e.g. note/velocity for note on and note off, Controller number/value for control change.

Examples

Example 563. Example for vstmidiout

```
/* orc */
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
gihandle1 vstinit "c:/vstplugins/cheeze/cheeze machine.dll",1
instr 3
ain1 = 0
ab1, ab2 vstaudio gihandle1, ain1, ain1
outs ab1, ab2
endin
instr 4
vstmidiout gihandle1,144,1,p4,p5
endin

/* sco */
i 3 0 21
i4 1 1 57 32
i4 3 1 60 100
i4 5 1 62 100
i4 7 1 64 100
i4 9 1 65 100
i4 11 1 67 100
i4 13 1 69 100
i4 15 3 71 100
i4 18 3 72 100
e
```

Credits

By: Andrés Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's vst~ object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

vstnote

vstnote — Sends a MIDI note with definite duration to a VST plugin.

Syntax

```
vstnote instance, kchan, knote, kveloc, kdur
```

Description

vstnote sends a MIDI note with definite duration to a VST plugin.

Initialization

instance - the number which identifies the plugin generated by *vstinit*.

Performance

kchan - The midi channel to trasnmit the note on.

knote - The midi note number to send.

kveloc - The midi note's velocity.

kdur - The midi note's duration in seconds.

Note: Be sure the instrument containing vstnote is not finished before the duration of the note, otherwise you'll have a 'hung' note.

Examples

Example 564. Example for vstnote

```
/* orc */
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
gihandle5 vstinit "c:/vstplugins/cheeze/cheeze machine.dll",1
instr 3
ain1 = 0
gal, ga2 vstplugg gihandle5, ain1, ain1
endin
instr 4
vstnote giHandle5, 1, p4, p5, p3
endin
instr 10
outs gal, ga2
endin

/* sco */
i 3 0 21
i 10 0 21
i4 1 3 57 55
i4 3 3 60 100
i4 5 3 62 100
i4 7 3 64 100
i4 9 2 65 100
i4 11 1 67 100
i4 13 1 69 100
i4 15 3 71 100
```

i4 18 3 72 100

See Also

vstinit

Credits

By: Andrés Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's vst~ object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

vstparamset, vstparamget

vstparamset — Used for parameter communication to and from a VST plugin.

Syntax

```
vstparamset instance, kparam, kvalue
```

```
kvalue vstparamget instance, kparam
```

Description

vstparamset and *vstparamget* are used for parameter communication to and from a VST plugin.

Initialization

instance - the number which identifies the plugin, to be passed to other vst4cs opcodes.

Performance

kparam - The number of the parameter to set or get.

kvalue - the value to set, or the the value returned by the plugin.

Parameters vary according to the plugin. To find out what parameters are available, use the verbose option when loading the plugin with vstinit.

Examples

Example 565. Example of vstparamset

```
/* orc */
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
gihandle1 vstinit "c:/vstplugins/cheeze/cheeze machine.dll",1
instr 3
ain1 = 0
abl, ab2 vstaudio gihandle1, ain1, ain1
outs abl, ab2
endin
instr 4
vstmidiout gihandle1,144,1,p4,p5
kline line 0,p3,1
vstparamset gihandle1, 3, kline
endin
```

```
/* sco */
i 3 0 21
i4 1 1 57 32
i4 3 1 60 100
i4 5 1 62 100
i4 7 1 64 100
i4 9 1 65 100
i4 11 1 67 100
i4 13 1 69 100
i4 15 3 71 100
i4 18 3 72 100
e
```

Credits

By: Andrés Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's vst~ object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

vstprogset

vstprogset — Loads parameter banks to a VST plugin.

Syntax

```
vstprogset instance, kprogram
```

Description

vstprogset sets one of the programs in an `.fxb` bank.

Initialization

instance - the number which identifies the plugin, to be passed to other vst4cs opcodes.

kprogram - the number of the program to set.

Examples

Example 566. Usage of vstprogset

```
/* orc */
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2
giHandle1 vstinit "c:/vstplugins/cheeze/cheeze machine.dll",1
instr 4
vstbankload gihandle1,"c:/vstplugins/cheeze/chengo'scheese.fxb"
vstprogset gihandle1, 4
vstinfo gihandle1
endin

/* sco */
i 3 0 21
i4 1 1 57 32
e
```

Credits

By: Andrés Cabrera and Michael Gogins

Uses code from Hermann Seib's VSTHost and Thomas Grill's vst~ object.

VST is a trademark of Steinberg Media Technologies GmbH. VST Plug-In Technology by Steinberg.

vsubv

vsubv — Performs subtraction between two vectorial control signals

Description

Performs subtraction between two vectorial control signals

Syntax

```
vsubv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

Performance

kelements - number of elements of the two vectors

kdstoffset - index offset for the destination (ifn1) table (Default=0)

ksrcoffset - index offset for the source (ifn2) table (Default=0)

kverbose - Selects whether or not warnings are printed (Default=0)

vsubv subtracts two vectorial control signals, that is, each element of *ifn2* is subtracted from the corresponding element of *ifn1*. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use *vcopy_i* opcode to copy it in another table. You can use *kdstoffset* and *ksrcoffset* to specify vectors in any location of the tables.

Negative values for *kdstoffset* and *ksrcoffset* are acceptable. If *kdstoffset* is negative, the out of range section of the vector will be discarded. If *ksrcoffset* is negative, the out of range elements will be assumed to be 0 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 0 (i.e. the destination vector will not be changed for these elements).

If the optional *kverbose* argument is different to 0, the opcode will print warning messages every k-pass if table lengths are exceeded.



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at k-rate (this means that every k-pass the vectors are subtracted). There's an i-rate version of this opcode called *vsubv_i*.



Note

Please note that the *elements* argument has changed in version 5.03 from i-rate to k-rate. This will change the opcode's behavior in the unusual cases where the i-rate variable *ielements* is changed inside the instrument, for example in:

```
instr 1
ielements = 10
vadd 1, 1, ielements
ielements = 20
vadd 2, 1, ielements
turnoff
endin
```

All these operators (*vaddv*, *vsubv*, *vmultv*, *vdivv*, *vpowv*, *vexpv*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Examples

Here is an example of the *vsubv* opcode. It uses the file *vsubv.csd* [examples/vsubv.csd].

Example 567. Example of the *vsubv* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o cigoto.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr=44100
ksmps=128
nchnls=2

instr 1
ifn1 = p4
ifn2 = p5
ielements = p6
idstoffset = p7
isrcoffset = p8
kval init 25
vsubv ifn1, ifn2, ielements, idstoffset, isrcoffset, 1
endin

instr 2 ;Printtable
itable = p4
isize = ftlen(itable)
kcount init 0
kval table kcount, itable
printk2 kval

if (kcount == isize) then
  turnoff
endif

kcount = kcount + 1
endin

</CsInstruments>
<CsScore>

f 1 0 16 -7 1 15 16
f 2 0 16 -7 1 15 2
```

```
i2 0.0 0.2 1
i2 0.2 0.2 2
i1 0.4 0.01 1 2 5 3 8
i2 0.8 0.2 1
i1 1.0 0.01 1 2 5 10 -2
i2 1.2 0.2 1
i1 1.4 0.01 1 2 8 14 0
i2 1.6 0.2 1
i1 1.8 0.01 1 2 8 0 14
i2 2.0 0.2 1
i1 2.2 0.002 1 1 8 5 2
i2 2.4 0.2 1
e
```

```
</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vsubv_i

vsubv_i — Performs subtraction between two vectorial control signals at init time.

Description

Performs subtraction between two vectorial control signals at init time.

Syntax

```
vsubv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]
```

Initialization

ifn1 - number of the table hosting the first vector to be processed

ifn2 - number of the table hosting the second vector to be processed

ielements - number of elements of the two vectors

idstoffset - index offset for the destination (*ifn1*) table (Default=0)

isrcoffset - index offset for the source (*ifn2*) table (Default=0)

Performance

vsubv_i subtracts two vectorial control signals, that is, each element of *ifn2* is subtracted from the corresponding element of *ifn1*. Each vectorial signal is hosted by a table (*ifn1* and *ifn2*). The number of elements contained in both vectors must be the same.

The result is a new vectorial control signal that overrides old values of *ifn1*. If you want to keep the old *ifn1* vector, use *vcopy_i* opcode to copy it in another table. You can use *idstoffset* and *isrcoffset* to specify vectors in any location of the tables.

Negative values for *idstoffset* and *isrcoffset* are acceptable. If *idstoffset* is negative, the out of range section of the vector will be discarded. If *isrcoffset* is negative, the out of range elements will be assumed to be 0 (i.e. the destination elements will not be changed). If elements for the destination vector are beyond the size of the table (including guard point), these elements are discarded (i.e. elements do not wrap around the tables). If elements for the source vector are beyond the table length, these elements are taken as 0 (i.e. the destination vector will not be changed for these elements).



Warning

Using the same table as source and destination table in versions earlier than 5.04, might produce unexpected behavior, so use with care.

This opcode works at init time. There's an k-rate version of this opcode called *vsubv*.

All these operators (*vaddv_i*, *vsubv_i*, *vmultv_i*, *vdivv_i*, *vpowv_i*, *vexpv_i*, *vcopy* and *vmap*) are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Credits

Written by Gabriel Maldonado. Optional arguments added by Andres Cabrera and Istvan Varga.

New in Csound 5 (Previously available only on CsoundAV)

vtable1k

vtable1k — Read a vector (several scalars simultaneously) from a table.

Description

This opcode reads vectors from tables at k-rate.

Syntax

```
vtable1k kfn,kout1 [, kout2, kout3, .... , koutN ]
```

Performance

kfn - table number

kout1...koutN - output vector elements

vtable1k is a reduced version of *vtablek*, it only allows to access the first vector (it is equivalent to *vtablek* with *kndx* = zero, but a bit faster). It is useful to easily and quickly convert a set of values stored in a table into a set of k-rate variables to be used in normal opcodes, instead of using individual *table* opcodes for each value.



Note

vtable1k is an unusual opcode as it produces its output on the right side arguments of the opcode.

Examples

Here is an example of the *vtable1k* opcode. It uses the files *vtable1k.csd* [examples/vtable1k.csd].

Example 568. Example of the vtable1k opcode.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 100
nchnls = 2

giElem init 13
giOutTab ftgen 1,0,128, 2,          0
giFreqTab ftgen 2,0,128,-7,        1,giElem, giElem+1
giSine ftgen 3,0,256,10, 1

FLpanel "This Panel contains a Slider Bank",500,400
FLslidBnk "mod1@mod2@mod3@amp@freq1@freq2@freq3@freqPo", giElem, giOutTab, 360, 600, 100, 1
FLpanel_end

FLrun

instr 1

kout1 init 0
kout2 init 0
```

```
kout3 init 0
kout4 init 0
kout5 init 0
kout6 init 0
kout7 init 0
kout8 init 0

vtablelk giOutTab, kout1 , kout2, kout3, kout4, kout5 , kout6, kout7, kout8
kmodindex1= 2 * db(kout1 * 80 )
kmodindex2= 2 * db(kout2 * 80 )
kmodindex3= 2 * db(kout3 * 80 )
kamp = 50 * db(kout4 * 70 )
kfreq1 = 1.1 * octave(kout5 * 10)
kfreq2 = 1.1 * octave(kout6 * 10)
kfreq3 = 1.1 * octave(kout7 * 10)
kfreq4 = 30 * octave(kout8 * 8)

amod1 oscili kmodindex1, kfreq1, giSine
amod2 oscili kmodindex2, kfreq2, giSine
amod3 oscili kmodindex3, kfreq3, giSine
aout oscili kamp, kfreq4+amod1+amod2+amod3, giSine

outs aout, aout
endin

</CsInstruments>
<CsScore>

i1 0 3600
f0 3600

</CsScore>
</CsoundSynthesizer>
```

See Also

vtablek

Credits

Written by Gabriel Maldonado.

New in Csound 5.06

vtablei

vtablei — Read vectors (from tables -or arrays of vectors).

Description

This opcode reads vectors from tables.

Syntax

```
vtablei  indx, ifn, interp, ixmode, iout1 [, iout2, iout3, .... , ioutN ]
```

Initialization

indx - Index into f-table, either a positive number range matching the table length (ixmode = 0) or a 0 to 1 range (ixmode != 0).

ifn - table number

iout1...ioutN - output vector elements

ixmode - index data mode. The default value is 0.

== 0 index is treated as a raw table location,

== 1 index is normalized (0 to 1).

interp - vtable (vector table) family of opcodes allows the user to switch between interpolated or non-interpolated output by means of the interp argument.

Performance

This opcode is useful in all cases in which one needs to access sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (iout1 , iout2, iout3, ioutN).

vtable (vector table) family of opcodes allows the user to switch between interpolated or non-interpolated output by means of the interp argument.

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using vtable, in order to correct eventual out-of-range values.



Note

Notice that *vtablei*'s output arguments are placed at the right of the opcode name, differently from usual (this style is already used in other opcodes using undefined lists of output arguments such as *fin* or *trigseq*).

Examples

Here is an example of the vtablei opcode. It uses the files *vtablei.csd* [examples/vtablei.csd]

Example 569. Example of the vtablei opcode.

```
<CsoundSynthesizer>
<CsOptions>
-odac -B441 -b441
</CsOptions>
<CsInstruments>

sr      =      44100
kr      =      100
ksmps   =      441
nchnls  =      2

gindx   init 0

      instr 1
kindex init 0
ktrig metro 0.5
if ktrig = 0 goto noevent
event "i", 2, 0, 0.5, kindex
kindex = kindex + 1
noevent:

      endin

      instr 2
iout1 init 0
iout2 init 0
iout3 init 0
iout4 init 0
indx = p4
vtablei indx, 1, 1, 0, iout1,iout2, iout3, iout4
print iout1, iout2, iout3, iout4
turnoff
      endin

</CsInstruments>
<CsScore>
f 1 0 32 10 1
i 1 0 20

</CsScore>
</CsoundSynthesizer>
```

See also

vtablea, vtablek, vtabi, vtablewi, vtabwi,

Credits

Example written by Andres Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

vtablek

vtablek — Read vectors (from tables -or arrays of vectors).

Description

This opcode reads vectors from tables at k-rate.

Syntax

```
vtablek kndx, kfn, kinterp, ixmode, kout1 [ , kout2, kout3, .... , koutN ]
```

Initialization

ixmode - index data mode. The default value is 0.

== 0 index is treated as a raw table location,

== 1 index is normalized (0 to 1).

kinterp - switch between interpolated or non-interpolated output. 0 -> non-interpolation , non-zero -> interpolation activated

Performance

kndx - Index into f-table, either a positive number range matching the table length (*ixmode* = 0) or a 0 to 1 range (*ixmode* != 0).

kfn - table number

kout1...koutN - output vector elements

This opcode is useful in all cases in which one needs to access sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (*kout1* , *kout2*, *kout3*, *koutN*).

vtablek allows the user to switch between interpolated or non-interpolated output at k-rate by means of *kinterp* argument.

vtablek allows also to switch the table number at k-rate (but this is possible only when vector frames of each used table have the same number of elements, otherwise unpredictable results could occur), as well as to choose indexing style (raw or normalized, see also *ixmode* argument of table opcode).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using *vtable*, in order to correct eventual out-of-range values.



Note

Notice that *vtablek*'s output arguments are placed at the left of the opcode name, differently from usual (this style is already used in other opcodes using undefined lists of output arguments such as *fin* or *trigseq*).

Examples

Here is an example of the `vtablek` opcode. It uses the files `vtablek.csd` [examples/vtablek.csd].

Example 570. Example of the `vtablek` opcode.

```
<CsoundSynthesizer>
<CsOptions>
-odac -B441 -b441
</CsOptions>
<CsInstruments>

sr      =      44100
kr      =      100
ksmps   =      441
nchnls  =      2

gkindx  init -1

      instr 1
kindex init 0
ktrig metro 0.5
if ktrig = 0 goto noevent
gkindx = gkindx + 1
noevent:

      endin

      instr 2
kout1 init 0
kout2 init 0
kout3 init 0
kout4 init 0
vtablek gkindx, 1, 1, 0, kout1,kout2, kout3, kout4
printk2 kout1
printk2 kout2
printk2 kout3
printk2 kout4
      endin

</CsInstruments>
<CsScore>
f 1 0 32 10 1
i 1 0 20
i 2 0 20
</CsScore>
</CsoundSynthesizer>
```

See also

`vtablea`, `vtablei`, `vtabk`, `vtablewk`, `vtabwk`,

Credits

Written by Gabriel Maldonado.

Example written by Andres Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

vtablea

vtablea — Read vectors (from tables -or arrays of vectors).

Description

This opcode reads vectors from tables at a-rate.

Syntax

```
vtablea andx, kfn, kinterp, ixmode, aout1 [, aout2, aout3, .... , aoutN ]
```

Initialization

ixmode - index data mode. The default value is 0.

== 0 index is treated as a raw table location,

== 1 index is normalized (0 to 1).

Performance

andx - Index into f-table, either a positive number range matching the table length (*ixmode* = 0) or a 0 to 1 range (*ixmode* != 0).

kfn - table number

kinterp - switch between interpolated or non-interpolated output. 0 -> non-interpolation , non-zero -> interpolation activated

aout1...aoutN - output vector elements

This opcode is useful in all cases in which one needs to access sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (*aout1* , *aout2*, *aout3*, *aoutN*).

vtablea allows the user to switch between interpolated or non-interpolated output at k-rate by means of *kinterp* argument.

vtablea allows also to switch the table number at k-rate (but this is possible only when vector frames of each used table have the same number of elements, otherwise unpredictable results could occur), as well as to choose indexing style (raw or normalized, see also *ixmode* argument of table opcode).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using *vtablea*, in order to correct eventual out-of-range values.



Note

Notice that *vtablea*'s output arguments are placed at the right of the opcode name, differently from usual (this style is already used in other opcodes using undefined lists of output arguments such as *fin* or *trigseq*).

See also

vtablek, vtablei, vtaba, vtablewa, vtabwa,

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vtablewi

vtablewi — Write vectors (to tables -or arrays of vectors).

Description

This opcode writes vectors to tables at init time.

Syntax

```
vtablewi  indx, ifn, ixmode, inarg1 [, inarg2, inarg3 , .... , inargN ]
```

Initialization

indx - Index into f-table, either a positive number range matching the table length (ixmode = 0) or a 0 to 1 range (ixmode != 0).

ifn - table number

ixmode - index data mode. The default value is 0.

== 0 index is treated as a raw table location,

== 1 index is normalized (0 to 1).

inarg1...inargN - output vector elements

Performance

This opcode is useful in all cases in which one needs to write sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (inarg1 , inarg2, inarg3, inargN).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using *vtablewi*, in order to correct eventual out-of-range values.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vtablewk

vtablewk — Write vectors (to tables -or arrays of vectors).

Description

This opcode writes vectors to tables at k-rate.

Syntax

```
vtablewk kndx, kfn, ixmode, kinarg1 [, kinarg2, kinarg3 , .... , kinargN ]
```

Initialization

ixmode - index data mode. The default value is 0. == 0 index is treated as a raw table location, == 1 index is normalized (0 to 1).

Performance

kndx - Index into f-table, either a positive number range matching the table length (*ixmode* = 0) or a 0 to 1 range (*ixmode* != 0).

kfn - table number

kinarg1...kinargN - output vector elements

This opcode is useful in all cases in which one needs to write sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (*kinarg1* , *kinarg2*, *kinarg3*, *kinargN*).

vtablewk allows also to switch the table number at k-rate (but this is possible only when vector frames of each used table have the same number of elements, otherwise unpredictable results could occur), as well as to choose indexing style (raw or normalized, see also *ixmode* argument of table opcode).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using *vtablewk*, in order to correct eventual out-of-range values.

Examples

Here is an example of the vtablewk opcode. It uses the files *vtablewk.csd* [examples/vtablewk.csd].

Example 571. Example of the vtablewk opcode.

```
<CsoundSynthesizer>
<CsOptions>
-odac -b441 -B441
</CsOptions>
<CsInstruments>
```

```
sr=44100
kr=4410
ksmps=10
```

```
nchnls=2

        instr 1
vcopy
ar random 0, 1
vtablewa ar
out ar,ar
        endin

</CsInstruments>
<CsScore>
f1 0 262144 -1 "beats.wav" 0 4 0
f2 0 262144 2 0

i1 0 4
i2 3 1

s
i1 0 4
i3 3 1
s

i1 0 4

</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado.

Example written by Andres Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

vtablewa

vtablewa — Write vectors (to tables -or arrays of vectors).

Description

This opcode writes vectors to tables at a-rate.

Syntax

```
vtablewa andx, kfn, ixmode, ainarg1 [, ainarg2, ainarg3 , .... , ainargN ]
```

Initialization

ixmode - index data mode. The default value is 0.

== 0 index is treated as a raw table location,

== 1 index is normalized (0 to 1).

Performance

andx - Index into f-table, either a positive number range matching the table length (*ixmode* = 0) or a 0 to 1 range (*ixmode* != 0).

kfn - table number

ainarg1...ainargN - input vector elements

This opcode is useful in all cases in which one needs to write sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (*ainarg1* , *ainarg2*, *ainarg3*, *ainargN*).

vtablewa allows also to switch the table number at k-rate (but this is possible only when vector frames of each used table have the same number of elements, otherwise unpredictable results could occur), as well as to choose indexing style (raw or normalized, see also *ixmode* argument of table opcode).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using *vtablewa*, in order to correct eventual out-of-range values.

Examples

Here is an example of the *vtablewa* opcode. It uses the files *vtablewa.csd* [examples/vtablewa.csd].

Example 572. Example of the vtablek opcode.

```
<CsoundSynthesizer>
<CsOptions>
;-ovtablewa.wav -W -b441 -B441
-odac -b441 -B441
</CsOptions>
<CsInstruments>
```



```
sr=44100
kr=441
ksmps=100
nchnls=2

instr 1
ilen = ftlen (1)

knew1 oscil 10000, 440, 3
knew2 oscil 15000, 440, 3, 0.5
kindex phasor 0.3
asig oscil 1, sr/ilen , 1
vtablewk kindex*ilen, 1, 0, knew1, knew2
out asig,asig
endin

</CsInstruments>
<CsScore>
f1 0 262144 -1 "beats.wav" 0 4 0
f2 0 262144 2 0
f3 0 1024 10 1

i1 0 10
</CsScore>
</CsoundSynthesizer>
```

Credits

Written by Gabriel Maldonado.

Example written by Andres Cabrera.

New in Csound 5 (Previously available only on CsoundAV)

vtabi

vtabi — Read vectors (from tables -or arrays of vectors).

Description

This opcode reads vectors from tables.

Syntax

```
vtabi  indx, ifn, iout1 [ , iout2, iout3, .... , ioutN ]
```

Initialization

indx - Index into f-table, either a positive number range matching the table length

ifn - table number

iout1...ioutN - output vector elements

Performance

This opcode is useful in all cases in which one needs to access sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (iout1 , iout2, iout3, ioutN).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using vtable, in order to correct eventual out-of-range values.

Notice that vtabi output arguments are placed at the left of the opcode name, differently from usual (this style is already used in other opcodes using undefined lists of output arguments such as fin or trigseq).

The **vtab** family is similar to **vtable**, but is much faster because interpolation is not available, table number cannot be changed after initialization, and only raw indexing is supported.



Note

Notice that *vtabi*'s output arguments are placed at the right of the opcode name, differently from usual (this style is already used in other opcodes using undefined lists of output arguments such as *fin* or *trigseq*).

Examples

For an example of the vtabi opcode usage, see *vtablei*.

See also

vtabk, *vtaba*, *vtablei*, *vtablewi*, *vtabwi*,

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vtabk

vtabk — Read vectors (from tables -or arrays of vectors).

Description

This opcode reads vectors from tables at k-rate.

Syntax

```
vtabk kndx, ifn, kout1 [ , kout2, kout3, .... , koutN ]
```

Initialization

ifn - table number

Performance

kndx - Index into f-table, either a positive number range matching the table length

kout1...koutN - output vector elements

This opcode is useful in all cases in which one needs to access sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.) . The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (kout1 , kout2, kout3, koutN).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using vtable, in order to correct eventual out-of-range values.

Notice that *vtabk* output arguments are placed at the left of the opcode name, differently from usual (this style is already used in other opcodes using undefined lists of output arguments such as *fin* or *trigseq*).

The **vtab** family is similar to **vtable**, but is much faster because interpolation is not available, table number cannot be changed after initialization, and only raw indexing is supported.



Note

Notice that *vtabk*'s output arguments are placed at the right of the opcode name, differently from usual (this style is already used in other opcodes using undefined lists of output arguments such as *fin* or *trigseq*).

Examples

For an example of the vtabk opcode usage, see *vtablek*.

See also

vtabi, *vtaba*, *vtablek*, *vtablewk*, *vtabwk*,

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vtaba

vtaba — Read vectors (from tables -or arrays of vectors).

Description

This opcode reads vectors from tables at a-rate.

Syntax

```
vtaba andx, ifn, aout1 [ , aout2, aout3, .... , aoutN ]
```

Initialization

ifn - table number

Performance

andx - Index into f-table, either a positive number range matching the table length

aout1...aoutN - output vector elements

This opcode is useful in all cases in which one needs to access sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (aout1 , aout2, aout3, aoutN).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using vtaba, in order to correct eventual out-of-range values.

Notice that **vtaba** output arguments are placed at the left of the opcode name, differently from usual (this style is already used in other opcodes using undefined lists of output arguments such as fin or trigseq).

The **vtab** family is similar to the **vtable** family, but is much faster because interpolation is not available, table number cannot be changed after initialization, and only raw indexing is supported.



Note

Notice that *vtaba*'s output arguments are placed at the right of the opcode name, differently from usual (this style is already used in other opcodes using undefined lists of output arguments such as *fin* or *trigseq*).

Examples

The usage of *vtaba* is similar to *vtablek*.

See also

vtabk, *vtabi*, *tablea*, *tablewa*, *vtabwa*,

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vtabwi

vtabwi — Write vectors (to tables -or arrays of vectors).

Description

This opcode writes vectors to tables at init time.

Syntax

```
vtabwi  indx, ifn, inarg1 [, inarg2, inarg3 , .... , inargN ]
```

Initialization

indx - Index into f-table, either a positive number range matching the table length (ixmode = 0) or a 0 to 1 range (ixmode != 0).

ifn - table number

inarg1...inargN - output vector elements

Performance

This opcode is useful in all cases in which one needs to write sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (inarg1 , inarg2, inarg3, inargN).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using *vtabwi*, in order to correct eventual out-of-range values.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vtabwk

vtabwk — Write vectors (to tables -or arrays of vectors).

Description

This opcode writes vectors to tables at a-rate.

Syntax

```
vtabwk kndx, ifn, kinarg1 [ , kinarg2, kinarg3 , .... , kinargN ]
```

Initialization

ifn - table number

Performance

kndx - Index into f-table, either a positive number range matching the table length (ixmode = 0) or a 0 to 1 range (ixmode != 0). *kinarg1...kinargN* - input vector elements

This opcode is useful in all cases in which one needs to write sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.) . The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (kinarg1 , kinarg2, kinarg3, kinargN).

vtabwk allows also to switch the table number at k-rate (but this is possible only when vector frames of each used table have the same number of elements, otherwise unpredictable results could occur), as well as to choose indexing style (raw or normalized, see also ixmode argument of table opcode).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using *vtabwk*, in order to correct eventual out-of-range values.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vtabwa

vtabwa — Write vectors (to tables -or arrays of vectors).

Description

This opcode writes vectors to tables at a-rate.

Syntax

```
vtabwa andx, ifn, ainarg1 [ , ainarg2, ainarg3 , .... , ainargN ]
```

Initialization

ifn - table number

Performance

andx - Index into f-table, either a positive number range matching the table length (ixmode = 0) or a 0 to 1 range (ixmode != 0).

ainarg1...ainargN - input vector elements

This opcode is useful in all cases in which one needs to write sets of values associated to unique indexes (for example, multi-channel samples, STFT bin frames, spectral formants, p-field based scores etc.). The number of elements of each vector (length of the vector) is determined by the number of optional arguments on the right (ainarg1 , ainarg2, ainarg3, ainargN).

vtabwa allows also to switch the table number at k-rate (but this is possible only when vector frames of each used table have the same number of elements, otherwise unpredictable results could occur), as well as to choose indexing style (raw or normalized, see also ixmode argument of table opcode).

Notice that no wrap nor limit mode is implemented. So, if an index attempt to access to a zone not allocated by the table, Csound will probably crash. However this drawback can be easily avoided by using wrap or limit opcodes applied to indexes before using *vtabwa*, in order to correct eventual out-of-range values.

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

vwrap

vwrap — Limiting and Wrapping Vectorial Signals

Description

Wraps elements of vectorial control signals.

Syntax

```
vwrap ifn, kmin, kmax, ielements
```

Initialization

ifn - number of the table hosting the vector to be processed

ielements - number of elements of the vector

Performance

kmin - minimum threshold value

kmax - maximum threshold value

vwrap wraps around each element of corresponding vector if it exceeds low or high thresholds.

These opcodes are similar to *limit*, *wrap* and *mirror*, but operate with a vectorial signal instead of with a scalar signal.

Result overrides old values of *ifn1*, if these are out of min/max interval. If you want to keep input vector, use *vcopy* opcode to copy it in another table.

All these opcodes are designed to be used together with other opcodes that operate with vectorial signals such as *bmscan*, *vcella*, *adsynt*, *adsynt2* etc.

Note: *bmscan* not yet available on Canonical Csound

Credits

Written by Gabriel Maldonado.

New in Csound 5 (Previously available only on CsoundAV)

waveset

waveset — A simple time stretch by repeating cycles.

Description

A simple time stretch by repeating cycles.

Syntax

```
ares waveset ain, krep [, ilen]
```

Initialization

ilen (optional, default=0) -- the length (in samples) of the audio signal. If *ilen* is set to 0, it defaults to half the given note length (p3).

Performance

ain -- the input audio signal.

krep -- the number of times the cycle is repeated.

The input is read and each complete cycle (two zero-crossings) is repeated *krep* times.

There is an internal buffer as the output is clearly slower than the input. Some care is taken if the buffer is too short, but there may be strange effects.

Examples

Here is an example of the waveset opcode. It uses the file *waveset.csd* [examples/waveset.csd], and *beats.wav* [examples/beats.wav].

Example 573. Example of the waveset opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o waveset.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - play an audio file.
instr 1
  asig soundin "beats.wav"
  out asig
endin
```

```
; Instrument #2 - stretch the audio file with waveset.
instr 2
  asig soundin "beats.wav"
  al waveset asig, 2

  out a1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for four seconds.
i 2 3 4
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch
February 2001

Example written by Kevin Conder.

New in version 4.11

weibull

weibull — Weibull distribution random number generator (positive values only).

Description

Weibull distribution random number generator (positive values only). This is an x-class noise generator

Syntax

```
ares weibull ksigma, ktau
```

```
ires weibull ksigma, ktau
```

```
kres weibull ksigma, ktau
```

Performance

ksigma -- scales the spread of the distribution.

ktau -- if greater than one, numbers near *ksigma* are favored. If smaller than one, small values are favored. If *t* equals 1, the distribution is exponential. Outputs only positive numbers.

For more detailed explanation of these distributions, see:

1. C. Dodge - T.A. Jerse 1985. Computer music. Schirmer books. pp.265 - 286
2. D. Lorrain. A panoply of stochastic cannons. In C. Roads, ed. 1989. Music machine . Cambridge, Massachusetts: MIT press, pp. 351 - 379.

Examples

Here is an example of the weibull opcode. It uses the file *weibull.csd* [examples/weibull.csd].

Example 574. Example of the weibull opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o weibull.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
```

```
; Generate a random number in a Weibull distribution.
; ksigma = 1
; ktau = 1

i1 weibull 1, 1

print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Its output should include a line like this:

```
instr 1:  i1 = 1.834
```

See Also

seed, betarand, bexprnd, cauchy, exprand, gauss, linrand, pcauchy, poisson, trirand, unirand

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Example written by Kevin Conder.

wgbow

wgbow — Creates a tone similar to a bowed string.

Description

Audio output is a tone similar to a bowed string, using a physical model developed from Perry Cook, but re-coded for Csound.

Syntax

```
ares wgbow kamp, kfreq, kpres, krat, kvibf, kvamp, ifn [, iminfreq]
```

Initialization

ifn -- table of shape of vibrato, usually a sine table, created by a function

iminfreq (optional) -- lowest frequency at which the instrument will play. If it is omitted it is taken to be the same as the initial *kfreq*. If *iminfreq* is negative, initialization will be skipped.

Performance

A note is played on a string-like instrument, with the arguments as below.

kamp -- amplitude of note.

kfreq -- frequency of note played.

kpres -- a parameter controlling the pressure of the bow on the string. Values should be about 3. The useful range is approximately 1 to 5.

krat -- the position of the bow along the string. Usual playing is about 0.127236. The suggested range is 0.025 to 0.23.

kvibf -- frequency of vibrato in Hertz. Suggested range is 0 to 12

kvamp -- amplitude of the vibrato

Examples

Here is an example of the wgbow opcode. It uses the file *wgbow.csd* [examples/wgbow.csd].

Example 575. Example of the wgbow opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o wgbow.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
```



```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 31129.60
  kfreq = 440
  kpres = 3.0
  krat = 0.127236
  kvibf = 6.12723
  ifn = 1

  ; Create an amplitude envelope for the vibrato.
  kv linseg 0, 0.5, 0, 1, 1, p3-0.5, 1
  kvamp = kv * 0.01

  al wgbow kamp, kfreq, kpres, krat, kvibf, kvamp, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 128 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

wgbowedbar

wgbowedbar — A physical model of a bowed bar.

Description

A physical model of a bowed bar, belonging to the Perry Cook family of waveguide instruments.

Syntax

```
ares wgbowedbar kamp, kfreq, kpos, kbowpres, kgain [, iconst] [, itvel] \  
    [, ibowpos] [, ilow]
```

Initialization

iconst (optional, default=0) -- an integration constant. Default is zero.

itvel (optional, default=0) -- either 0 or 1. When *ktvel* = 0, the bow velocity follows an ADSR style trajectory. When *ktvel* = 1, the value of the bow velocity decays in an exponentially.

ibowpos (optional, default=0) -- the position on the bow, which affects the bow velocity trajectory.

ilow (optional, default=0) -- lowest frequency required

Performance

kamp -- amplitude of signal

kfreq -- frequency of signal

kpos -- position of the bow on the bar, in the range 0 to 1

kbowpres -- pressure of the bow (as in *wgbowed*)

kgain -- gain of filter. A value of about 0.809 is suggested.

Examples

Here is an example of the wgbowedbar opcode. It uses the file *wgbowedbar.csd* [examples/wg-bowedbar.csd].

Example 576. Example of the wgbowedbar opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  No messages  
-odac        -iadc     -d      ;;RT audio I/O  
; For Non-realtime output leave only the line below:  
; -o wgbowedbar.wav -W ;; for file output any platform  
</CsOptions>  
<CsInstruments>  
  
; Initialize the global variables.  
sr = 44100  
kr = 4410
```

```
ksmps = 10
nchnls = 1

instr 1
; pos = [0, 1]
; bowpress = [1, 10]
; gain = [0.8, 1]
; intr = [0, 1]
; trackvel = [0, 1]
; bowpos = [0, 1]

kb line 0.5, p3, 0.1
kp line 0.6, p3, 0.7
kc line 1, p3, 1

a1 wgbowedbar p4, cpspch(p5), kb, kp, 0.995, p6, 0

out a1
endin

</CsInstruments>
<CsScore>

i1 0 3 32000 7.00 0
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 4.07

wgbrass

wgbrass — Creates a tone related to a brass instrument.

Description

Audio output is a tone related to a brass instrument, using a physical model developed from Perry Cook, but re-coded for Csound.

Syntax

```
ares wgbrass kamp, kfreq, ktens, iatt, kvibf, kvamp, ifn [, iminfreq]
```

Initialization

iatt -- time taken to reach full pressure

ifn -- table of shape of vibrato, usually a sine table, created by a function

iminfreq -- lowest frequency at which the instrument will play. If it is omitted it is taken to be the same as the initial *kfreq*. If *iminfreq* is negative, initialization will be skipped.

Performance

A note is played on a brass-like instrument, with the arguments as below.

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

ktens -- lip tension of the player. Suggested value is about 0.4

kvibf -- frequency of vibrato in Hertz. Suggested range is 0 to 12

kvamp -- amplitude of the vibrato



NOTE

This is rather poor, and at present uncontrolled. Needs revision, and possibly more parameters.

Examples

Here is an example of the wgbrass opcode. It uses the file *wgbrass.csd* [examples/wgbrass.csd].

Example 577. Example of the wgbrass opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc     -d      ;;RT audio I/O
```

```
; For Non-realtime ouput leave only the line below:
; -o wgbrass.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 31129.60
  kfreq = 440
  ktens = 0.4
  iatt = 0.1
  kvibf = 6.137
  ifn = 1

  ; Create an amplitude envelope for the vibrato.
  kvamp line 0, p3, 0.5

  al wgbrass kamp, kfreq, ktens, iatt, kvibf, kvamp, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 128 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

wgclar

wgclar — Creates a tone similar to a clarinet.

Description

Audio output is a tone similar to a clarinet, using a physical model developed from Perry Cook, but re-coded for Csound.

Syntax

```
ares wgclar kamp, kfreq, kstiff, iatt, idetk, kngain, kvibf, kvamp, ifn \  
    [, iminfreq]
```

Initialization

iatt -- time in seconds to reach full blowing pressure. 0.1 seems to correspond to reasonable playing. A longer time gives a definite initial wind sound.

idetk -- time in seconds taken to stop blowing. 0.1 is a smooth ending

ifn -- table of shape of vibrato, usually a sine table, created by a function

iminfreq (optional) -- lowest frequency at which the instrument will play. If it is omitted it is taken to be the same as the initial *kfreq*. If *iminfreq* is negative, initialization will be skipped.

Performance

A note is played on a clarinet-like instrument, with the arguments as below.

kamp -- Amplitude of note.

kfreq -- Frequency of note played.

kstiff -- a stiffness parameter for the reed. Values should be negative, and about -0.3. The useful range is approximately -0.44 to -0.18.

kngain -- amplitude of the noise component, about 0 to 0.5

kvibf -- frequency of vibrato in Hertz. Suggested range is 0 to 12

kvamp -- amplitude of the vibrato

Examples

Here is an example of the wgclar opcode. It uses the file *wgclar.csd* [examples/wgclar.csd].

Example 578. Example of the wgclar opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in    No messages
```

```
-odac          -iadc      -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o wgclar.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp init 31129.60
  kfreq = 440
  kstiff = -0.3
  iatt = 0.1
  idetk = 0.1
  kngain = 0.2
  kvibf = 5.735
  kvamp = 0.1
  ifn = 1

  al wgclar kamp, kfreq, kstiff, iatt, idetk, kngain, kvibf, kvamp, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

wgflute

wgflute — Creates a tone similar to a flute.

Description

Audio output is a tone similar to a flute, using a physical model developed from Perry Cook, but re-coded for Csound.

Syntax

```
ares wgflute kamp, kfreq, kjet, iatt, idetk, kngain, kvibf, kvamp, ifn \  
    [, iminfreq] [, ijetrf] [, iendrf]
```

Initialization

iatt -- time in seconds to reach full blowing pressure. 0.1 seems to correspond to reasonable playing.

idetk -- time in seconds taken to stop blowing. 0.1 is a smooth ending

ifn -- table of shape of vibrato, usually a sine table, created by a function

iminfreq (optional) -- lowest frequency at which the instrument will play. If it is omitted it is taken to be the same as the initial kfreq. If *iminfreq* is negative, initialization will be skipped.

ijetrf (optional, default=0.5) -- amount of reflection in the breath jet that powers the flute. Default value is 0.5.

iendrf (optional, default=0.5) -- reflection coefficient of the breath jet. Default value is 0.5. Both *ijetrf* and *iendrf* are used in the calculation of the pressure differential.

Performance

kamp -- Amplitude of note.

kfreq -- Frequency of note played. While it can be varied in performance, I have not tried it.

kjet -- a parameter controlling the air jet. Values should be positive, and about 0.3. The useful range is approximately 0.08 to 0.56.

kngain -- amplitude of the noise component, about 0 to 0.5

kvibf -- frequency of vibrato in Hertz. Suggested range is 0 to 12

kvamp -- amplitude of the vibrato

Examples

Here is an example of the wgflute opcode. It uses the file *wgflute.csd* [examples/wgflute.csd].

Example 579. Example of the wgflute opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.


```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc          -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o wgflute.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 31129.60
  kfreq = 440
  kjet = 0.32
  iatt = 0.1
  idetk = 0.1
  kngain = 0.15
  kvibf = 5.925
  kvamp = 0.05
  ifn = 1

  al wgflute kamp, kfreq, kjet, iatt, idetk, kngain, kvibf, kvamp, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

wgpluck

wgpluck — A high fidelity simulation of a plucked string.

Description

A high fidelity simulation of a plucked string, using interpolating delay-lines.

Syntax

ares **wgpluck** *icps*, *iamp*, *kpick*, *iplk*, *idamp*, *ifilt*, *axcite*

Initialization

icps -- frequency of plucked string

iamp -- amplitude of string pluck

iplk -- point along the string, where it is plucked, in the range of 0 to 1. 0 = no pluck

idamp -- damping of the note. This controls the overall decay of the string. The greater the value of *idamp*, the faster the decay. Negative values will cause an increase in output over time.

ifilt -- control the attenuation of the filter at the bridge. Higher values cause the higher harmonics to decay faster.

Performance

kpick -- proportion of the way along the point to sample the output.

axcite -- a signal which excites the string.

A string of frequency *icps* is plucked with amplitude *iamp* at point *iplk*. The decay of the virtual string is controlled by *idamp* and *ifilt* which simulate the bridge. The oscillation is sampled at the point *kpick*, and excited by the signal *axcite*.

Examples

The following example produces a moderately long note with rapidly decaying upper partials. It uses the file *wgpluck.csd* [examples/wgpluck.csd].

Example 580. An example of the wgpluck opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o wgpluck.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
```

```
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  icps = 220
  iamp = 20000
  kpick = 0.5
  iplk = 0
  idamp = 10
  ifilt = 1000

  axcite oscil 1, 1, 1
  apluck wgluck icps, iamp, kpick, iplk, idamp, ifilt, axcite

  out apluck
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

The following example produces a shorter, brighter note. It uses the file *wgluck_brighter.csd* [examples/wgluck_brighter.csd].

Example 581. An example of the wgluck opcode with a shorter, brighter note.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadac     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o wgluck_brighter.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  icps = 220
  iamp = 20000
  kpick = 0.5
  iplk = 0
  idamp = 30
  ifilt = 10

  axcite oscil 1, 1, 1
  apluck wgluck icps, iamp, kpick, iplk, idamp, ifilt, axcite

  out apluck
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e
```

```
</CsScore>  
</CsoundSynthesizer>
```

Credits

Author: Michael A. Casey
M.I.T.
Cambridge, Mass.
1997

New in Version 3.47

wgpluck2

wgpluck2 — Physical model of the plucked string.

Description

wgpluck2 is an implementation of the physical model of the plucked string, with control over the pluck point, the pickup point and the filter. Based on the Karplus-Strong algorithm.

Syntax

```
ares wgpluck2 iplk, kamp, icps, kpick, krefl
```

Initialization

iplk -- The point of pluck is *iplk*, which is a fraction of the way up the string (0 to 1). A pluck point of zero means no initial pluck.

icps -- The string plays at *icps* pitch.

Performance

kamp -- Amplitude of note.

kpick -- Proportion of the way along the string to sample the output.

krefl -- the coefficient of reflection, indicating the lossiness and the rate of decay. It must be strictly between 0 and 1 (it will complain about both 0 and 1).

Examples

Here is an example of the *wgpluck2* opcode. It uses the file *wgpluck2.csd* [examples/wgpluck2.csd].

Example 582. Example of the *wgpluck2* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o wgpluck2.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  iplk = 0.75
  kamp = 30000
  icps = 220
  kpick = 0.75
  krefl = 0.5
```

```
apluck wgpluck2 iplk, kamp, icps, kpick, krefl
out apluck
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

repluck

Credits

Author: John ffitch (after Perry Cook)
University of Bath, Codemist Ltd.
Bath, UK

New in Csound version 3.47

wguide1

wguide1 — A simple waveguide model consisting of one delay-line and one first-order lowpass filter.

Description

A simple waveguide model consisting of one delay-line and one first-order lowpass filter.

Syntax

```
ares wguide1 asig, xfreq, kcutoff, kfeedback
```

Performance

asig -- the input of excitation noise.

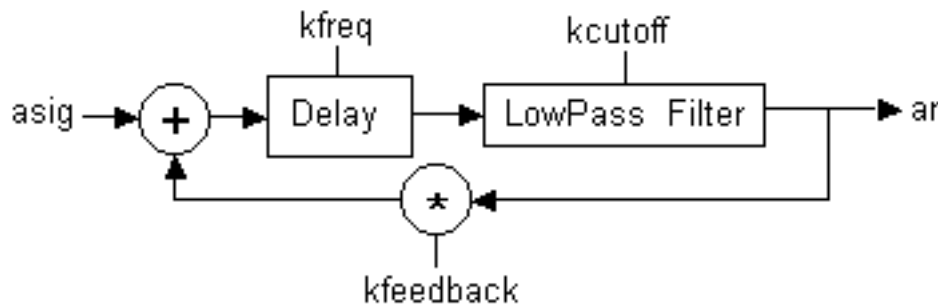
xfreq -- the frequency (i.e. the inverse of delay time) Changed to x-rate in Csound version 3.59.

kcutoff -- the filter cutoff frequency in Hz.

kfeedback -- the feedback factor.

wguide1 is the most elemental waveguide model, consisting of one delay-line and one first-order lowpass filter.

Implementing waveguide algorithms as opcodes, instead of orc instruments, allows the user to set *kr* different than *sr*, allowing better performance particularly when using real-time.



wguide1.

Examples

Here is an example of the *wguide1* opcode. It uses the file *wguide1.csd* [examples/wguide1.csd].

Example 583. Example of the *wguide1* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

<CsoundSynthesizer>

```
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in   No messages
-odac         -iadc      -d          ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o wguidel.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1 - a simple noise waveform.
instr 1
; Generate some noise.
asig noise 20000, 0.5

out asig
endin

; Instrument #2 - a waveguide example.
instr 2
; Generate some noise.
asig noise 20000, 0.5

; Run it through a wave-guide model.
kfreq init 200
kcutoff init 3000
kfeedback init 0.8
awgl wguidel asig, kfreq, kcutoff, kfeedback

out awgl
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for 2 seconds.
i 1 0 2
; Play Instrument #2 for 2 seconds.
i 2 2 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

wguide2

Credits

Author: Gabriel Maldonado
Italy
October 1998

Example written by Kevin Conder.

New in Csound version 3.49

wguide2

wguide2 — A model of beaten plate consisting of two parallel delay-lines and two first-order low-pass filters.

Description

A model of beaten plate consisting of two parallel delay-lines and two first-order lowpass filters.

Syntax

```
ares wguide2 asig, xfreq1, xfreq2, kcutoff1, kcutoff2, \  
      kfeedback1, kfeedback2
```

Performance

asig -- the input of excitation noise

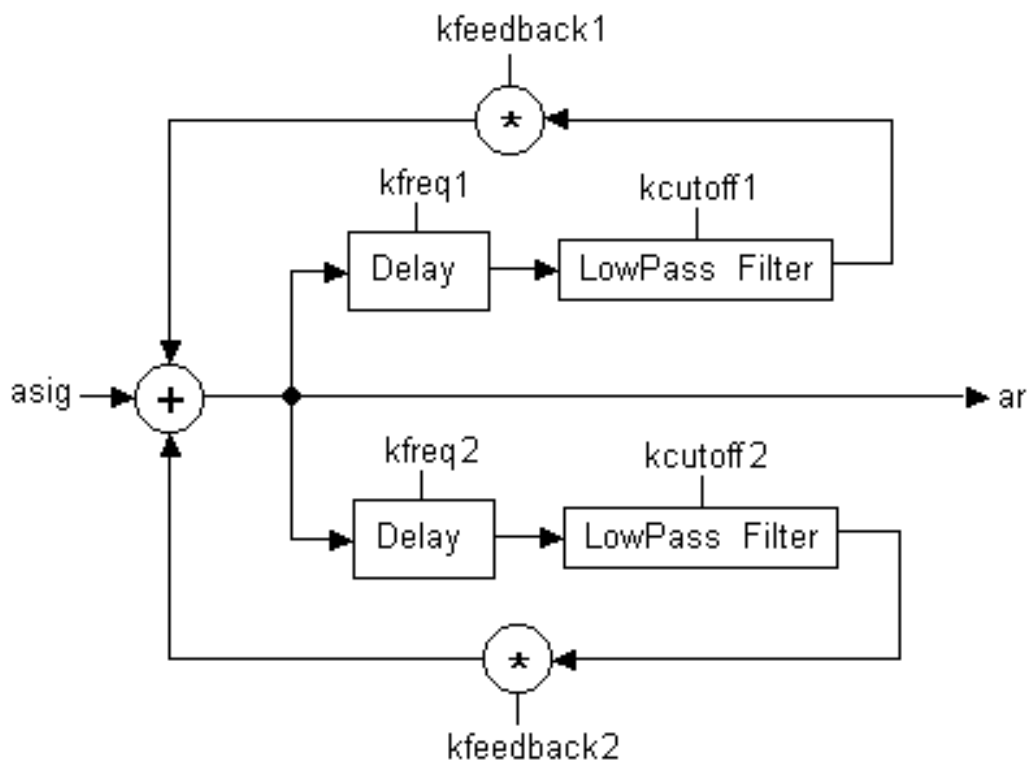
xfreq1, *xfreq2* -- the frequency (i.e. the inverse of delay time) Changed to x-rate in Csound version 3.59.

kcutoff1, *kcutoff2* -- the filter cutoff frequency in Hz.

kfeedback1, *kfeedback2* -- the feedback factor

wguide2 is a model of beaten plate consisting of two parallel delay-lines and two first-order lowpass filters. The two feedback lines are mixed and sent to the delay again each cycle.

Implementing waveguide algorithms as opcodes, instead of orc instruments, allows the user to set *kr* different than *sr*, allowing better performance particularly when using real-time.



wguide2.



Note

As a rule of thumb, to avoid making *wguide2* unstable, the sum of the two feedback values should be below 0.5.

Examples

Here is an example of the *wguide2* opcode. It uses the file *wguide2.csd* [examples/wguide2.csd].

Example 584. Example of the *wguide1* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o wguide1.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
sr = 44100
nchnls = 2
instr 1
  afrq line 50, 10, 100
  asig oscil 3000, afrq, 1
  aenv expon 1,10,0.000001
  aexc = aenv*asig
  ares wguide2 aexc, 500, 1200, 777, 1500, 0.2, 0.25
  out ares,asig
endin
```

```
</CsInstruments>
<CsScore>
f1 0 4096 10 1
i1 0 3
e
</CsScore>
</CsoundSynthesizer>
```

See Also

wguide1

Credits

Author: Gabriel Maldonado
Italy
October 1998

New in Csound version 3.49

Example written by John ffitch

wrap

`wrap` — Wraps-around the signal that exceeds the low and high thresholds.

Description

Wraps-around the signal that exceeds the low and high thresholds.

Syntax

```
ares wrap asig, klow, khigh
```

```
ires wrap isig, ilow, ihigh
```

```
kres wrap ksig, klow, khigh
```

Initialization

isig -- input signal

ilow -- low threshold

ihigh -- high threshold

Performance

xsig -- input signal

klow -- low threshold

khigh -- high threshold

`wrap` wraps-around the signal that exceeds the low and high thresholds.

This opcode is useful in several situations, such as table indexing or for clipping and modeling a-rate, i-rate or k-rate signals. `wrap` is also useful for wrap-around of table data when the maximum index is not a power of two (see *table* and *tablei*). Another use of `wrap` is in cyclical event repeating, with arbitrary cycle length.

See Also

limit, *mirror*

Credits

Author: Gabriel Maldonado
Italy

New in Csound version 3.49

wterrain

wterrain — A simple wave-terrain synthesis opcode.

Description

A simple wave-terrain synthesis opcode.

Syntax

```
aout wterrain kamp, kpch, k_xcenter, k_ycenter, k_xradius, k_yradius, \  
      itabx, itaby
```

Initialization

itabx, itaby -- The two tables that define the terrain.

Performance

The output is the result of drawing an ellipse with axes *k_xradius* and *k_yradius* centered at (*k_xcenter*, *k_ycenter*), and traversing it at frequency *kpch*.

Examples

Here is an example of the wterrain opcode. It uses the file *wterrain.csd* [examples/wterrain.csd].

Example 585. Example of the wterrain opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsOptions>  
; Select audio/midi flags here according to platform  
; Audio out  Audio in  No messages  
-odac       -iadc      -d      ;;RT audio I/O  
; For Non-realtime output leave only the line below:  
; -o wterrain.wav -W ;; for file output any platform  
</CsOptions>  
<CsInstruments>  
  
; Initialize the global variables.  
sr = 44100  
kr = 4410  
ksmps = 10  
nchnls = 1  
  
instr 1  
kdc1k linseg 0, 0.01, 1, p3-0.02, 1, 0.01, 0  
kcx line 0.1, p3, 1.9  
krx linseg 0.1, p3/2, 0.5, p3/2, 0.1  
kpch line cpspch(p4), p3, p5 * cpspch(p4)  
a1 wterrain 10000, kpch, kcx, kcx, -krx, krx, p6, p7  
a1 dcblock a1  
      out a1*kdc1k  
endin  
  
</CsInstruments>  
<CsScore>  
  
f1 0 8192 10 1 0 0.33 0 0.2 0 0.14 0 0.11  
f2 0 4096 10 1
```

```
i1      0      4      7.00 1 1 1
i1      4      4      6.07 1 1 2
i1      8      8      6.00 1 2 2
e
```

```
</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Matthew Gillard
New in version 4.19

xadsr

`xadsr` — Calculates the classical ADSR envelope.

Description

Calculates the classical ADSR envelope

Syntax

```
ares xadsr iatt, idec, islev, irel [, idel]
```

```
kres xadsr iatt, idec, islev, irel [, idel]
```

Initialization

iatt -- duration of attack phase

idec -- duration of decay

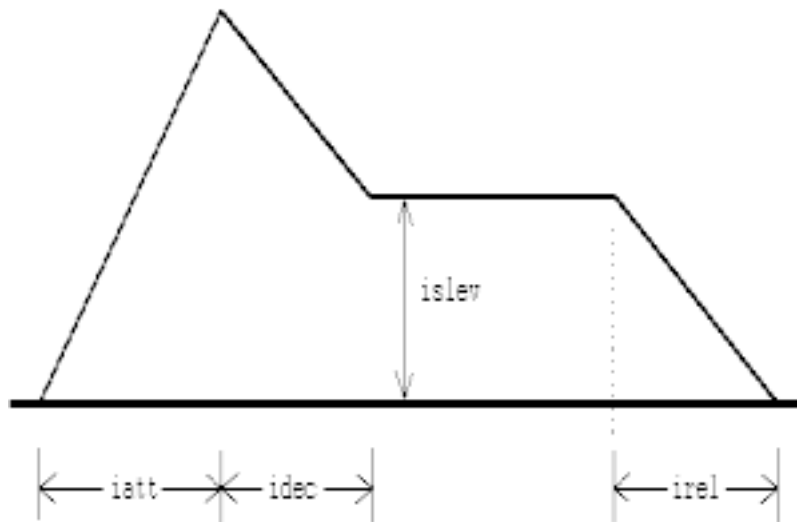
islev -- level for sustain phase

irel -- duration of release phase

idel -- period of zero before the envelope starts

Performance

The envelope is in the range 0 to 1 and may need to be scaled further. The envelope may be described as:



Picture of an ADSR envelope.

The length of the sustain is calculated from the length of the note. This means *xadsr* is not suitable for use with MIDI events, use *mxadsr* instead. The opcode *xadsr* is identical to *adsr* except it uses exponential, rather than linear, line segments.

xadsr is new in Csound version 3.51.

See Also

adsr, madsr, mxadsr

Credits

Author: John ffitch

xin

xin — Passes variables from a user-defined opcode block,

Description

The *xin* and *xout* opcodes copy variables to and from the opcode definition, allowing communication with the calling instrument.

The types of input and output variables are defined by the parameters *intypes* and *outtypes*.



Notes

- *xin* and *xout* should be called only once, and *xin* should precede *xout*, otherwise an init error and deactivation of the current instrument may occur.
- These opcodes actually run only at i-time. Performance time copying is done by the user opcode call. This means that skipping *xin* or *xout* with *kgoto* has no effect, while skipping with *igoto* affects both init and performance time operation.

Syntax

```
xinarg1 [, xinarg2] ... [xinargN] xin
```

Performance

xinarg1, *xinarg2*, ... - input arguments. The number and type of variables must agree with the user-defined opcode's *intypes* declaration. However, *xin* does not check for incorrect use of init-time and control-rate variables.

The syntax of a user-defined opcode block is as follows:

```
opcode name, outtypes, intypes
xinarg1 [, xinarg2] [, xinarg3] ... [xinargN] xin
[setksmps iksmps]
... the rest of the instrument's code.
xout xoutarg1 [, xoutarg2] [, xoutarg3] ... [xoutargN]
endop
```

The new opcode can then be used with the usual syntax:

```
[xinarg1] [, xinarg2] ... [xinargN] name [xoutarg1] [, xoutarg2] ... [xoutargN] [, iksmps]
```

Examples

See the example for the *opcode* opcode.

See Also

endop, opcode, setksmps, xout

Credits

Author: Istvan Varga, 2002; based on code by Matt J. Ingalls

New in version 4.22

xout

`xout` — Retrieves variables from a user-defined opcode block,

Description

The *xin* and *xout* opcodes copy variables to and from the opcode definition, allowing communication with the calling instrument.

The types of input and output variables are defined by the parameters *intypes* and *outtypes*.



Notes

- *xin* and *xout* should be called only once, and *xin* should precede *xout*, otherwise an init error and deactivation of the current instrument may occur.
- These opcodes actually run only at i-time. Performance time copying is done by the user opcode call. This means that skipping *xin* or *xout* with *kgoto* has no effect, while skipping with *igoto* affects both init and performance time operation.

Syntax

```
xout xoutarg1 [, xoutarg2] ... [, xoutargN]
```

Performance

xoutarg1, *xoutarg2*, ... - output arguments. The number and type of variables must agree with the user-defined opcode's *outtypes* declaration. However, *xout* does not check for incorrect use of init-time and control-rate variables.

The syntax of a user-defined opcode block is as follows:

```
opcode name, outtypes, intypes
xinarg1 [, xinarg2] [, xinarg3] ... [xinargN] xin
[setksmps iksmps]
... the rest of the instrument's code.
xout xoutarg1 [, xoutarg2] [, xoutarg3] ... [xoutargN]
endop
```

The new opcode can then be used with the usual syntax:

```
[xinarg1] [, xinarg2] ... [xinargN] name [xoutarg1] [, xoutarg2] ... [xoutargN] [, iksmps]
```

Examples

See the example for the *opcode* opcode.

See Also

endop, opcode, setksmps, xin

Credits

Author: Istvan Varga, 2002; based on code by Matt J. Ingalls

New in version 4.22

xscanmap

xscanmap — Allows the position and velocity of a node in a scanned process to be read.

Description

Allows the position and velocity of a node in a scanned process to be read.

Syntax

```
kpos, kvel xscanmap iscan, kamp, kvamp [, iwhich]
```

Initialization

iscan -- which scan process to read

iwhich (optional) -- which node to sense. The default is 0.

Performance

kamp -- amount to amplify the *kpos* value.

kvamp -- amount to amplify the *kvel* value.

The internal state of a node is read. This includes its position and velocity. They are amplified by the *kamp* and *kvamp* values.

Credits

Author: John ffitich

New in version 4.20

xscansmap

xscansmap — Allows the position and velocity of a node in a scanned process to be read.

Description

Allows the position and velocity of a node in a scanned process to be read.

Syntax

```
xscansmap kpos, kvel, iscan, kamp, kvamp [, iwhich]
```

Initialization

iscan -- which scan process to read

iwhich (optional) -- which node to sense. The default is 0.

Performance

kpos -- the node's position.

kvel -- the node's velocity.

kamp -- amount to amplify the *kpos* value.

kvamp -- amount to amplify the *kvel* value.

The internal state of a node is read. This includes its position and velocity. They are amplified by the *kamp* and *kvamp* values.

Credits

New in version 4.21

November 2002. Thanks to Rasmus Ekman for pointing this opcode out.

xscans

xscans — Fast scanned synthesis waveform and the wavetable generator.

Description

Experimental version of *scans*. Allows much larger matrices and is faster and smaller but removes some (unused?) flexibility. If liked, it will replace the older opcode as it is syntax compatible but extended.

Syntax

```
ares xscans kamp, kfreq, ifntraj, id [, iorder]
```

Initialization

ifntraj -- table containing the scanning trajectory. This is a series of numbers that contains addresses of masses. The order of these addresses is used as the scan path. It should not contain values greater than the number of masses, or negative numbers. See the *introduction to the scanned synthesis section*.

id -- If positive, the ID of the opcode. This will be used to point the scanning opcode to the proper waveform maker. If this value is negative, the absolute of this value is the wavetable on which to write the waveshape. That wavetable can be used later from an other opcode to generate sound. The initial contents of this table will be destroyed.

iorder (optional, default=0) -- order of interpolation used internally. It can take any value in the range 1 to 4, and defaults to 4, which is quartic interpolation. The setting of 2 is quadratic and 1 is linear. The higher numbers are slower, but not necessarily better.

Performance

kamp -- output amplitude. Note that the resulting amplitude is also dependent on instantaneous value in the wavetable. This number is effectively the scaling factor of the wavetable.

kfreq -- frequency of the scan rate

Matrix Format

The new matrix format is a list of connections, one per line linking point x to point y. There is no weight given to the link; it is assumed to be unity. The list is preceded by the line <MATRIX> and ends with a </MATRIX> line

For example, a circular string of 8 would be coded as

```
<MATRIX>
0 1
1 0
1 2
2 1
2 3
3 2
3 4
4 3
4 5
5 4
5 6
6 5
6 7
</MATRIX>
```

```
7 6  
0 7  
</MATRIX>
```

Credits

Written by John ffitch.

New in version 4.20

Examples

For an example, see the documentation on *scans*.

See Also

scans, *xscanu*

xscanu

xscanu — Compute the waveform and the wavetable for use in scanned synthesis.

Description

Experimental version of *scanu*. Allows much larger matrices and is faster and smaller but removes some (unused?) flexibility. If liked, it will replace the older opcode as it is syntax compatible but extended.

Syntax

```
xscanu init, irate, ifnvel, ifnmass, ifnstif, ifncentr, ifndamp, kmass, \
      kstif, kcentr, kdamp, ileft,  iright, kpos, kstrngth, ain, idisp, id
```

Initialization

init -- the initial position of the masses. If this is a negative number, then the absolute of *init* signifies the table to use as a hammer shape. If *init* > 0, the length of it should be the same as the intended mass number, otherwise it can be anything.

irate -- update rate.

ifnvel -- the ftable that contains the initial velocity for each mass. It should have the same size as the intended mass number.

ifnmass -- ftable that contains the mass of each mass. It should have the same size as the intended mass number.

ifnstif --

- *either* an ftable that contains the spring stiffness of each connection. It should have the same size as the square of the intended mass number. The data ordering is a row after row dump of the connection matrix of the system.
- *or* a string giving the name of a file in the MATRIX format

ifncentr -- ftable that contains the centering force of each mass. It should have the same size as the intended mass number.

ifndamp -- the ftable that contains the damping factor of each mass. It should have the same size as the intended mass number.

ileft -- If *init* < 0, the position of the left hammer (*ileft* = 0 is hit at leftmost, *ileft* = 1 is hit at rightmost).

iright -- If *init* < 0, the position of the right hammer (*iright* = 0 is hit at leftmost, *iright* = 1 is hit at rightmost).

idisp -- If 0, no display of the masses is provided.

id -- If positive, the ID of the opcode. This will be used to point the scanning opcode to the proper waveform maker. If this value is negative, the absolute of this value is the wavetable on which to write the waveshape. That wavetable can be used later from an other opcode to generate sound. The initial contents of this table will be destroyed.

Performance

kmass -- scales the masses

kstif -- scales the spring stiffness

kcentr -- scales the centering force

kdamp -- scales the damping

kpos -- position of an active hammer along the string (*kpos* = 0 is leftmost, *kpos* = 1 is rightmost). The shape of the hammer is determined by *init* and the power it pushes with is *kstrngth*.

kstrngth -- power that the active hammer uses

ain -- audio input that adds to the velocity of the masses. Amplitude should not be too great.

Matrix Format

The new matrix format is a list of connections, one per line linking point *x* to point *y*. There is no weight given to the link; it is assumed to be unity. The list is preceded by the line `<MATRIX>` and ends with a `</MATRIX>` line

For example, a circular string of 8 would be coded as

```
<MATRIX>
0 1
1 0
1 2
2 1
2 3
3 2
3 4
4 3
4 5
5 4
5 6
6 5
6 7
7 6
0 7
</MATRIX>
```

Credits

Written by John ffitch.

New in version 4.20

Examples

For an example, see the documentation on *scans*.

See Also

scanu, *xscans*

xtratim

xtratim — Extend the duration of real-time generated events.

Description

Extend the duration of real-time generated events and handle their extra life (Usually for usage along with *release* instead of *linenr*, *linsegr*, etc).

Syntax

```
xtratim iextradur
```

Initialization

iextradur -- additional duration of current instrument instance

Performance

xtratim extends current MIDI-activated note duration by *iextradur* seconds after the corresponding noteoff message has deactivated the current note itself. It is usually used in conjunction with *release*. This opcode has no output arguments.

This opcode is useful for implementing complex release-oriented envelopes, whose duration is not known when the envelope starts (e.g. for real-time MIDI generated events).

Examples

Here is a simple example of the xtratim opcode. It uses the file *xtratim.csd* [examples/xtratim.csd].

Example 586. Example of the xtratim opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

This example shows how to generate a release segment for an ADSR envelope after a MIDI noteoff is received, extending the duration with *xtratim* and using *release* to check whether the note is on the release phase.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  Silent MIDI in
-odac        -idac     -d      -M0    ;;;realtime I/O
</CsOptions>
<CsInstruments>
;Simple usage of the xtratim opcode

instr 1
  inum notnum
  icps cpsmidi
  iamp ampmidi 4000
;
;----- complex envelope block -----
xtratim 1 ;extra-time, i.e. release dur
krel init 0
krel release ;outputs release-stage flag (0 or 1 values)
if (krel == 1) kgoto rel ;if in release-stage goto release section
;
```

```
;***** attack and sustain section *****
kmp1 linseg 0, .03, 1, .05, 1, .07, 0, .08, .5, 4, 1, 50, 1
kmp = kmp1*iamp
kgoto done
;
;----- release section -----
rel:
kmp2 linseg 1, .3, .2, .7, 0
kmp = kmp1*kmp2*iamp
done:
;-----
al oscili kmp, icps, 1
out al
endin

</CsInstruments>
<CsScore>
f 0 3600 ;dummy table to wait for realtime MIDI events
e
</CsScore>
</CsoundSynthesizer>
```

Here is a more elaborate example of the *xtratim* opcode. It uses the file *xtratim-2.csd* [examples/xtratim-2.csd].

Example 587. More complex example of the *xtratim* opcode.

This example shows how to generate a release segment for an ADSR envelope after a MIDI noteoff is received, extending the duration with *xtratim* and using *release* to check whether the note is on the release phase. Two envelopes are generated simultaneously for the left and right channels.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out Audio in Silent MIDI in
-odac -idac -d -M0 ;;;realtime I/O
</CsOptions>
<CsInstruments>
;xtratim example by Jonathan Murphy Dec. 2006

sr = 44100
ksmps = 32
nchnls = 2

; sine wave for oscillators
gisin ftgen 1, 0, 4096, 10, 1
; set volume initially to midpoint
ctrlinit 1, 7, 64

;;; simple two oscil, two envelope synth
instr 1

; frequency
kcps cpsmidib
; initial velocity (noteon)
ivel veloc

; master volume
kamp ctrl7 1, 7, 0, 127
kamp = kamp * ivel

; parameters for aenv1
iatt1 = 0.03
idec1 = 1
isus1 = 0.25
irel1 = 1
; parameters for aenv2
iatt2 = 0.06
idec2 = 2
isus2 = 0.5
irel2 = 2

; extra (release) time allocated
xtratim (irel1>irel2 ? irel1 : irel2)
; krel is used to trigger envelope release
krel init 0
krel release
; if noteoff received, krel == 1, otherwise krel == 0
if (krel == 1) kgoto rel
```

```

                                ; attack, decay, sustain segments
atmp1    linseg    0, iatt1, 1, idec1, isus1 , 1, isus1
atmp2    linseg    0, iatt2, 1, idec2, isus2 , 1, isus2
aenv1    = atmp1
aenv2    = atmp2
                                kgoto    done

                                ; release segment
rel:
atmp3    linseg    1, irel1, 0, 1, 0
atmp4    linseg    1, irel2, 0, 1, 0
aenv1    = atmp1 * atmp3 ;to go from the current value (in case
aenv2    = atmp2 * atmp4 ;the attack hasn't finished) to the release.

                                ; control oscillator amplitude using envelopes
done:
aoscl    oscil     aenv1, kcps, 1
aoscl    oscil     aenv2, kcps * 1.5, 1
aoscl    = aoscl * kamp
aoscl    = aoscl * kamp

                                ; send aoscl to left channel, aoscl2 to right,
                                ; release times are noticeably different
                                outs      aoscl, aoscl2

                                endin

</CsInstruments>
<CsScore>

f 0 3600 ;dummy table to wait for realtime MIDI events

</CsScore>
</CsoundSynthesizer>
```

See Also

linenr, release

Credits

Author: Gabriel Maldonado

Italy

Examples by Gabriel Maldonado and Jonathan Murphy

New in Csound version 3.47

xyin

xyin — Sense the cursor position in an output window

Description

Sense the cursor position in an output window. When *xyin* is called the position of the mouse within the output window is used to reply to the request. This simple mechanism does mean that only one *xyin* can be used accurately at once. The position of the mouse is reported in the output window.

Syntax

```
kx, ky xyin iprd, ixmin, ixmax, iymn, iymax [, ixinit] [, iyinit]
```

Initialization

iprd -- period of cursor sensing (in seconds). Typically .1 seconds.

xmin, xmax, ymin, ymax -- edge values for the x-y coordinates of a cursor in the input window.

ixinit, iyinit (optional) -- initial x-y coordinates reported; the default values are 0,0. If these values are not within the given min-max range, they will be coerced into that range.

Performance

xyin samples the cursor x-y position in an input window every *iprd* seconds. Output values are repeated (not interpolated) at the k-rate, and remain fixed until a new change is registered in the window. There may be any number of input windows. This unit is useful for real-time control, but continuous motion should be avoided if *iprd* is unusually small.



Note

Depending on your platform and distribution, you might need to enable displays using the *--displays* command line flag.

Examples

Here is an example of the *xyin* opcode. It uses the file *xyin.csd* [examples/xyin.csd].

Example 588. Example of the *xyin* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      --displays ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o xyin.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
```

```
ksmpps = 10
nchnls = 1

; Instrument #1.
instr 1
; Print and capture values every 0.1 seconds.
iprd = 0.1
; The x values are from 1 to 30.
ixmin = 1
ixmax = 30
; The y values are from 1 to 30.
iymin = 1
iymax = 30
; The initial values for X and Y are both 15.
ixinit = 15
iyinit = 15

; Get the values kx and ky using the xyin opcode.
kx, ky xyin iprd, ixmin, ixmax, iymin, iymax, ixinit, iyinit

; Print out the values of kx and ky.
printks "kx=%f, ky=%f\\n", iprd, kx, ky

; Play an oscillator, use the x values for amplitude and
; the y values for frequency.
kamp = kx * 1000
kcps = ky * 220
a1 oscil kamp, kcps, 1

out a1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 30 seconds.
i 1 0 30
e

</CsScore>
</CsSoundSynthesizer>
```

As the values of kx and ky change, they will be printed out like this:

```
kx=8.612036, ky=22.677933
kx=10.765685, ky=15.644135
```

Credits

Example written by Kevin Conder.

zACL

zACL — Clears one or more variables in the za space.

Description

Clears one or more variables in the za space.

Syntax

```
zACL kfirst, klast
```

Performance

kfirst -- first zk or za location in the range to clear.

klast -- last zk or za location in the range to clear.

zACL clears one or more variables in the za space. This is useful for those variables which are used as accumulators for mixing a-rate signals at each cycle, but which must be cleared before the next set of calculations.

Examples

Here is an example of the zACL opcode. It uses the file *zACL.csd* [examples/zACL.csd].

Example 589. Example of the zACL opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d          ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o zACL.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zAKinit 1, 1

; Instrument #1 -- a simple waveform.
instr 1
; Generate a simple sine waveform.
asin oscil 20000, 440, 1

; Send the sine waveform to za variable #1.
zaw asin, 1
endin

; Instrument #2 -- generates audio output.
instr 2
; Read za variable #1.
a1 zar 1
```



```
; Generate the audio output.
out al

; Clear the za variables, get them ready for
; another pass.
zawl 0, 1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zamod, zar, zaw, zawm, ziw, ziwM

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zakinit

zakinit — Establishes zak space.

Description

Establishes zak space. Must be called only once.

Syntax

```
zakinit isizea, isizek
```

Initialization

isizea -- the number of audio rate locations for a-rate patching. Each location is actually an array which is *ksmps* long.

isizek -- the number of locations to reserve for floats in the zk space. These can be written and read at i- and k-rates.

Performance

At least one location each is always allocated for both za and zk spaces. There can be thousands or tens of thousands za and zk ranges, but most pieces probably only need a few dozen for patching signals. These patching locations are referred to by number in the other zak opcodes.

To run *zakinit* only once, put it outside any instrument definition, in the orchestra file header, after *sr*, *kr*, *ksmps*, and *nchnls*.



Note

Zak channels count from 0, so if you define 1 channel, the only valid channel is channel 0.

Examples

Here is an example of the *zakinit* opcode. It uses the file *zakinit.csd* [examples/zakinit.csd].

Example 590. Example of the *zakinit* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o zakinit.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 4410
nchnls = 1
```

```
; Initialize the ZAK space.
; Create 3 a-rate variables and 5 k-rate variables.
zakinit 2, 3

instr 1 ; a simple waveform.
      ; Generate a simple sine waveform.
      asin oscil 20000, 440, 1

      ; Send the sine waveform to za variable #1.
      zaw asin, 1
endin

instr 2 ; generates audio output.
      ; Read za variable #1.
      a1 zar 1

      ; Generate audio output.
      out a1

      ; Clear the za variables, get them ready for
      ; another pass.
      zac1 0, 3
endin

instr 3 ; increments k-type channels
      k0 zkr 0
      k1 zkr 1
      k2 zkr 2

      zkw k0+1, 0
      zkw k1+5, 1
      zkw k2+10, 2
endin

instr 4 ; displays values from k-type channels
      k0 zkr 0
      k1 zkr 1
      k2 zkr 2

; The total count for k0 is 30, since there are 10
; control blocks per second and instruments 3 and 4
; are on for 3 seconds.
      printf "k0 = %i\n", k0, k0
      printf "k1 = %i\n", k1, k1
      printf "k2 = %i\n", k2, k2
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

i 1 0 1
i 2 0 1

i 3 0 3
i 4 0 3
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zamod

zamod — Modulates one a-rate signal by a second one.

Description

Modulates one a-rate signal by a second one.

Syntax

ares **zamod** asig, kzamod

Performance

asig -- the input signal

kzamod -- controls which za variable is used for modulation. A positive value means additive modulation, a negative value means multiplicative modulation. A value of 0 means no change to *asig*.

zamod modulates one a-rate signal by a second one, which comes from a za variable. The location of the modulating variable is controlled by the i-rate or k-rate variable *kzamod*. This is the a-rate version of *zkmod*.

Examples

Here is an example of the zamod opcode. It uses the file *zamod.csd* [examples/zamod.csd].

Example 591. Example of the zamod opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o zamod.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 2 a-rate variables and 2 k-rate variables.
zakinit 2, 2

; Instrument #1 -- a simple waveform.
instr 1
; Vary an a-rate signal linearly from 20,000 to 0.
asig line 20000, p3, 0

; Send the signal to za variable #1.
zaw asig, 1
endin

; Instrument #2 -- generates audio output.
instr 2
; Generate a simple sine wave.
```

```
asin oscil 1, 440, 1

; Modify the sine wave, multiply its amplitude by
; za variable #1.
al zamod asin, -1

; Generate the audio output.
out al

; Clear the za variables, prepare them for
; another pass.
zaci 0, 2
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
; Play Instrument #2 for 2 seconds.
i 2 0 2
e

</CsScore>
</CsSoundSynthesizer>
```

See Also

zaci, ziw, ziwm

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zar

zir — Reads from a location in za space at a-rate.

Description

Reads from a location in za space at a-rate.

Syntax

ares **zar** kndx

Performance

kndx -- points to the za location to be read.

zar reads the array of floats at *kndx* in za space, which are ksmps number of a-rate floats to be processed in a k cycle.

Examples

Here is an example of the zar opcode. It uses the file *zar.csd* [examples/zar.csd].

Example 592. Example of the zar opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac       -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o zar.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a simple waveform.
instr 1
; Generate a simple sine waveform.
asin oscil 20000, 440, 1

; Send the sine waveform to za variable #1.
zaw asin, 1
endin

; Instrument #2 -- generates audio output.
instr 2
; Read za variable #1.
a1 zar 1

; Generate audio output.
out a1
```

```
    ; Clear the za variables, get them ready for
    ; another pass.
    zACL 0, 1
endin

</CsInstruments>
<CsScore>

    ; Table #1, a sine wave.
    f 1 0 16384 10 1

    ; Play Instrument #1 for one second.
    i 1 0 1
    ; Play Instrument #2 for one second.
    i 2 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zarg, zir, zkr

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zarg

zarg — Reads from a location in za space at a-rate, adds some gain.

Description

Reads from a location in za space at a-rate, adds some gain.

Syntax

```
ares zarg kndx, kgain
```

Initialization

kndx -- points to the za location to be read.

kgain -- multiplier for the a-rate signal.

Performance

zarg reads the array of floats at *kndx* in za space, which are ksmpls number of a-rate floats to be processed in a k cycle. *zarg* also multiplies the a-rate signal by a k-rate value *kgain*.

Examples

Here is an example of the *zarg* opcode. It uses the file *zarg.csd* [examples/zarg.csd].

Example 593. Example of the *zarg* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o zarg.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a simple waveform.
instr 1
; Generate a simple sine waveform, with an amplitude
; between 0 and 1.
asin oscil 1, 440, 1

; Send the sine waveform to za variable #1.
zaw asin, 1
endin

; Instrument #2 -- generates audio output.
```



```
instr 2
; Read za variable #1, multiply its amplitude by 20,000.
a1 zarg 1, 20000

; Generate audio output.
out a1

; Clear the za variables, get them ready for
; another pass.
zacl 0, 1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 0 1
e

</CsScore>
</CsSoundSynthesizer>
```

See Also

zar, zir, zkr

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zaw

zaw — Writes to a za variable at a-rate without mixing.

Description

Writes to a za variable at a-rate without mixing.

Syntax

zaw asig, kndx

Performance

asig -- value to be written to the za location.

kndx -- points to the zk or za location to which to write.

zaw writes *asig* into the za variable specified by *kndx*.

These opcodes are fast, and always check that the index is within the range of zk or za space. If not, an error is reported, 0 is returned, and no writing takes place.

Examples

Here is an example of the zaw opcode. It uses the file *zaw.csd* [examples/zaw.csd].

Example 594. Example of the zaw opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o zaw.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a simple waveform.
instr 1
; Generate a simple sine waveform.
asin oscil 20000, 440, 1

; Send the sine waveform to za variable #1.
zaw asin, 1
endin

; Instrument #2 -- generates audio output.
instr 2
; Read za variable #1.
```

```
al zar 1

; Generate the audio output.
out al

; Clear the za variables, get them ready for
; another pass.
zac1 0, 1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zawm, ziw, ziwm, zkw, zkwm

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zawm

zawm — Writes to a za variable at a-rate with mixing.

Description

Writes to a za variable at a-rate with mixing.

Syntax

```
zawm asig, kndx [, imix]
```

Initialization

imix (optional, default=1) -- indicates if mixing should occur.

Performance

asig -- value to be written to the za location.

kndx -- points to the zk or za location to which to write.

These opcodes are fast, and always check that the index is within the range of zk or za space. If not, an error is reported, 0 is returned, and no writing takes place.

zawm is a mixing opcode, it adds the signal to the current value of the variable. If no *imix* is specified, mixing always occurs. *imix* = 0 will cause overwriting like *ziw*, *zkw*, and *zaw*. Any other value will cause mixing.

Caution: When using the mixing opcodes *ziwm*, *zkwm*, and *zawm*, care must be taken that the variables mixed to, are zeroed at the end (or start) of each k- or a-cycle. Continuing to add signals to them, can cause their values can drift to astronomical figures.

One approach would be to establish certain ranges of zk or za variables to be used for mixing, then use *zkcl* or *zacl* to clear those ranges.

Examples

Here is an example of the *zawm* opcode. It uses the file *zawm.csd* [examples/zawm.csd].

Example 595. Example of the *zawm* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o zawm.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
```

```
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a basic instrument.
instr 1
  ; Generate a simple sine waveform.
  asin oscil 15000, 440, 1

  ; Mix the sine waveform with za variable #1.
  zawm asin, 1
endin

; Instrument #2 -- another basic instrument.
instr 2
  ; Generate another waveform with a different frequency.
  asin oscil 15000, 880, 1

  ; Mix this sine waveform with za variable #1.
  zawm asin, 1
endin

; Instrument #3 -- generates audio output.
instr 3
  ; Read za variable #1, containing both waveforms.
  al zar 1

  ; Generate the audio output.
  out al

  ; Clear the za variables, get them ready for
  ; another pass.
  zacl 0, 1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 0 1
; Play Instrument #3 for one second.
i 3 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zaw, ziw, ziwm, zkw, zkwm

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zfilter2

zfilter2 — Performs filtering using a transposed form-II digital filter lattice with radial pole-shearing and angular pole-warping.

Description

General purpose custom filter with time-varying pole control. The filter coefficients implement the following difference equation:

$$(1)*y(n) = b0*x[n] + b1*x[n-1] + \dots + bM*x[n-M] - a1*y[n-1] - \dots - aN*y[n-N]$$

the system function for which is represented by:

$$H(Z) = \frac{B(Z)}{A(Z)} = \frac{b0 + b1*Z^{-1} + \dots + bM*Z^{-M}}{1 + a1*Z^{-1} + \dots + aN*Z^{-N}}$$

Syntax

```
ares zfilter2 asig, kdamp, kfreq, iM, iN, ib0, ib1, ..., ibM, \  
      ial, ia2, ..., iaN
```

Initialization

At initialization the number of zeros and poles of the filter are specified along with the corresponding zero and pole coefficients. The coefficients must be obtained by an external filter-design application such as Matlab and specified directly or loaded into a table via *GEN01*. With *zfilter2*, the roots of the characteristic polynomials are solved at initialization so that the pole-control operations can be implemented efficiently.

Performance

The *filter2* opcodes perform filtering using a transposed form-II digital filter lattice with no time-varying control. *zfilter2* uses the additional operations of radial pole-shearing and angular pole-warping in the Z plane.

Pole shearing increases the magnitude of poles along radial lines in the Z-plane. This has the affect of altering filter ring times. The k-rate variable *kdamp* is the damping parameter. Positive values (0.01 to 0.99) increase the ring-time of the filter (hi-Q), negative values (-0.01 to -0.99) decrease the ring-time of the filter, (lo-Q).

Pole warping changes the frequency of poles by moving them along angular paths in the Z plane. This operation leaves the shape of the magnitude response unchanged but alters the frequencies by a constant factor (preserving 0 and p). The k-rate variable *kfreq* determines the frequency warp factor. Positive values (0.01 to 0.99) increase frequencies toward p and negative values (-0.01 to -0.99) decrease frequencies toward 0.

Since *filter2* implements generalized recursive filters, it can be used to specify a large range of general DSP algorithms. For example, a digital waveguide can be implemented for musical instrument modeling using a pair of *delayr* and *delayw* opcodes in conjunction with the *filter2* opcode.

Examples

A controllable second-order IIR filter operating on an a-rate signal:

```
a1 zfilter2 asig, kdamp, kfreq, 1, 2, 1, ia1, ia2 ; controllable a-rate ; IIR filter
```

See Also

filter2

Credits

Author: Michael A. Casey
M.I.T.
Cambridge, Mass.
1997

New in Version 3.47

zir

zir — Reads from a location in zk space at i-rate.

Description

Reads from a location in zk space at i-rate.

Syntax

```
ir zir indx
```

Initialization

indx -- points to the zk location to be read.

Performance

zir reads the signal at *indx* location in zk space.

Examples

Here is an example of the zir opcode. It uses the file *zir.csd* [examples/zir.csd].

Example 596. Example of the zir opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o zir.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a simple instrument.
instr 1
; Set the zk variable #1 to 32.594.
ziw 32.594, 1
endin

; Instrument #2 -- prints out zk variable #1.
instr 2
; Read the zk variable #1 at i-rate.
il zir 1

; Print out the value of zk variable #1.
print il
endin
```



```
</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zar, zarg, zkr

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

ziw

ziw — Writes to a zk variable at i-rate without mixing.

Description

Writes to a zk variable at i-rate without mixing.

Syntax

```
ziw isig, indx
```

Initialization

isig -- initializes the value of the zk location.

indx -- points to the zk or za location to which to write.

Performance

ziw writes *isig* into the zk variable specified by *indx*.

These opcodes are fast, and always check that the index is within the range of zk or za space. If not, an error is reported, 0 is returned, and no writing takes place.

Examples

Here is an example of the ziw opcode. It uses the file *ziw.csd* [examples/ziw.csd].

Example 597. Example of the ziw opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ziw.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a simple instrument.
instr 1
; Set zk variable #1 to 64.182.
ziw 64.182, 1
endin

; Instrument #2 -- prints out zk variable #1.
instr 2
```

```
; Read zk variable #1 at i-rate.
il zir 1

; Print out the value of zk variable #1.
print il
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zaw, zawm, ziwm, zkw, zkwm

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

ziwm

ziwm — Writes to a zk variable to an i-rate variable with mixing.

Description

Writes to a zk variable to an i-rate variable with mixing.

Syntax

```
ziwm isig, indx [, imix]
```

Initialization

isig -- initializes the value of the zk location.

indx -- points to the zk location location to which to write.

imix (optional, default=1) -- indicates if mixing should occur.

Performance

ziwm is a mixing opcode, it adds the signal to the current value of the variable. If no *imix* is specified, mixing always occurs. *imix* = 0 will cause overwriting like *ziw*, *zkw*, and *zaw*. Any other value will cause mixing.

Caution: When using the mixing opcodes *ziwm*, *zkwm*, and *zawm*, care must be taken that the variables mixed to, are zeroed at the end (or start) of each k- or a-cycle. Continuing to add signals to them, can cause their values can drift to astronomical figures.

One approach would be to establish certain ranges of zk or za variables to be used for mixing, then use *zkcl* or *zacl* to clear those ranges.

Examples

Here is an example of the *ziwm* opcode. It uses the file *ziwm.csd* [examples/ziwm.csd].

Example 598. Example of the *ziwm* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o ziwm.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
```

```
; Create 1 a-rate variable and 1 k-rate variable.
zakin 1, 1

; Instrument #1 -- a simple instrument.
instr 1
  ; Add 20.5 to zk variable #1.
  ziwm 20.5, 1
endin

; Instrument #2 -- another simple instrument.
instr 2
  ; Add 15.25 to zk variable #1.
  ziwm 15.25, 1
endin

; Instrument #3 -- prints out zk variable #1.
instr 3
  ; Read zk variable #1 at i-rate.
  i1 zir 1

  ; Print out the value of zk variable #1.
  ; It should be 35.75 (20.5 + 15.25)
  print i1
endin

</CsInstruments>
<CsScore>

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 0 1
; Play Instrument #3 for one second.
i 3 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zaw, zawm, ziw, zkw, zkwm

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zkcl

zkcl — Clears one or more variables in the zk space.

Description

Clears one or more variables in the zk space.

Syntax

```
zkcl kfirst, klast
```

Performance

ksig -- the input signal

kfirst -- first zk or za location in the range to clear.

klast -- last zk or za location in the range to clear.

zkcl clears one or more variables in the zk space. This is useful for those variables which are used as accumulators for mixing k-rate signals at each cycle, but which must be cleared before the next set of calculations.

Examples

Here is an example of the zkcl opcode. It uses the file *zkcl.csd* [examples/zkcl.csd].

Example 599. Example of the zkcl opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o zkcl.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a simple waveform.
instr 1
; Linearly vary a k-rate signal from 220 to 1760.
kline line 220, p3, 1760

; Add the linear signal to zk variable #1.
zkw kline, 1
endin

; Instrument #2 -- generates audio output.
instr 2
```

```
; Read zk variable #1.
kfreq zkr 1

; Use the value of zk variable #1 to vary
; the frequency of a sine waveform.
a1 oscil 20000, kfreq, 1

; Generate the audio output.
out a1

; Clear the zk variables, get them ready for
; another pass.
zkcl 0, 1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for three seconds.
i 1 0 3
; Play Instrument #2 for three seconds.
i 2 0 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zacr, zkwm, zkw, zkmod, zkr

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zkmod

zkmod — Facilitates the modulation of one signal by another.

Description

Facilitates the modulation of one signal by another.

Syntax

```
kres zkmod ksig, kzkmod
```

Performance

ksig -- the input signal

kzkmod -- controls which zk variable is used for modulation. A positive value means additive modulation, a negative value means multiplicative modulation. A value of 0 means no change to *ksig*. *kzkmod* can be i-rate or k-rate

zkmod facilitates the modulation of one signal by another, where the modulating signal comes from a zk variable. Either additive or multiplicative modulation can be specified.

Examples

Here is an example of the zkmod opcode. It uses the file *zkmod.csd* [examples/zkmod.csd].

Example 600. Example of the zkmod opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac          -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o zkmod.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 2

; Initialize the ZAK space.
; Create 2 a-rate variables and 2 k-rate variables.
zakinit 2, 2

; Instrument #1 -- a signal with jitter.
instr 1
; Generate a k-rate signal goes from 30 to 2,000.
kline line 30, p3, 2000

; Add the signal into zk variable #1.
zkw kline, 1
endin

; Instrument #2 -- generates audio output.
instr 2
; Create a k-rate signal modulated the jitter opcode.
```



```
kamp init 20
kcpsmin init 40
kcpsmax init 60
kjtr jitter kamp, kcpsmin, kcpsmax

; Get the frequency values from zk variable #1.
kfreq zkr 1
; Add the the frequency values in zk variable #1 to
; the jitter signal.
kjfreq zkmod kjtr, 1

; Use a simple sine waveform for the left speaker.
aleft oscil 20000, kfreq, 1
; Use a sine waveform with jitter for the right speaker.
aright oscil 20000, kjfreq, 1

; Generate the audio output.
outs aleft, aright

; Clear the zk variables, prepare them for
; another pass.
zkcl 0, 2
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
; Play Instrument #2 for 2 seconds.
i 2 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zamod, zkcl, zkr, zkwm, zkw

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zkr

zkr — Reads from a location in zk space at k-rate.

Description

Reads from a location in zk space at k-rate.

Syntax

```
kres zkr kndx
```

Initialization

kndx -- points to the zk location to be read.

Performance

zkr reads the array of floats at *kndx* in zk space.

Examples

Here is an example of the zkr opcode. It uses the file *zkr.csd* [examples/zkr.csd].

Example 601. Example of the zkr opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o zkr.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a simple waveform.
instr 1
; Linearly vary a k-rate signal from 440 to 880.
kline line 440, p3, 880

; Add the linear signal to zk variable #1.
zkw kline, 1
endin

; Instrument #2 -- generates audio output.
instr 2
; Read zk variable #1.
kfreq zkr 1

; Use the value of zk variable #1 to vary
```

```
; the frequency of a sine waveform.
a1 oscil 20000, kfreq, 1

; Generate the audio output.
out a1

; Clear the zk variables, get them ready for
; another pass.
zkcl 0, 1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for one second.
i 1 0 1
; Play Instrument #2 for one second.
i 2 0 1
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zar, zarg, zir, zkcl, zkmod, zkwm, zkw

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zkw

zkw — Writes to a zk variable at k-rate without mixing.

Description

Writes to a zk variable at k-rate without mixing.

Syntax

zkw ksig, kndx

Performance

ksig -- value to be written to the zk location.

kndx -- points to the zk or za location to which to write.

zkw writes *ksig* into the zk variable specified by *kndx*.

Examples

Here is an example of the *zkw* opcode. It uses the file *zkw.csd* [examples/zkw.csd].

Example 602. Example of the *zkw* opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac      -iadc      -d      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o zkw.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a simple waveform.
instr 1
; Linearly vary a k-rate signal from 100 to 1,000.
kline line 100, p3, 1000

; Add the linear signal to zk variable #1.
zkw kline, 1
endin

; Instrument #2 -- generates audio output.
instr 2
; Read zk variable #1.
kfreq zkr 1

; Use the value of zk variable #1 to vary
; the frequency of a sine waveform.
```

```
al oscil 20000, kfreq, 1

; Generate the audio output.
out al

; Clear the zk variables, get them ready for
; another pass.
zkcl 0, 1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
; Play Instrument #2 for two seconds.
i 2 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

zaw, zawm, ziw, ziwm, zkr, zkwm

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

zkwm

zkwm — Writes to a zk variable at k-rate with mixing.

Description

Writes to a zk variable at k-rate with mixing.

Syntax

```
zkwm ksig, kndx [, imix]
```

Initialization

imix (optional, default=1) -- indicates if mixing should occur.

Performance

ksig -- value to be written to the zk location.

kndx -- points to the zk or za location to which to write.

zkwm is a mixing opcode, it adds the signal to the current value of the variable. If no *imix* is specified, mixing always occurs. *imix* = 0 will cause overwriting like *ziw*, *zkw*, and *zaw*. Any other value will cause mixing.

Caution: When using the mixing opcodes *ziwm*, *zkwm*, and *zawm*, care must be taken that the variables mixed to, are zeroed at the end (or start) of each k- or a-cycle. Continuing to add signals to them, can cause their values can drift to astronomical figures.

One approach would be to establish certain ranges of zk or za variables to be used for mixing, then use *zkcl* or *zacl* to clear those ranges.

Examples

Here is an example of the *zkwm* opcode. It uses the file *zkwm.csd* [examples/zkwm.csd].

Example 603. Example of the zkwm opcode.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in  No messages
-odac        -iadc     -d      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o zkwm.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Initialize the ZAK space.
```

```
; Create 1 a-rate variable and 1 k-rate variable.
zakinit 1, 1

; Instrument #1 -- a basic instrument.
instr 1
; Generate a k-rate signal.
; The signal goes from 30 to 20,000 then back to 30.
kramp linseg 30, p3/2, 20000, p3/2, 30

; Mix the signal into the zk variable #1.
zkwm kramp, 1
endin

; Instrument #2 -- another basic instrument.
instr 2
; Generate another k-rate signal.
; This is a low frequency oscillator.
klfo lfo 3500, 2

; Mix this signal into the zk variable #1.
zkwm klfo, 1
endin

; Instrument #3 -- generates audio output.
instr 3
; Read zk variable #1, containing a mix of both signals.
kamp zkr 1

; Create a sine waveform. Its amplitude will vary
; according to the values in zk variable #1.
a1 oscil kamp, 880, 1

; Generate the audio output.
out a1

; Clear the zk variable, get it ready for
; another pass.
zkcl 0, 1
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; Play Instrument #1 for 5 seconds.
i 1 0 5
; Play Instrument #2 for 5 seconds.
i 2 0 5
; Play Instrument #3 for 5 seconds.
i 3 0 5
e

</CsScore>
</CsSoundSynthesizer>
```

See Also

zaw, zawm, ziw, ziwm, zkcl, zkw, zkr

Credits

Author: Robin Whittle
Australia
May 1997

New in version 3.45

Example written by Kevin Conder.

Score Statements and GEN Routines

Score Statements

The statements used in scores are:

- *a* - Advance score time by a specified amount
- *b* - Resets the clock
- *e* - Marks the end of the last section of the score
- *f* - Causes a *GEN subroutine* to place values in a stored function table
- *i* - Makes an instrument active at a specific time and for a certain duration
- *m* - Sets a named mark in the score
- *n* - Repeats a marked section
- *q* - Used to quiet an instrument
- *r* - Starts a repeated section
- *s* - Marks the end of a section
- *t* - Sets the tempo
- *v* - Provides for locally variable time warping of score events
- *x* - Skip the rest of the current section
- *{* - Begins a non-sectional, nestable loop
- *}* - Ends a non-sectional, nestable loop

a Statement (or Advance Statement)

a — Advance score time by a specified amount.

Description

This causes score time to be advanced by a specified amount without producing sound samples.

Syntax

a p1 p2 p3

Performance

p1	Carries no meaning. Usually zero.
p2	Action time, in beats, at which advance is to begin.
p3	Number of beats to advance without producing sound.
p4	
p5	These carry no meaning.
p6	
.	
.	

Special Considerations

This statement allows the beat count within a score section to be advanced without generating intervening sound samples. This can be of use when a score section is incomplete (the beginning or middle is missing) and the user does not wish to generate and listen to a lot of silence.

p2, action time, and p3, number of beats, are treated as in *i statements*, with respect to sorting and modification by *t statements*.

An *a statement* will be temporarily inserted in the score by the Score Extract feature when the extracted segment begins later than the start of a Section. The purpose of this is to preserve the beat count and time count of the original score for the benefit of the peak amplitude messages which are reported on the user console.

Whenever an *a statement* is encountered by a performing orchestra, its presence and effect will be reported on the user's console.

b Statement

b Statement — This statement resets the clock.

Description

This statement resets the clock.

Syntax

b p1

Performance

p1 -- Specifies how the clock is to be set.

Special Considerations

p1 is the number of beats by which p2 values of subsequent *i statements* are modified. If p1 is positive, the clock is reset forward, and subsequent notes appear later, the number of beats specified by p1 being added to the note's p2. If p1 is negative, the clock is reset backward, and subsequent notes appear earlier, the number of beats specified by p1 being subtracted from the note's p2. There is no cumulative affect. The clock is reset with each *b statement*. If p1 = 0, the clock is returned to its original position, and subsequent notes appear at their specified p2.

Examples

```
i1      0      2
i1      10     888

b 5
i2      1      1      440      ; set the clock "forward"
i2      2      1      480      ; start time = 6
                                   ; start time = 7

b -1
i3      3      2      3.1415    ; set the clock back
i3      5.5    1      1.1111    ; start time = 2
                                   ; start time = 4.5

b 0
i4      10     200    7          ; reset clock to normal
                                   ; start time = 10
```

Credits

Explanation suggested and example provided by Paul Winkler. (Csound Version 4.07)

e Statement

e statement — This statement may be used to mark the end of the last section of the score.

Description

This statement may be used to mark the end of the last section of the score.

Syntax

e [*time*]

Performance

The first p-field *time* is optional and if present determines the end time (length in beats) of the final section of the score. This time must be after the end of the last event otherwise it will have no effect. "Always on" instruments will end at the given time. Extending the section in this way is useful to avoid prematurely cutting off reverb tails or other effects.

Special Considerations

The *e statement* is contextually identical to an *s statement*. Additionally, the *e statement* terminates all signal generation (including indefinite performance) and closes all input and output files.

If an *e statement* occurs before the end of a score, all subsequent score lines will be ignored.

The *e statement* is optional in a score file yet to be sorted. If a score file has no *e statement*, then Sort processing will supply one.

f Statement (or Function Table Statement)

f Statement (or Function Table Statement) — Causes a GEN subroutine to place values in a stored function table.

Description

This causes a GEN subroutine to place values in a stored function table for use by instruments.

Syntax

f *p1* *p2* *p3* *p4* *p5* ... *PMAX*

Performance

p1 -- Table number by which the stored function will be known. A negative number requests that the table be destroyed.

p2 -- Action time of function generation (or destruction) in beats.

p3 -- Size of function table (i.e. number of points) Must be a power of 2, or a power-of-2 plus 1 if this number is positive. Maximum table size is 16777216 (2^{24}) points.

p4 -- Number of the GEN routine to be called (see *GEN ROUTINES*). A negative value will cause rescaling to be omitted.

p5 ... *PMAX* -- Parameters whose meaning is determined by the particular GEN routine.

Special Considerations

Function tables are arrays of floating-point values. You can create a simple sine wave using the line:

```
f 1 0 1024 10 1
```

This table uses *GEN10* to fill the table.

Historically, due to older platform constraints, Csound could only accept tables whose size was a power of two. This limitation has been removed in recent versions, and you can freely create tables of any size. However, to create a table whose size is not a power of two (or power of two plus one), you must specify the size as a negative number.



Note

Not all opcodes accept tables whose size is not a power of two, since they may depend on this for internal optimization.

For arrays whose length is a power of 2, space allocation always provides for 2^n points plus an additional *guard point*. The guard point value, used during interpolated lookup, can be automatically set to reflect the table's purpose: If *size* is an exact power of 2, the guard point will be a copy of the first point; this is appropriate for *interpolated wrap-around lookup* as in *oscili*, etc., and should even be used for non-interpolating *oscil* for safe consistency. If *size* is set to $2^n + 1$, the guard point value automatically extends the contour of table values; this is appropriate for single-scan functions such in *envplx*, *oscill*, *oscilli*, etc.

Table space is allocated in primary memory, along with instrument data space. The maximum table number used to be 200. This has been changed to be limited by memory only. (Currently there is an internal soft limit of 300, this is automatically extended as required.)

An existing function table can be removed by an *f statement* containing a negative p1 and an appropriate action time. A function table can also be removed by the generation of another table with the same p1. Functions are not automatically erased at the end of a score section.

p2 action time is treated in the same way as in *i statements* with respect to sorting and modification by *t statements*. If an *f statement* and an *i statement* have the same p2, the sorter gives the *f statement* precedence so that the function table will be available during note initialization.



Warning

The maximum number of p-fields accepted in the score is determined by PMAX (a compilation time variable). PMAX is currently set to 1000. This may discard values entered when using *GEN02*. To overcome this, use *GEN23* or *GEN28* to read the values from a file.

An *f 0 statement* (zero p1, positive p2) may be used to create an action time with no associated action. Such time markers are useful for padding out a score section (see *s statement*) and for letting Csound run from realtime events only (e.g. using only MIDI input without score events). The time given is the number of seconds Csound will run. If you want Csound to run for 10 hours, use:

```
f0 36000
```

The simplest way to fill a table (f1) with 0's is:

```
f1 0 xx 2 0
```

where xx = table size.

The simplest way to fill a table (f1) with *any* single value is:

```
f1 0 xx -7 yy xx yy
```

where xx = table size and yy = any single value

In both of the above examples, table size (p3) must be a power of 2 or power-of-2 plus 1.

See also

GEN ROUTINES

Credits

Updated August 2002 thanks to a note from Rasmus Ekman. There is no longer a hard limit of 200 function tables.

i Statement (Instrument or Note Statement)

i — Makes an instrument active at a specific time and for a certain duration.

Description

This statement calls for an instrument to be made active at a specific time and for a certain duration. The parameter field values are passed to that instrument prior to its initialization, and remain valid throughout its Performance.

Syntax

i p1 p2 p3 p4 ...

Initialization

p1 -- Instrument number, usually a non-negative integer. An optional fractional part can provide an additional tag for specifying ties between particular notes of consecutive clusters. A negative *p1* (including tag) can be used to turn off a particular “held” note.

p2 -- Starting time in arbitrary units called beats.

p3 -- Duration time in beats (usually positive). A negative value will initiate a held note (see also *ihold*). A negative value can also be used for 'always on' instruments like reverberation. These notes are not terminated by *s statements*. A zero value will invoke an initialization pass without performance (see also *instr*).

p4 ... -- Parameters whose significance is determined by the instrument.

Performance

Beats are evaluated as seconds, unless there is a *t statement* in this score section or a *-t flag* in the command-line.

Starting or action times are relative to the beginning of a section (see *s statement*), which is assigned time 0.

Note statements within a section may be placed in any order. Before being sent to an orchestra, unordered score statements must first be processed by Sorter, which will reorder them by ascending *p2* value. Notes with the same *p2* value will be ordered by ascending *p1*; if the same *p1*, then by ascending *p3*.

Notes may be stacked, i.e., a single instrument can perform any number of notes simultaneously. (The necessary copies of the instrument's data space will be allocated dynamically by the orchestra loader.) Each note will normally turn off when its *p3* duration has expired, or on receipt of a MIDI noteoff signal. An instrument can modify its own duration either by changing its *p3* value during note initialization, or by prolonging itself through the action of a *linenr* or *xtratim* unit.

An instrument may be turned on and left to perform indefinitely either by giving it a negative *p3* or by including an *ihold* in its *i-time* code. If a held note is active, an *i statement with matching p1* will not cause a new allocation but will take over the data space of the held note. The new *pfields* (including *p3*) will now be in effect, and an *i-time* pass will be executed in which the units can either be newly initialized or allowed to continue as required for a tied note (see *tigoto*). A held note may be succeeded either by another held note or by a note of finite duration. A held note will continue to perform across section endings (see *s statement*). It is halted only by *turnoff* or by an *i statement* with negative matching *p1* or by an *e statement*.

It is possible to have multiple instances (usually, but not necessarily, notes of different pitches) of the same instrument, held simultaneously, via negative p3 values. The instrument can then be fed new parameters from the score. This is useful for avoiding long hard-coded *linsegs*, and can be accomplished by adding a decimal part to the instrument number.

For example, to hold three copies of instrument 10 in a simple chord:

```
i10.1    0    -1    7.00
i10.2    0    -1    7.04
i10.3    0    -1    7.07
```

Subsequent *i* statements can refer to the same sounding note instances, and if the instrument definition is done properly, the new p-fields can be used to alter the character of the notes in progress. For example, to bend the previous chord up an octave and release it:

```
i10.1    1     1    8.00
i10.2    1     1    8.04
i10.3    1     1    8.07
```



Tip

When turning off notes, bear in mind that `i 1.1 == i 1.10` and `i 1.1 != i 1.01`. The maximum number of decimal places that can be used depends on the precision Csound was compiled with (See *Csound Double (64-bit) vs. Float (32-bit)*)

The instrument definition has to take this into account, however, especially if clicks are to be avoided (see the example below).

Note that the decimal instrument number notation cannot be used in conjunction with real-time MIDI. In this case, the instrument would be monophonic while a note was held.

Notes being tied to previous instances of the same instrument, should skip most initialization by means of *tigoto*, except for the values entered in score. For example, all table reading opcodes in the instrument, should usually be skipped, as they store their phase internally. If this is suddenly changed, there will be audible clicks in the output.

Note that many opcodes (such as *delay* and *reverb*) are prepared for optional initialization. To use this feature, the *tival opcode* is suitable. Therefore, they need not be hidden by a *tigoto* jump.

Beginning with Csound version 3.53, strings are recognized in p-fields for opcodes that accept them (*convolve*, *adsyn*, *diskin*, etc.). There may be only one string per score line.

Special Considerations

The maximum instrument number used to be 200. This has been changed to be limited by memory only (currently there is an internal soft limit of 200; this is automatically extended as required).

Examples

Here is an instrument which can find out whether it is tied to a previous note (*tival* returns 1), and whether it is held (negative p3). Attack and release are handled accordingly:

```
instr 10
  icps      init      cpspch(p4)           ; Get target pitch from score event
  iportime  init      abs(p3)/7           ; Portamento time dep on note length
```

```
iamp0    init      p5                      ; Set default amps
iamp1    init      p5
iamp2    init      p5

itie     tival
if itie == 1      igoto nofadein          ; Check if this note is tied,
iamp0     init      0                      ; if not fade in

nofadein:
  if p3 < 0        igoto nofadeout        ; Check if this note is held, if not fade out
  iamp2     init      0

nofadeout:
  ; Now do amp from the set values:
  kamp     linseg    iamp0, .03, iamp1, abs(p3)-.03, iamp2

  ; Skip rest of initialization on tied note:
  tigoto     tieskip

  kcps     init      icps                    ; Init pitch for untied note
  kcps     port      icps, iportime, icps    ; Drift towards target pitch

  kpw      oscil    .4, rnd(1), 1, rnd(.7)    ; A simple triangle-saw oscil
  ar       vco      kamp, kcps, 3, kpw+.5, 1, 1/icps

  ; (Used in testing - one may set ipch to cpspch(p4+2)
  ;      and view output spectrum)
  ;      ar oscil kamp, kcps, 1

  out      ar

tieskip:                      ; Skip some initialization on tied note
endin
```

A simple score using three instances of the above instrument:

```
f1    0 8192 10 1              ; Sine
i10.1  0    -1    7.00    10000
i10.2  0    -1    7.04
i10.3  0    -1    7.07
i10.1  1    -1    8.00
i10.2  1    -1    8.04
i10.3  1    -1    8.07
i10.1  2     1    7.11
i10.2  2     1    8.04
i10.3  2     1    8.07
e
```

Credits

Additional text (Csound Version 4.07) explaining tied notes, edited by Rasmus Ekman from a note by David Kirsh, posted to the Csound mailing list. Example instrument by Rasmus Ekman.

Updated August 2002 thanks to a note from Rasmus Ekman. There is no longer a hard limit of 200 instruments.

m Statement (Mark Statement)

`m` — Sets a named mark in the score.

Description

Sets a named mark in the score, which can be used by an *n statement*.

Syntax

`m p1`

Initialization

p1 -- Name of mark.

Performance

This can be helpful in setting a up verse and chorus structure in the score. Names may contain letters and numerals.

Credits

Author: John fitch
University of Bath/Codemist Ltd.
Bath, UK
April, 1998

New in Csound version 3.48

n Statement

n — Repeats a section.

Description

Repeats a section from the referenced *m statement*.

Syntax

n *p1*

Initialization

p1 -- Name of mark to repeat.

Performance

This can be helpful in setting a up verse and chorus structure in the score. Names may contain letters and numerals.

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
April 1998

New in Csound version 3.48

q Statement

q statement — This statement may be used to quiet an instrument.

Description

This statement may be used to quiet an instrument.

Syntax

q *p1* *p2* *p3*

Performance

p1 -- Instrument number to mute/unmute.

p2 -- Action time in beats.

p3 -- determines whether the instrument is muted/unmuted. The value of 0 means the instrument is muted, other values mean it is unmuted.

Note that this does not affect instruments that are already running at time *p2*. It blocks any attempt to start one afterwards.

r Statement (Repeat Statement)

r — Starts a repeated section.

Description

Starts a repeated section, which lasts until the next *s*, *r* or *e statement*.

Syntax

r *p1* *p2*

Initialization

p1 -- Number of times to repeat the section.

p2 -- Macro(name) to advance with each repetition (optional).

Performance

In order that the sections may be more flexible than simple editing, the macro named *p2* is given the value of 1 for the first time through the section, 2 for the second, and 3 for the third. This can be used to change p-field parameters, or ignored.



Warning

Because of serious problems of interaction with macro expansion, sections must start and end in the same file, and not in a macro.

Examples

Here is an example of the *r* statement. It uses the file *r.sco* [examples/r.csd].

Example 604. Example of the *r* statement.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o r.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; The score's p4 parameter has the number of repeats.
kreps = p4
; The score's p5 parameter has our note's frequency.
kcps = p5

; Print the number of repeats.
printks "Repeated %i time(s).\n", 1, kreps
```

```
; Generate a nice beep.
al oscil 20000, kcps, 1
out al
endin

</CsInstruments>
<CsScore>

; Table #1, a sine wave.
f 1 0 16384 10 1

; We'll repeat this section 6 times. Each time it
; is repeated, its macro REPS_MACRO is incremented.
r6 REPS_MACRO

; Play Instrument #1.
; p4 = the r statement's macro, REPS_MACRO.
; p5 = the frequency in cycles per second.
i 1 00.10 00.10 $REPS_MACRO 1760
i 1 00.30 00.10 $REPS_MACRO 880
i 1 00.50 00.10 $REPS_MACRO 440
i 1 00.70 00.10 $REPS_MACRO 220

; Marks the end of the section.
s

e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK
April, 1998

New in Csound version 3.48

Example written by Kevin Conder

s Statement

s — Marks the end of a section.

Description

The *s statement* marks the end of a section.

Syntax

s [*time*]

Initialization

The first p-field *time* is optional and if present determines the end time (length in beats) of the section. This time must be after the end of the last event in the section otherwise it will have no effect. This can be used to create a pause before the beginning of the next section or to allow "always on" instruments such as effects to play by themselves for some length of time.

Performance

Sorting of the *i statement*, *f statement* and *a statement* by action time is done section by section.

Time warping for the *t statement* is done section by section.

All action times within a section are relative to its beginning. A section statement establishes a new relative time of 0, but has no other reinitializing effects (e.g. stored function tables are preserved across section boundaries).

A section is considered complete when all action times and finite durations have been satisfied (i.e., the "length" of a section is determined by the last occurring action or turn-off). A section can be extended by the use of an *fo statement* or by supplying the optional *p1* value to the *s statement*.

A section ending automatically invokes a purge of inactive instrument and data spaces.



Note

- Since score statements are processed section by section, the amount of memory required depends on the maximum number of score statements in a section. Memory allocation is dynamic, and the user will be informed as extra memory blocks are requested during score processing.
- For the end of the final section of a score, the *s statement* is optional; the *e statement* may be used instead.

t Statement (Tempo Statement)

t — Sets the tempo.

Description

This statement sets the tempo and specifies the accelerations and decelerations for the current section. This is done by converting beats into seconds.

Syntax

`t p1 p2 p3 p4 ... (unlimited)`

Initialization

p1 -- Must be zero.

p2 -- Initial tempo on beats per minute.

p3, p5, p7,... -- Times in beats per minute (in non-decreasing order).

p4, p6, p8,... -- Tempi for the referenced beat times.

Performance

Time and Tempo-for-that-time are given as ordered couples that define points on a "tempo vs. time" graph. (The time-axis here is in beats so is not necessarily linear.) The beat-rate of a Section can be thought of as a movement from point to point on that graph: motion between two points of equal height signifies constant tempo, while motion between two points of unequal height will cause an accelerando or ritardando accordingly. The graph can contain discontinuities: two points given equal times but different tempi will cause an immediate tempo change.

Motion between different tempos over non-zero time is inverse linear. That is, an accelerando between two tempos M1 and M2 proceeds by linear interpolation of the single-beat durations from 60/M1 to 60/M2.

The first tempo given must be for beat 0.

A tempo, once assigned, will remain in effect from that time-point unless influenced by a succeeding tempo, i.e. the last specified tempo will be held to the end of the section.

A *t statement* applies only to the score section in which it appears. Only one *t statement* is meaningful in a section; it can be placed anywhere within that section. If a score section contains no *t statement*, then beats are interpreted as seconds (i.e. with an implicit *t 0 60* statement).

N.B. If the CSound command includes a *-t flag*, the interpreted tempo of all score *t statements* will be overridden by the command-line tempo.

v Statement

v — Provides for locally variable time warping of score events.

Description

The *v statement* provides for locally variable time warping of score events.

Syntax

v p1

Initialization

p1 -- Time warp factor (must be positive).

Performance

The *v statement* takes effect with the following *i statement*, and remains in effect until the next *v statement*, *s statement*, or *e statement*.

Examples

The value of p1 is used as a multiplier for the start times (p2) of subsequent *i statements*.

```
i1  0 1  ; note1
v2
i1  1 1  ; note2
```

In this example, the second note occurs two beats after the first note, and is twice as long.

Although the *v statement* is similar to the *t statement*, the *v statement* is local in operation. That is, *v* affects only the following notes, and its effect may be cancelled or changed by another *v statement*.

Carried values are unaffected by the *v statement* (see *Carry*).

```
i1  0 1  ; note1
v2
i1  1 .  ; note2
i1  2 .  ; note3
v1
i1  3 .  ; note4
i1  4 .  ; note5
e
```

In this example, note3 and note5 occur simultaneously, while note4 actually occurs before note3, that is, at its original place. Durations are unaffected.

```
i1  0 1
v2
i.  + .
i.  . .
```


In this example, the *v statement* has no effect.

x Statement

x — Skip the rest of the current section.

Description

This statement may be used to skip the rest of the current section.

Syntax

x anything

Initialization

All pfields are ignored.

{ Statement

{ — Begins a non-sectional, nestable loop.

Description

The *{* and *}* *statements* can be used to repeat a group of score statements. These loops do not constitute independent score sections and thus may repeat events within the same section. Multiple loops may overlap in time or be nested within each other.

Syntax

```
{ p1 p2
```

Initialization

p1 -- Number of times to repeat the loop.

p2 -- A macro name that is automatically defined at the beginning of the loop and whose value is advanced with each repetition (optional). The initial value is zero and the final value is (*p1* - 1).

Performance

The *{ statement* is used in conjunction with the *}* *statement* to define repeating groups of other score events. A score loop begins with the *{ statement* which defines the number of repetitions and a unique macro name that will contain the current loop counter. The body of a loop can contain any number of other events (including sectional breaks) and is terminated by a *}* *statement* on its own line. The *}* *statement* takes no parameters.

The use of the term "loop" here does not imply any sort of temporal succession to the loop iterations. In other words, the *p2* values of the events inside of the loop are not automatically incremented by the length of the loop in each repetition. This is actually an advantage since it allows groups of simultaneous events to be easily defined as well. The loop macro can be used along with *score expressions* to increase the start times of events or to vary the events in any other way desired for each iteration. The macro is incremented by one for each repetition. Note that unlike the *r statement*, the value of the macro the first time through the loop is zero (0), not one (1). Therefore the final value is one less than the number of repetitions.

Score loops are a very powerful tool. While similar to the section repeat facility (the *r statement*), their chief advantage is that the score events in successive iterations of the loop are not separated by a section termination. Thus, it is possible to create multiple loops that overlap in time. Loops also can be nested within each other to a depth of 39 levels.



Warning

Because of serious problems of interaction with macro expansion, loops must start and end in the same file, and not in a macro.

Examples

Here are some examples of the *{* and *}* *statements*.

Example 605. Sequentially repeat a three-note phrase four times.

```
{ 4 CNT
i1 [0.00 + 0.75 * $CNT.] 0.2 220
i1 [0.25 + 0.75 * $CNT.] . 440
i1 [0.50 + 0.75 * $CNT.] . 880
}
```

is interpreted as

```
i1 0.00 0.2 220
i1 0.25 . 440
i1 0.50 . 880

i1 0.75 0.2 220
i1 1.00 . 440
i1 1.25 . 880

i1 1.50 0.2 220
i1 1.75 . 440
i1 2.00 . 880

i1 2.25 0.2 220
i1 2.50 . 440
i1 2.75 . 880
```

Example 606. Create a group of simultaneous harmonic partials.

In this example, *p4* is assumed to be the frequency of the note event.

```
{ 8 PARTIAL
i1 0 1 [100 * ($PARTIAL. + 1)]
}
```

is interpreted as

```
i1 0 1 100
i1 0 1 200
i1 0 1 300
i1 0 1 400
i1 0 1 500
i1 0 1 600
i1 0 1 700
i1 0 1 800
```

Here is a full example of the `{` and `}` statements. It uses the file *leftbrace.csd* [examples/leftbrace.csd].

Example 607. An example of nested loops to create several inharmonic sine clusters.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o abs.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
nchnls = 2

gaReverbSend init 0
```

```
; a simple sine wave partial
instr 1
  idur = p3
  iamp = p4
  ifreq = p5
  aenv linseg 0.0, 0.1*idur, iamp, 0.6*idur, iamp, 0.3*idur, 0.0
  aosc oscili aenv, ifreq, 1
  vincr gaReverbSend, aosc
endin

; global reverb instrument
instr 2
  al, ar reverbbsc gaReverbSend, gaReverbSend, 0.85, 12000
  outs gaReverbSend+al, gaReverbSend+ar
  clear gaReverbSend
endin

</CsInstruments>
<CsScore>
f1 0 4096 10 1

{ 4 CNT
{ 8 PARTIAL
; start time duration amplitude frequency
i1 [0.5 * $CNT.] [1 + ($CNT * 0.2)] [500 + (~ * 200)] [800 + (200 * $CNT.) + ($PARTIAL. *
}]
}
}

i2 0 6
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: Gabriel Maldonado

New in Csound version 3.52 (?). (Fixed in version 5.08).

} Statement

} — Ends a non-sectional, nestable loop.

Description

The *{* and *}* *statements* can be used to repeat a group of score statements. These loops do not constitute independent score sections and thus may repeat events within the same section. Multiple loops may overlap in time or be nested within each other.

Syntax

}

Initialization

All pfields are ignored.

Performance

The *}* *statement* is used in conjunction with the *{ statement* to define repeating groups of other score events. A score loop begins with the *{ statement* which defines the number of repetitions and a unique macro name that will contain the current loop counter. The body of a loop can contain any number of other events (including sectional breaks) and is terminated by a *}* *statement* on its own line. The *}* *statement* takes no parameters.

See the documentation for the *{ statement* for further details.

Examples

See the examples in the entry for the *{ statement*.

Credits

Author: Gabriel Maldonado

New in Csound version 3.52 (?). (Fixed in version 5.08).

GEN Routines

GEN routines are used as data generators for function tables. When a function table is created using the *f score statement* the GEN function is given as its fourth argument. A negative GEN number implies that the function is not rescaled, and maintains its original values.

Sine/Cosine Generators:

- *GEN09* - Composite waveforms made up of weighted sums of simple sinusoids.
- *GEN10* - Composite waveforms made up of weighted sums of simple sinusoids.
- *GEN11* - Additive set of cosine partials.
- *GEN19* - Composite waveforms made up of weighted sums of simple sinusoids.

- *GEN30* - Generates harmonic partials by analyzing an existing table.
- *GEN33* - Generate composite waveforms by mixing simple sinusoids.
- *GEN34* - Generate composite waveforms by mixing simple sinusoids.

Line/Exponential Segment Generators:

- *GEN05* - Constructs functions from segments of exponential curves.
- *GEN06* - Generates a function comprised of segments of cubic polynomials.
- *GEN07* - Constructs functions from segments of straight lines.
- *GEN08* - Generate a piecewise cubic spline curve.
- *GEN16* - Creates a table from a starting value to an ending value.
- *GEN25* - Construct functions from segments of exponential curves in breakpoint fashion.
- *GEN27* - Construct functions from segments of straight lines in breakpoint fashion.

File Access GEN Routines:

- *GEN01* - Transfers data from a soundfile into a function table.
- *GEN23* - Reads numeric values from a text file.
- *GEN28* - Reads a text file which contains a time-tagged trajectory.

Numeric Value Access GEN Routines

- *GEN02* - Transfers data from immediate pfields into a function table.
- *GEN17* - Creates a step function from given x-y pairs.
- *GEN52* - Creates an interleaved multichannel table from the specified source tables, in the format expected by the *ftconv* opcode.

Window Function GEN Routines

- *GEN20* - Generates functions of different windows.

Random Function GEN Routines

- *GEN21* - Generates tables of different random distributions.
- *GEN40* - Generates a random distribution using a distribution histogram.
- *GEN41* - Generates a random list of numerical pairs.

- *GEN42* - Generates a random distribution of discrete ranges of values.
- *GEN43* - Loads a PVOCEX file containing a PV analysis.

Waveshaping GEN Routines

- *GEN03* - Generates a stored function table by evaluating a polynomial.
- *GEN13* - Stores a polynomial whose coefficients derive from the Chebyshev polynomials of the first kind.
- *GEN14* - Stores a polynomial whose coefficients derive from Chebyshevs of the second kind.
- *GEN15* - Creates two tables of stored polynomial functions.

Amplitude Scaling GEN Routines

- *GEN04* - Generates a normalizing function.
- *GEN12* - Generates the log of a modified Bessel function of the second kind.
- *GEN24* - Reads numeric values from another allocated function-table and rescales them.

Mixing GEN Routines

- *GEN18* - Writes composite waveforms made up of pre-existing waveforms.
- *GEN31* - Mixes any waveform specified in an existing table.
- *GEN32* - Mixes any waveform, resampled with either FFT or linear interpolation.

Pitch and Tuning GEN Routines

- *GEN51* - fills a table with a fully customized micro-tuning scale, in the manner of Csound op-codes *cpstun*, *cpstuni* and *cpstmid*.

Named GEN Routines

Csound's GEN routines can be extended with GEN function plugins. There is currently a simple GEN plugin that provides exponential and hyperbolic tangent functions. These GEN functions are not called by number, but by name.

- "tanh" - fills a table from a hyperbolic tangent formula.
- "exp" - fills a table from a hyperbolic tangent formula.

GEN01

GEN01 — Transfers data from a soundfile into a function table.

Description

This subroutine transfers data from a soundfile into a function table.

Syntax

```
f#  time  size  1  filcod  skiptime  format  channel
```

Performance

size -- number of points in the table. Ordinarily a power of 2 or a power-of-2 plus 1 (see *f statement*); the maximum tablesize is 16777216 (2^{24}) points. The allocation of table memory can be *deferred* by setting this parameter to 0; the size allocated is then the number of points in the file (probably not a power-of-2), and the table is not usable by normal oscillators, but it is usable by a *loscil* unit. The soundfile can also be mono or stereo.

filcod -- integer or character-string denoting the source soundfile name. An integer denotes the file *soundin.filcod* ; a character-string (in double quotes, spaces permitted) gives the filename itself, optionally a full pathname. If not a full path, the file is sought first in the current directory, then in that given by the environment variable SSDIR (if defined) then by SFDIR. See also *soundin*.

skiptime -- begin reading at *skiptime* seconds into the file.

channel -- channel number to read in. 0 denotes read all channels.

format -- specifies the audio data-file format:

1 - 8-bit signed character	4 - 16-bit short integers
2 - 8-bit A-law bytes	5 - 32-bit long integers
3 - 8-bit U-law bytes	6 - 32-bit floats

If *format* = 0 the sample format is taken from the soundfile header, or by default from the CSound -o command-line flag.



Note

- Reading stops at end-of-file or when the table is full. Table locations not filled will contain zeros.
- If p4 is positive, the table will be post-normalized (rescaled to a maximum absolute value of 1 after generation). A negative p4 will cause rescaling to be skipped.

Examples

Here is a simple example of the GEN01 routine. It uses the files *gen01.csd* [examples/gen01.csd], and *beats.wav* [examples/beats.wav]. It uses the audio file “beats.wav”, here is its diagram:



Diagram of the waveform generated by GEN01.

Example 608. A simple example of the GEN01 routine.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o gen01.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kcps = 1
  ifn = 1
  ibas = 1

  ; Play the audio sample stored in Table #1.
  al loscil kamp, kcps, ifn, ibas
  out al
endin

</CsInstruments>
<CsScore>

; Table #1: read an audio file (using GEN01).
f 1 0 131072 1 "beats.wav" 0 4 0

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Here is another example of the GEN01 routine. Csound will automatically compute the tablesizes because we have set it to 0. This example uses the files *gen01computed.csd* [examples/gen01computed.csd], and *beats.wav* [examples/beats.wav].

Example 609. An example of the GEN01 routine with a computed tablesize.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o gen01computed.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

```
; Instrument #1.
instr 1
  kamp = 30000
  kcps = 1
  ifn = 1
  ibas = 1

  ; Play the audio sample stored in Table #1.
  al loscil kamp, kcps, ifn, ibas
  out al
endin

</CsInstruments>
<CsScore>

; Table #1: an audio file (using GEN01).
; Since our table size is 0, Csound will compute it.
f 1 0 0 1 "beats.wav" 0 0 0

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Examples written by Kevin Conder

December 2002. Thanks goes to Kanata Motohashi for fixing mistakes in the examples.

September 2003. Thanks goes to Dr. Richard Boulanger for pointing out the references to the AIFF file format. GEN01 also works with WAV files.

GEN02

GEN02 — Transfers data from immediate pfields into a function table.

Description

This subroutine transfers data from immediate pfields into a function table.

Syntax

```
f # time size 2 v1 v2 v3 ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see *f statement*). The maximum tablesize is 16777216 (2^{24}) points.

v1, *v2*, *v3*, etc. -- values to be copied directly into the table space. The number of values is limited by the compile-time variable *PMAX*, which controls the maximum pfields (currently 1000). The values copied may include the table guard point; any table locations not filled will contain zeros.



Note

If *p4* (the GEN routine number) is positive, the table will be post-normalized (rescaled to a maximum absolute value of 1 after generation). A negative *p4* will cause rescaling to be skipped. You will usually want to use -2 with this GEN function, so that your values are not normalized.

Examples

Here is a simple example of the GEN02 routine. It uses the files *gen02.csd* [examples/gen02.csd]. It places 12 values plus an explicit wrap-around guard value into a table of size next-highest power of 2. Rescaling is inhibited. Here is its diagram:



Diagram of the waveform generated by GEN02.

Example 610. A simple example of the GEN02 routine.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gen02.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
```

```
instr 1
; Create an index over the length of our entire note.
kcps init 1/p3
kndx phasor kcps

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kamp tablei kndx, ifn, ixmode

; Create a sine wave, use the Table #1 values to control
; the amplitude. This creates a sound with a long attack.
al oscil kamp*30000, 440, 2
out al
endin

</CsInstruments>
<CsScore>

; Table #1: an envelope with a long attack (using GEN02).
f 1 0 16 2 0 1 2 3 4 5 6 7 8 9 10 11 0
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN17

Credits

December 2002. Thanks to Rasmus Ekman, corrected the limit of the *PMAX* variable.

GEN03

GEN03 — Generates a stored function table by evaluating a polynomial.

Description

This subroutine generates a stored function table by evaluating a polynomial in x over a fixed interval and with specified coefficients.

Syntax

```
f # time size 3 xval1 xval2 c0 c1 c2 ... cn
```

Initialization

size -- number of points in the table. Must be a power of 2 or a power-of-2 plus 1.

xval1, *xval2* -- left and right values of the x interval over which the polynomial is defined ($xval1 < xval2$). These will produce the 1st stored value and the (power-of-2 plus 1)th stored value respectively in the generated function table.

c0, *c1*, *c2*, ..., *cn* -- coefficients of the n th-order polynomial

$$C_0 + C_1x + C_2x^2 + \dots + C_nx^n$$

Coefficients may be positive or negative real numbers; a zero denotes a missing term in the polynomial. The coefficient list begins in p7, providing a current upper limit of 144 terms.



Note

- The defined segment [$fn(xval1)$, $fn(xval2)$] is evenly distributed. Thus a 512-point table over the interval $[-1,1]$ will have its origin at location 257 (at the start of the 2nd half). Provided the extended guard point is requested, both $fn(-1)$ and $fn(1)$ will exist in the table.
- *GEN03* is useful in conjunction with *table* or *tablei* for audio waveshaping (sound modification by non-linear distortion). Coefficients to produce a particular formant from a sinusoidal lookup index of known amplitude can be determined at preprocessing time using algorithms such as Chebyshev formulae. See also *GEN13*.

Examples

Here is a simple example of the GEN03 routine. It uses the files *gen03.csd* [examples/gen03.csd]. It fills a table with a 4th order polynomial function over the x -interval -1 to 1. The origin will be at the offset position 512. The function is post-normalized. Here is its diagram:



Diagram of the waveform generated by GEN03.

Example 611. A simple example of the GEN03 routine.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o gen03.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create an index over the length of our entire note.
kcps init 1/p3
kndx phasor kcps

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kamp table kndx, ifn, ixmode

; Create a sine wave, use the Table #1 values to control
; the amplitude.
a1 oscil kamp*30000, 440, 2
out a1
endin

</CsInstruments>
<CsScore>

; Table #1: a polynomial function (using GEN03).
f 1 0 1025 3 -1 1 5 4 3 2 2 1
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN13, *GEN14*, and *GEN15*.

GEN04

GEN04 — Generates a normalizing function.

Description

This subroutine generates a normalizing function by examining the contents of an existing table.

Syntax

```
f # time size 4 source# sourcemode
```

Initialization

size -- number of points in the table. Should be power-of-2 plus 1. Must not exceed (except by 1) the size of the source table being examined; limited to just half that size if the sourcemode is of type offset (see below).

source # -- table number of stored function to be examined.

sourcemode -- a coded value, specifying how the source table is to be scanned to obtain the normalizing function. Zero indicates that the source is to be scanned from left to right. Non-zero indicates that the source has a bipolar structure; scanning will begin at the mid-point and progress outwards, looking at pairs of points equidistant from the center.



Note

- The normalizing function derives from the progressive absolute maxima of the source table being scanned. The new table is created left-to-right, with stored values equal to $1/(\text{absolute maximum so far scanned})$. Stored values will thus begin with $1/(\text{first value scanned})$, then get progressively smaller as new maxima are encountered. For a source table which is normalized (values ≤ 1), the derived values will range from $1/(\text{first value scanned})$ down to 1. If the first value scanned is zero, that inverse will be set to 1.
- The normalizing function from *GEN04* is not itself normalized.
- *GEN04* is useful for scaling a table-derived signal so that it has a consistent peak amplitude. A particular application occurs in waveshaping when the carrier (or indexing) signal is less than full amplitude.

Examples

```
f 2 0 512 4 1 1
```

This creates a normalizing function for use in connection with the *GEN03* table 1 example. Mid-point bipolar offset is specified.

GEN05

GEN05 — Constructs functions from segments of exponential curves.

Description

Constructs functions from segments of exponential curves.

Syntax

```
f # time size 5 a n1 b n2 c ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

a, *b*, *c*, etc. -- ordinate values, in odd-numbered pfields p5, p7, p9, . . . These must be nonzero and must be alike in sign.

n1, *n2*, etc. -- length of segment (no. of storage locations), in even-numbered pfields. Cannot be negative, but a zero is meaningful for specifying discontinuous waveforms (e.g. in the example below). The sum $n1 + n2 + \dots$ will normally equal *size* for fully specified functions. If the sum is smaller, the function locations not included will be set to zero; if the sum is greater, only the first *size* locations will be stored.



Note

- If p4 is positive, functions are post-normalized (rescaled to a maximum absolute value of 1 after generation). A negative p4 will cause rescaling to be skipped.
- Discrete-point linear interpolation implies an increase or decrease along a segment by equal differences between adjacent locations; exponential interpolation implies that the progression is by equal ratio. In both forms the interpolation from *a* to *b* is such as to assume that the value *b* will be attained in the $n + 1$ th location. For discontinuous functions, and for the segment encompassing the end location, this value will not actually be reached, although it may eventually appear as a result of final scaling.

Examples

Here is a simple example of the GEN05 routine. It uses the files *gen05.csd* [examples/gen05.csd]. It will create a nice percussive amplitude envelope. Here is its diagram:



Diagram of the waveform generated by GEN05.

Example 612. A simple example of the GEN05 routine.

```
<CsoundSynthesizer>
```

```
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o gen05.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create an index over the length of our entire note.
kcps init 1/p3
kndx phasor kcps

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kamp table kndx, ifn, ixmode

; Create a sine wave, use the Table #1 values to control
; the amplitude. This creates a nice percussive sound.
al oscil kamp*30000, 440, 2
out al
endin

</CsInstruments>
<CsScore>

; Table #1: a percussive envelope (using GEN05).
f 1 0 64 5 1 2 120 60 1 1 0.001 1
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN06, *GEN07*, and *GEN08*

Credits

Example written by Kevin Conder

GEN06

GEN06 — Generates a function comprised of segments of cubic polynomials.

Description

This subroutine will generate a function comprised of segments of cubic polynomials, spanning specified points just three at a time.

Syntax

`f # time size 6 a n1 b n2 c n3 d ...`

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

a, c, e, ... -- local maxima or minima of successive segments, depending on the relation of these points to adjacent inflexions. May be either positive or negative.

b, d, f, ... -- ordinate values of points of inflexion at the ends of successive curved segments. May be positive or negative.

n1, n2, n3 ... -- number of stored values between specified points. Cannot be negative, but a zero is meaningful for specifying discontinuities. The sum $n1 + n2 + \dots$ will normally equal *size* for fully specified functions. (for details, see *GEN05*).



Note

GEN06 constructs a stored function from segments of cubic polynomial functions. Segments link ordinate values in groups of 3: point of inflexion, maximum/minimum, point of inflexion. The first complete segment encompasses *b, c, d* and has length $n2 + n3$, the next encompasses *d, e, f* and has length $n4 + n5$, etc. The first segment (*a, b* with length *n1*) is partial with only one inflexion; the last segment may be partial too. Although the inflexion points *b, d, f ...* each figure in two segments (to the left and right), the slope of the two segments remains independent at that common point (i.e. the 1st derivative will likely be discontinuous). When *a, c, e...* are alternately maximum and minimum, the inflexion joins will be relatively smooth; for successive maxima or successive minima the inflexions will be comb-like.

Examples

Here is a simple example of the GEN06 routine. It uses the files *gen06.csd* [examples/gen06.csd]. It creates a curve running 0 to 1 to -1, with a minimum, maximum and minimum at these values respectively. Inflexions are at .5 and 0 and are relatively smooth. Here is its diagram:



Diagram of the waveform generated by GEN06.

Example 613. A simple example of the GEN06 routine.

<CsoundSynthesizer>

```
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gen06.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create an index over the length of our entire note.
kcps init 1/p3
kndx phasor kcps

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kval table kndx, ifn, ixmode

; Generate a sine waveform, use our Table #1 value to
; vary its frequency by 100 Hz from its base frequency.
ibasefreq = 440
kfreq = kval * 100
al oscil 20000, ibasefreq + kfreq, 2
out al
endin

</CsInstruments>
<CsScore>

; Table #1: a curve (using GEN06).
f 1 0 65 6 0 16 0.5 16 1 16 0 16 -1
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN05, *GEN07*, and *GEN08*

GEN07

GEN07 — Constructs functions from segments of straight lines.

Description

Constructs functions from segments of straight lines.

Syntax

```
f #      time      size  7  a  n1  b  n2  c  ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

a, *b*, *c*, etc. -- ordinate values, in odd-numbered pfields p5, p7, p9, . . .

n1, *n2*, etc. -- length of segment (no. of storage locations), in even-numbered pfields. Cannot be negative, but a zero is meaningful for specifying discontinuous waveforms (e.g. in the example below). The sum $n1 + n2 + \dots$ will normally equal *size* for fully specified functions. If the sum is smaller, the function locations not included will be set to zero; if the sum is greater, only the first *size* locations will be stored.



Note

- If p4 is positive, functions are post-normalized (rescaled to a maximum absolute value of 1 after generation). A negative p4 will cause rescaling to be skipped.
- Discrete-point linear interpolation implies an increase or decrease along a segment by equal differences between adjacent locations; exponential interpolation implies that the progression is by equal ratio. In both forms the interpolation from *a* to *b* is such as to assume that the value *b* will be attained in the $n + 1$ th location. For discontinuous functions, and for the segment encompassing the end location, this value will not actually be reached, although it may eventually appear as a result of final scaling.

Examples

Here is a simple example of the GEN07 routine. It uses the file *gen07.csd* [examples/gen07.csd]. It will create a single-cycle sawtooth whose discontinuity is mid-way in the stored function. Here is its diagram:



Diagram of the waveform generated by GEN07.

Example 614. A simple example of the GEN07 routine.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using com-

mand line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac          -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o gen07.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kcps = 440
  ifn = 1

  ; Play the sine wave stored in Table #1.
  al oscil kamp, kcps, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1: a sawtooth wave (using GEN07).
f 1 0 256 7 0 128 1 0 -1 128 0

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN05, *GEN06*, and *GEN08*

GEN08

GEN08 — Generate a piecewise cubic spline curve.

Description

This subroutine will generate a piecewise cubic spline curve, the smoothest possible through all specified points.

Syntax

```
f # time size 8 a n1 b n2 c n3 d ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

a, *b*, *c*, etc. -- ordinate values of the function.

n1, *n2*, *n3* ... -- length of each segment measured in stored values. May not be zero, but may be fractional. A particular segment may or may not actually store any values; stored values will be generated at integral points from the beginning of the function. The sum $n1 + n2 + \dots$ will normally equal *size* for fully specified functions.



Note

- *GEN08* constructs a stored table from segments of cubic polynomial functions. Each segment runs between two specified points but depends as well on their neighbors on each side. Neighboring segments will agree in both value and slope at their common point. (The common slope is that of a parabola through that point and its two neighbors). The slope at the two ends of the function is constrained to be zero (flat).
- *Hint*: to make a discontinuity in slope or value in the function as stored, arrange a series of points in the interval between two stored values; likewise for a non-zero boundary slope.

Examples

Here is a simple example of the GEN08 routine. It uses the file *gen08.csd* [examples/gen08.csd]. It will create a curve with a smooth hump in the middle, going briefly negative outside the hump then flat at its ends. Here is its diagram:



Diagram of the waveform generated by GEN08.

Example 615. A simple example of the GEN08 routine.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gen08.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create an index over the length of our entire note.
kcps init 1/p3
kndx phasor kcps

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kval table kndx, ifn, ixmode

; Generate a sine waveform, use our Table #1 value to
; vary its frequency by 100 Hz from its base frequency.
ibasefreq = 440
kfreq = kval * 100
al oscil 20000, ibasefreq + kfreq, 2
out al
endin

</CsInstruments>
<CsScore>

; Table #1: a curve with a smooth hump (using GEN08).
f 1 0 65 8 0 16 0 16 1 16 0 16 0
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for two seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN05, *GEN06*, and *GEN07*

GEN09

GEN09 — Generate composite waveforms made up of weighted sums of simple sinusoids.

Description

These subroutines generate composite waveforms made up of weighted sums of simple sinusoids. The specification of each contributing partial requires 3 p-fields using *GEN09*.

Syntax

```
f # time size 9 pna stra phsa pnb strb phsb ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

pna, *pnb*, etc. -- partial no. (relative to a fundamental that would occupy *size* locations per cycle) of sinusoid a, sinusoid b, etc. Must be positive, but need not be a whole number, i.e., non-harmonic partials are permitted. Partial may be in any order.

stra, *strb*, etc. -- strength of partials *pna*, *pnb*, etc. These are relative strengths, since the composite waveform may be rescaled later. Negative values are permitted and imply a 180 degree phase shift.

phsa, *phsb*, etc. -- initial phase of partials *pna*, *pnb*, etc., expressed in degrees (0-360).



Note

- These subroutines generate stored functions as sums of sinusoids of different frequencies. The two major restrictions on *GEN10* that the partials be harmonic and in phase do not apply to *GEN09* or *GEN19*.

In each case the composite wave, once drawn, is then rescaled to unity if p4 was positive. A negative p4 will cause rescaling to be skipped.

Examples

Here is a simple example of the GEN09 routine. It uses the file *gen09.csd* [examples/gen09.csd]. It will generate a cosine wave, a sine wave with an initial phase of 90 degrees. Here is its diagram:



Diagram of the waveform generated by GEN09.

Example 616. A simple example of the GEN09 routine.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>  
<CsoundOptions>
```

```

; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o gen09.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kcps = 440
  ifn = 1

  ; Play the waveform stored in Table #1.
  al oscil kamp, kcps, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1: a cosine wave (using GEN09).
; This is a sine wave with an initial phase of 90 degrees.
f 1 0 16384 9 1 1 90

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>

```

Here is another example of the GEN09 routine. It uses the file *gen09square.csd* [examples/gen09square.csd]. It combines partials 1, 3 and 9 in the relative strengths in which they are found in a square wave, except that partial 9 is upside down. It will be rescaled, here is its diagram:



Diagram of the waveform generated by GEN09.

Example 617. A square wave generated by the GEN09 routine.

```

<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime ouput leave only the line below:
; -o gen09square.wav -W ;;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kcps = 440
  ifn = 1

  ; Play the waveform stored in Table #1.
  al oscil kamp, kcps, ifn
  out al
endin

```

```
</CsInstruments>
<CsScore>

; Table #1: an approximation of a square wave (using GEN09).
f 1 0 16384 9 1 3 0 3 1 0 9 0.3333 180

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN10, GEN19

Credits

The simple example was written by Kevin Conder.

GEN10

GEN10 — Generate composite waveforms made up of weighted sums of simple sinusoids.

Description

These subroutines generate composite waveforms made up of weighted sums of simple sinusoids. The specification of each contributing partial requires 1 pfield using *GEN10*.

Syntax

```
f # time size 10 str1 str2 str3 str4 ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

str1, *str2*, *str3*, etc. -- relative strengths of the fixed harmonic partial numbers 1,2,3, etc., beginning in p5. Partial not required should be given a strength of zero.



Note

- These subroutines generate stored functions as sums of sinusoids of different frequencies. The two major restrictions on *GEN10* that the partials be harmonic and in phase do not apply to *GEN09* or *GEN19*.

In each case the composite wave, once drawn, is then rescaled to unity if p4 was positive. A negative p4 will cause rescaling to be skipped.

Examples

Here is a simple example of the GEN10 routine. It uses the file *gen10.csd* [examples/gen10.csd]. It will generate a simple sine wave. Here is its diagram:



Diagram of the waveform generated by GEN10.

Example 618. A simple example of the GEN10 routine.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gen10.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>
```

```
; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kcps = 440
  ifn = 1

  ; Play the sine wave stored in Table #1.
  a1 oscil kamp, kcps, ifn
  out a1
endin

</CsInstruments>
<CsScore>

; Table #1: a simple sine wave (using GEN10).
f 1 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN09, *GEN11*, and *GEN19*.

Credits

Example written by Kevin Conder

GEN11

GEN11 — Generates an additive set of cosine partials.

Description

This subroutine generates an additive set of cosine partials, in the manner of Csound generators *buzz* and *gbuzz*.

Syntax

```
f # time size ll nh [lh] [r]
```

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

nh -- number of harmonics requested. Must be positive.

lh(optional) -- lowest harmonic partial present. Can be positive, zero or negative. The set of partials can begin at any partial number and proceeds upwards; if *lh* is negative, all partials below zero will reflect in zero to produce positive partials without phase change (since cosine is an even function), and will add constructively to any positive partials in the set. The default value is 1

r(optional) -- multiplier in an amplitude coefficient series. This is a power series: if the *lh*th partial has a strength coefficient of *A* the (*lh* + *n*)th partial will have a coefficient of $A * r^n$, i.e. strength values trace an exponential curve. *r* may be positive, zero or negative, and is not restricted to integers. The default value is 1.



Note

- This subroutine is a non-time-varying version of the CSound *buzz* and *gbuzz* generators, and is similarly useful as a complex sound source in subtractive synthesis. With *lh* and *r* present it parallels *gbuzz*; with both absent or equal to 1 it reduces to the simpler *buzz* (i.e. *nh* equal-strength harmonic partials beginning with the fundamental).
- Sampling the stored waveform with an oscillator is more efficient than using the dynamic buzz units. However, the spectral content is invariant and care is necessary, lest the higher partials exceed the Nyquist during sampling to produce fold-over.

Examples

Here is a simple example of the GEN11 routine. It uses the file *gen11.csd* [examples/gen11.csd]. It will generate a simple cosine wave. Here is its diagram:



Diagram of the waveform generated by GEN11.

Example 619. A simple example of the GEN11 routine.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out  Audio in
-odac      -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o genll.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
  kamp = 30000
  kcps = 440
  ifn = 1

  ; Play the cosine wave stored in Table #1.
  al oscil kamp, kcps, ifn
  out al
endin

</CsInstruments>
<CsScore>

; Table #1: a simple cosine wave (using GEN11).
f 1 0 16384 11 1 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN10

Credits

Example written by Kevin Conder

GEN12

GEN12 — Generates the log of a modified Bessel function of the second kind.

Description

This generates the log of a modified Bessel function of the second kind, order 0, suitable for use in amplitude-modulated FM.

Syntax

```
f # time size 12 xint
```

Initialization

size -- number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see *f statement*). The normal value is power-of-2 plus 1.

xint -- specifies the *x* interval [0 to +*xint*] over which the function is defined.



Note

- This subroutine draws the natural log of a modified Bessel function of the second kind, order 0 (commonly written as I subscript 0), over the *x*-interval requested. The call should have rescaling inhibited.
- The function is useful as an amplitude scaling factor in cycle-synchronous amplitude-modulated FM. (See Palamin & Palamin, *J. Audio Eng. Soc.*, 36/9, Sept. 1988, pp.671-684.) The algorithm is interesting because it permits the normally symmetric FM spectrum to be made asymmetric around a frequency other than the carrier, and is thereby useful for formant positioning. By using a table lookup index of $I(r - 1/r)$, where I is the FM modulation index and r is an exponential parameter affecting partial strengths, the Palamin algorithm becomes relatively efficient, requiring only oscil's, table lookups, and a single *exp* call.

Examples

Here is a simple example of the GEN12 routine. It uses the file *gen12.csd* [examples/gen12.csd]. It generates the function $\ln(I_0(x))$ from 0 to 20. Here is its diagram:



Diagram of the waveform generated by GEN12.

Example 620. A simple example of the GEN12 routine.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
```



```
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gen12.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create an index over the length of our entire note.
kcps init 1/p3
kndx phasor kcps

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kamp tablei kndx, ifn, ixmode

; Create a sine wave, use the Table #1 values to control
; the amplitude. This creates a sound with a long attack.
a1 oscil kamp*30000, 440, 2
out a1
endin

</CsInstruments>
<CsScore>

; Table #1: a modified Bessel function (using GEN12).
f 1 0 2049 12 20
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

Credits

Example written by Kevin Conder

GEN13

GEN13 — Stores a polynomial whose coefficients derive from the Chebyshev polynomials of the first kind.

Description

Uses Chebyshev coefficients to generate stored polynomial functions which, under waveshaping, can be used to split a sinusoid into harmonic partials having a pre-definable spectrum.

Syntax

```
f # time size 13 xint xamp h0 h1 h2 ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see *f statement*). The normal value is power-of-2 plus 1.

xint -- provides the left and right values $[-xint, +xint]$ of the *x* interval over which the polynomial is to be drawn. These subroutines both call *GEN03* to draw their functions; the *p5* value here is therefor expanded to a negative-positive *p5*, *p6* pair before *GEN03* is actually called. The normal value is 1.

xamp -- amplitude scaling factor of the sinusoid input that is expected to produce the following spectrum.

h0, *h1*, *h2*, etc. -- relative strength of partials 0 (DC), 1 (fundamental), 2 ... that will result when a sinusoid of amplitude

$xamp * \text{int}(\text{size}/2)/xint$

is waveshaped using this function table. These values thus describe a frequency spectrum associated with a particular factor *xamp* of the input signal.

GEN13 is the function generator normally employed in standard waveshaping. It stores a polynomial whose coefficients derive from the Chebyshev polynomials of the first kind, so that a driving sinusoid of strength *xamp* will exhibit the specified spectrum at output. Note that the evolution of this spectrum is generally not linear with varying *xamp*. However, it is bandlimited (the only partials to appear will be those specified at generation time); and the partials will tend to occur and to develop in ascending order (the lower partials dominating at low *xamp*, and the spectral richness increasing for higher values of *xamp*). A negative *hn* value implies a 180 degree phase shift of that partial; the requested full-amplitude spectrum will not be affected by this shift, although the evolution of several of its component partials may be. The pattern $+,+,-,-,+,+,\dots$ for *h0*, *h1*, *h2*... will minimize the normalization problem for low *xamp* values (see above), but does not necessarily provide the smoothest pattern of evolution.

Examples

Here is a simple example of the GEN13 routine. It uses the file *gen13.csd* [examples/gen13.csd]. It creates a function which, under waveshaping, will split a sinusoid into 3 odd-harmonic partials of relative strength 5:3:1. Here is its diagram:



Diagram of the waveform generated by GEN13.

Example 621. A simple example of the GEN13 routine.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gen13.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create an index over the length of our entire note.
kcps init 1/p3
kndx phasor kcps

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kval table kndx, ifn, ixmode

; Generate a sine waveform, use our Table #1 value to
; vary its frequency by 100 Hz from its base frequency.
ibasefreq = 440
kfreq = kval * 100
a1 oscil 20000, ibasefreq + kfreq, 2
out a1
endin

</CsInstruments>
<CsScore>

; Table #1: a polynomial function (using GEN13).
f 1 0 1025 13 1 1 0 5 0 3 0 1
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN03, GEN14, and GEN15.

GEN14

GEN14 — Stores a polynomial whose coefficients derive from Chebyshevs of the second kind.

Description

Uses Chebyshev coefficients to generate stored polynomial functions which, under waveshaping, can be used to split a sinusoid into harmonic partials having a pre-definable spectrum.

Syntax

```
f # time size 14 xint xamp h0 h1 h2 ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see *f statement*). The normal value is power-of-2 plus 1.

xint -- provides the left and right values $[-xint, +xint]$ of the x interval over which the polynomial is to be drawn. These subroutines both call *GEN03* to draw their functions; the p5 value here is therefore expanded to a negative-positive p5, p6 pair before *GEN03* is actually called. The normal value is 1.

xamp -- amplitude scaling factor of the sinusoid input that is expected to produce the following spectrum.

h0, *h1*, *h2*, etc. -- relative strength of partials 0 (DC), 1 (fundamental), 2 ... that will result when a sinusoid of amplitude

$xamp * \text{int}(\text{size}/2)/xint$

is waveshaped using this function table. These values thus describe a frequency spectrum associated with a particular factor *xamp* of the input signal.



Note

- *GEN13* is the function generator normally employed in standard waveshaping. It stores a polynomial whose coefficients derive from the Chebyshev polynomials of the first kind, so that a driving sinusoid of strength *xamp* will exhibit the specified spectrum at output. Note that the evolution of this spectrum is generally not linear with varying *xamp*. However, it is bandlimited (the only partials to appear will be those specified at generation time); and the partials will tend to occur and to develop in ascending order (the lower partials dominating at low *xamp*, and the spectral richness increasing for higher values of *xamp*). A negative *hn* value implies a 180 degree phase shift of that partial; the requested full-amplitude spectrum will not be affected by this shift, although the evolution of several of its component partials may be. The pattern +, +, -, -, +, +, ... for *h0, h1, h2...* will minimize the normalization problem for low *xamp* values (see above), but does not necessarily provide the smoothest pattern of evolution.
- *GEN14* stores a polynomial whose coefficients derive from Chebyshevs of the second kind.

Examples

Here is a simple example of the GEN14 routine. It uses the file *gen14.csd* [examples/gen14.csd]. It creates a function which, under waveshaping, will split a sinusoid into 3 odd-harmonic partials of relative strength 5:3:1. Here is its diagram:



Diagram of the waveform generated by GEN14.

Example 622. A simple example of the GEN14 routine.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out      Audio in
-odac             -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gen14.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create an index over the length of our entire note.
kcps init 1/p3
kndx phasor kcps

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kval table kndx, ifn, ixmode

; Generate a sine waveform, use our Table #1 value to
; vary its frequency by 100 Hz from its base frequency.
ibasefreq = 440
kfreq = kval * 100
a1 oscil 20000, ibasefreq + kfreq, 2
out a1
endin

</CsInstruments>
<CsScore>

; Table #1: a polynomial function (using GEN14).
f 1 0 1025 14 1 1 0 5 0 3 0 1
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 2 seconds.
i 1 0 2
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN03, *GEN13*, and *GEN15*.

Credits

Example written by Kevin Conder

GEN15

GEN15 — Creates two tables of stored polynomial functions.

Description

This subroutine creates two tables of stored polynomial functions, suitable for use in phase quadrature operations.

Syntax

```
f # time size 15 xint xamp h0 phs0 h1 phs1 h2 phs2 ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see *f statement*). The normal value is power-of-2 plus 1.

xint -- provides the left and right values $[-xint, +xint]$ of the x interval over which the polynomial is to be drawn. This subroutine will eventually call *GEN03* to draw both functions; this *p5* value is therefor expanded to a negative-positive *p5*, *p6* pair before *GEN03* is actually called. The normal value is 1.

xamp -- amplitude scaling factor of the sinusoid input that is expected to produce the following spectrum.

h0, *h1*, *h2*, ... *hn* -- relative strength of partials 0 (DC), 1 (fundamental), 2 ... that will result when a sinusoid of amplitude

$xamp * \text{int}(\text{size}/2)/xint$

is waveshaped using this function table. These values thus describe a frequency spectrum associated with a particular factor *xamp* of the input signal.

phs0, *phs1*, ... -- phase in degrees of desired harmonics *h0*, *h1*, ... when the two functions of *GEN15* are used with phase quadrature.



Note

GEN15 creates two tables of equal size, labeled *f #* and *f # + 1*. Table *#* will contain a Chebyshev function of the first kind, drawn using *GEN03* with partial strengths *h0cos(phs0)*, *h1cos(phs1)*, ... Table *#+1* will contain a Chebyshev function of the 2nd kind by calling *GEN14* with partials *h1sin(phs1)*, *h2sin(phs2)*,... (note the harmonic displacement). The two tables can be used in conjunction in a waveshaping network that exploits phase quadrature.

See Also

GEN03, *GEN13*, and *GEN14*.

GEN16

GEN16 — Creates a table from a starting value to an ending value.

Description

Creates a table from *beg* value to *end* value of *dur* steps.

Syntax

```
f # time size 16 beg dur type end
```

Initialization

size -- number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see *f statement*). The normal value is power-of-2 plus 1.

beg -- starting value

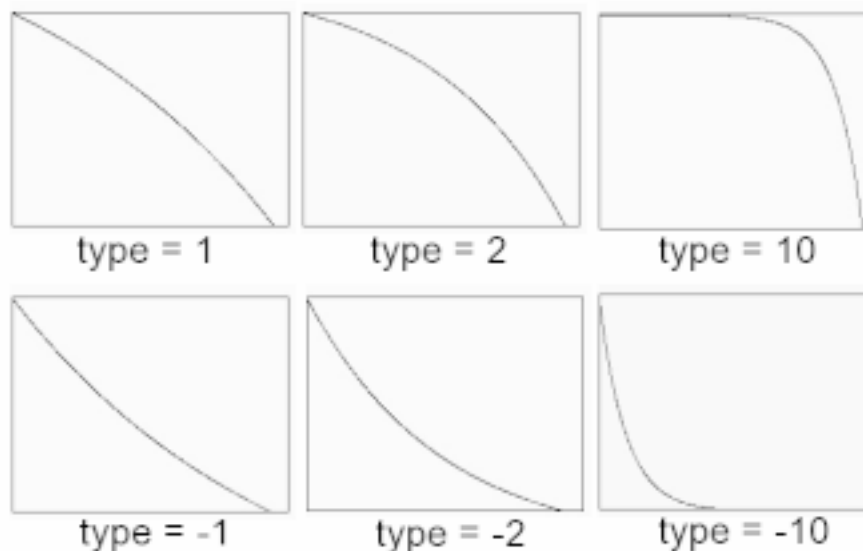
dur -- number of segments

type -- if 0, a straight line is produced. If non-zero, then *GEN16* creates the following curve, for *dur* steps:

$$\text{beg} + (\text{end} - \text{beg}) * (1 - \exp(i * \text{type} / (\text{dur} - 1))) / (1 - \exp(\text{type}))$$

end -- value after *dur* segments

Here are some examples of the curves generated for different values of *type*:



Tables generated by GEN16 for different values of *type*.



Note

If *type* > 0, there is a slowly rising (concave) or slowly decaying (convex) curve, while if *itype* < 0, the curve is fast rising (convex) or fast decaying (concave). See also *transeg*.

Example 623. A simple example of the GEN16 routine.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gen16.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

sr = 44100
ksmps = 128
nchnls = 1

instr 1
  kcps init 1/p3
  kndx phasor kcps

  ifn = p4
  ixmode = 1
  kval table kndx, ifn, ixmode

  ibasefreq = 440
  kfreq = kval * ibasefreq
  al oscil 20000, ibasefreq + kfreq, 1
  out al
endin

</CsInstruments>
<CsScore>

f 1 0 16384 10 1

f 2 0 1024 16 1 1024 1 0
f 3 0 1024 16 1 1024 2 0
f 4 0 1024 16 1 1024 10 0
f 5 0 1024 16 1 1024 -1 0
f 6 0 1024 16 1 1024 -2 0
f 7 0 1024 16 1 1024 -10 0

i 1 0 2 2
i 1 + . 3
i 1 + . 4
i 1 + . 5
i 1 + . 6
i 1 + . 7

e

</CsScore>
</CsoundSynthesizer>
```

Credits

Author: John ffitch
University of Bath, Codemist. Ltd.
Bath, UK
October, 2000

New in Csound version 4.09

GEN17

GEN17 — Creates a step function from given x-y pairs.

Description

This subroutine creates a step function from given x-y pairs.

Syntax

```
f # time size 17 x1 a x2 b x3 c ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see *f statement*). The normal value is power-of-2 plus 1.

x1, *x2*, *x3*, etc. -- x-ordinate values, in ascending order, 0 first.

a, *b*, *c*, etc. -- y-values at those x-ordinates, held until the next x-ordinate.



Note

This subroutine creates a step function of x-y pairs whose y-values are held to the right. The right-most y-value is then held to the end of the table. The function is useful for mapping one set of data values onto another, such as MIDI note numbers onto sampled sound ftable numbers (see *loscil*).

Examples

```
f 1 0 128 -17 0 1 12 2 24 3 36 4 48 5 60 6 72 7 84 8
```

This describes a step function with 8 successively increasing levels, each 12 locations wide except the last which extends its value to the end of the table. Rescaling is inhibited. Indexing into this table with a MIDI note-number would retrieve a different value every octave up to the eighth, above which the value returned would remain the same.

See Also

GEN02

GEN18

GEN18 — Writes composite waveforms made up of pre-existing waveforms.

Description

Writes composite waveforms made up of pre-existing waveforms. Each contributing waveform requires 4 pfields and can overlap with other waveforms.

Syntax

```
f # time size 18 fna ampa starta finisha fnb ampb startb finishb ...
```

Initialization

size -- number of points in the table. Must be a power-of-2 plus 1 (see f statement).

fna, fnb, etc. -- pre-existing table number to be written into the table.

ampa, ampb, etc. -- strength of wavefoms. These are relative strengths, since the composite waveform may be rescaled later. Negative values are permitted and imply a 180 degree phase shift.

starta, startb, etc. -- where to start writing the fn into the table.

finisha, finishb, etc. -- where to stop writing the fn into the table.

Examples

```
f 1 0 4096 10 1  
f 2 0 1025 18 1 1 0 512 1 1 513 1025
```

f2 consists of two copies of f1 written in to locations 0-512 and 513-1025.

Deprecated Names

GEN18 was called *GEN22* in version 4.18. The name was changed due to a conflict with DirectC-sound.

Credits

Author: William “Pete” Moss
University of Texas at Austin
Austin, Texas USA
January 2002

New in version 4.18, changed in version 4.19

GEN19

GEN19 — Generate composite waveforms made up of weighted sums of simple sinusoids.

Description

These subroutines generate composite waveforms made up of weighted sums of simple sinusoids. The specification of each contributing partial requires 4 p-fields using *GEN19*.

Syntax

```
f # time size 19 pna stra phsa dcoa pnb strb phsb dcob ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

pna, *pnb*, etc. -- partial no. (relative to a fundamental that would occupy *size* locations per cycle) of sinusoid a, sinusoid b, etc. Must be positive, but need not be a whole number, i.e., non-harmonic partials are permitted. Partial may be in any order.

stra, *strb*, etc. -- strength of partials *pna*, *pnb*, etc. These are relative strengths, since the composite waveform may be rescaled later. Negative values are permitted and imply a 180 degree phase shift.

phsa, *phsb*, etc. -- initial phase of partials *pna*, *pnb*, etc., expressed in degrees.

dcoa, *dcob*, etc. -- DC offset of partials *pna*, *pnb*, etc. This is applied *after* strength scaling, i.e. a value of 2 will lift a 2-strength sinusoid from range [-2,2] to range [0,4] (before later rescaling).



Note

- These subroutines generate stored functions as sums of sinusoids of different frequencies. The two major restrictions on *GEN10* that the partials be harmonic and in phase do not apply to *GEN09* or *GEN19*.

In each case the composite wave, once drawn, is then rescaled to unity if p4 was positive. A negative p4 will cause rescaling to be skipped.

Examples

Here is a simple example of the GEN19 routine. It uses the file *gen19.csd* [examples/gen19.csd]. It will generate a nice bell curve, here is its diagram:



Diagram of the waveform generated by GEN19.

Example 624. A simple example of the GEN19 routine.

See the sections *Real-time Audio* and *Command Line Flags* for more information on using command line flags.

```
<CsoundSynthesizer>
<CsOptions>
; Select audio/midi flags here according to platform
; Audio out   Audio in
-odac         -iadc      ;;RT audio I/O
; For Non-realtime output leave only the line below:
; -o gen19.wav -W ;; for file output any platform
</CsOptions>
<CsInstruments>

; Initialize the global variables.
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

; Instrument #1.
instr 1
; Create an index over the length of our entire note.
kcps init 1/p3
kndx phasor kcps

; Read Table #1 with our index.
ifn = 1
ixmode = 1
kval table kndx, ifn, ixmode

; Generate a sine waveform, use our Table #1 value to
; vary its frequency by 100 Hz from its base frequency.
ibasefreq = 440
kfreq = kval * 100
a1 oscil 20000, ibasefreq + kfreq, 2
out a1
endin

</CsInstruments>
<CsScore>

; Table #1: a bell curve (using GEN19).
f 1 0 16384 -19 1 1 260 1
; Table #2, a sine wave.
f 2 0 16384 10 1

; Play Instrument #1 for 3 seconds.
i 1 0 3
e

</CsScore>
</CsoundSynthesizer>
```

See Also

GEN09 and *GEN10*

Credits

Example written by Kevin Conder

GEN20

GEN20 — Generates functions of different windows.

Description

This subroutine generates functions of different windows. These windows are usually used for spectrum analysis or for grain envelopes.

Syntax

```
f # time size 20 window max [opt]
```

Initialization

size -- number of points in the table. Must be a power of 2 (+ 1).

window -- Type of window to generate:

- 1 = Hamming
- 2 = Hanning
- 3 = Bartlett (triangle)
- 4 = Blackman (3-term)
- 5 = Blackman - Harris (4-term)
- 6 = Gaussian
- 7 = Kaiser
- 8 = Rectangle
- 9 = Sync

max -- For negative p4 this will be the absolute value at window peak point. If p4 is positive or p4 is negative and p6 is missing the table will be post-rescaled to a maximum value of 1.

opt -- Optional argument required by the Gaussian window and the Kaiser window.

Examples

```
f      1      0      1024      20      5
```

This creates a function which contains a 4 - term Blackman - Harris window with maximum value of 1.

```
f      1      0      1024     -20      2      456
```

This creates a function that contains a Hanning window with a maximum value of 456.

```
f      1      0      1024      -20      1
```

This creates a function that contains a Hamming window with a maximum value of 1.

```
f      1      0      1024      20      7      1      2
```

This creates a function that contains a Kaiser window with a maximum value of 1. The extra argument specifies how "open" the window is, for example a value of 0 results in a rectangular window and a value of 10 in a Hamming like window.

```
f      1      0      1024      20      6      1      2
```

This creates a function that contains a Gaussian window with a maximum value of 1. The extra argument specifies how broad the window is, as the standard deviation of the curve; in this example the s.d. is 2. The default value is 1.

For diagrams, see *Window Functions*

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK

New in Csound version 3.2

Optional argument to Gaussian added in 5.10

GEN21

GEN21 — Generates tables of different random distributions.

Description

This generates tables of different random distributions. (See also *betarand*, *bexprnd*, *cauchy*, *exprand*, *gauss*, *linrand*, *pcauchy*, *poisson*, *trirand*, *unirand*, and *weibull*)

Syntax

```
f # time size 21 type level [arg1 [arg2]]
```

Initialization

time and *size* are the usual GEN function arguments. *level* defines the amplitude. Note that GEN21 is not self-normalizing as are most other GEN functions. *type* defines the distribution to be used as follow:

- 1 = Uniform (positive numbers only)
- 2 = Linear (positive numbers only)
- 3 = Triangular (positive and negative numbers)
- 4 = Exponential (positive numbers only)
- 5 = Biexponential (positive and negative numbers)
- 6 = Gaussian (positive and negative numbers)
- 7 = Cauchy (positive and negative numbers)
- 8 = Positive Cauchy (positive numbers only)
- 9 = Beta (positive numbers only)
- 10 = Weibull (positive numbers only)
- 11 = Poisson (positive numbers only)

Of all these cases only 9 (Beta) and 10 (Weibull) need extra arguments. Beta needs two arguments and Weibull one.

Examples

```
f1 0 1024 21 1 ; Uniform (white noise)
f1 0 1024 21 6 ; Gaussian
f1 0 1024 21 9 1 1 2 ; Beta (note that level precedes arguments)
f1 0 1024 21 10 1 2 ; Weibull
```

All of the above additions were designed by the author between May and December 1994, under the supervision of Dr. Richard Boulanger.

Credits

Author: Paris Smaragdis
MIT, Cambridge
1995

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK

New in Csound version 3.2

GEN22

GEN22 — Deprecated.

Description

Deprecated as of version 4.19. Use the *GENI8* routine instead.

GEN23

GEN23 — Reads numeric values from a text file.

Description

This subroutine reads numeric values from an external ASCII file.

Syntax

```
f # time size -23 "filename.txt"
```

Initialization

"filename.txt" -- numeric values contained in "filename.txt" (which indicates the complete pathname of the character file to be read), can be separated by spaces, tabs, newline characters or commas. Also, words that contains non-numeric characters can be used as comments since they are ignored.

size -- number of points in the table. Must be a power of 2 , power of 2 + 1, or zero. If *size* = 0, table size is determined by the number of numeric values in *filename.txt*. (New in Csound version 3.57)



Note

All characters following ';' (comment) are ignored until next line (numbers too).

Credits

Author: Gabriel Maldonado
Italy
February, 1998

New in Csound version 3.47

GEN24

GEN24 — Reads numeric values from another allocated function-table and rescales them.

Description

This subroutine reads numeric values from another allocated function-table and rescales them according to the max and min values given by the user.

Syntax

```
f # time size -24 ftable min max
```

Initialization

#, time, size -- the usual GEN parameters. See *f* statement.

ftable -- *ftable* must be an already allocated table with the same size as this function.

min, max -- the rescaling range.



Note

This GEN is useful, for example, to eliminate the starting offset in exponential segments allowing a real starting from zero.

Credits

Author: Gabriel Maldonado

New in Csound version 4.16

GEN25

GEN25 — Construct functions from segments of exponential curves in breakpoint fashion.

Description

These subroutines are used to construct functions from segments of exponential curves in breakpoint fashion.

Syntax

```
f # time size 25 x1 y1 x2 y2 x3 ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

x1, *x2*, *x3*, etc. -- locations in table at which to attain the following *y* value. Must be in increasing order. If the last value is less than *size*, then the rest will be set to zero. Should not be negative but can be zero.

y1, *y2*, *y3*, etc. -- Breakpoint values attained at the location specified by the preceding *x* value. These must be non-zero and must be alike in sign.



Note

If *p4* is positive, functions are post-normalized (rescaled to a maximum absolute value of 1 after generation). A negative *p4* will cause rescaling to be skipped.

See Also

f statement, *GEN27*

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK

New in Csound version 3.49

GEN27

GEN27 — Construct functions from segments of straight lines in breakpoint fashion.

Description

Construct functions from segments of straight lines in breakpoint fashion.

Syntax

```
f # time size 27 x1 y1 x2 y2 x3 ...
```

Initialization

size -- number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see *f statement*).

x1, *x2*, *x3*, etc. -- locations in table at which to attain the following *y* value. Must be in increasing order. If the last value is less than *size*, then the rest will be set to zero. Should not be negative but can be zero.

y1, *y2*, *y3*, etc. -- Breakpoint values attained at the location specified by the preceding *x* value.



Note

If *p4* is positive, functions are post-normalized (rescaled to a maximum absolute value of 1 after generation). A negative *p4* will cause rescaling to be skipped.

Examples

```
f 1 0 257 27 0 0 100 1 200 -1 256 0
```

This describes a function which begins at 0, rises to 1 at the 100th table location, falls to -1, by the 200th location, and returns to 0 by the end of the table. The interpolation is linear.

See Also

f statement, *GEN25*

Credits

Author: John ffitch
University of Bath/Codemist Ltd.
Bath, UK

New in Csound version 3.49

GEN28

GEN28 — Reads a text file which contains a time-tagged trajectory.

Description

This function generator reads a text file which contains sets of three values representing the xy coordinates and a time-tag for when the signal should be placed at that location, allowing the user to define a time-tagged trajectory. The file format is in the form:

```
time1  X1  Y1
time2  X2  Y2
time3  X3  Y3
```

The configuration of the xy coordinates in space places the signal in the following way:

- a1 is -1, 1
- a2 is 1, 1
- a3 is -1, -1
- a4 is 1, -1

This assumes a loudspeaker set up as a1 is left front, a2 is right front, a3 is left back, a4 is right back. Values greater than 1 will result in sounds being attenuated as if in the distance. *GEN28* creates values to 10 milliseconds of resolution.

Syntax

```
f # time size 28 ifilcod
```

Initialization

size -- number of points in the table. Must be 0. *GEN28* takes 0 as the size and automatically allocates memory.

ifilcod -- character-string denoting the source file name. A character-string (in double quotes, spaces permitted) gives the filename itself, optionally a full pathname. If not a full path, the named file is sought in the current directory.

Examples

```
f1 0 0 28 "move"
```

The file "move" should look like:

```
0  -1  1
1   1  1
```

2	4	4
2.1	-4	-4
3	10	-10
5	-40	0

Since *GEN28* creates values to 10 milliseconds of resolution, there will be 500 values created by interpolating X1 to X2 to X3 and so on, and Y1 to Y2 to Y3 and so on, over the appropriate number of values that are stored in the function table. The sound will begin in the left front, over 1 second it will move to the right front, over another second it move further into the distance but still in the right front, then in just 1/10th of a second it moves to the left rear, a bit distant. Finally over the last .9 seconds the sound will move to the right rear, moderately distant, and it comes to rest between the two left channels (due west!), quite distant.

Credits

Author: Richard Karpen
Seattle, Wash
1998

New in Csound version 3.48

GEN30

GEN30 — Generates harmonic partials by analyzing an existing table.

Description

Extracts a range of harmonic partials from an existing waveform.

Syntax

```
f # time size 30 src minh maxh [ref_sr] [interp]
```

Performance

src -- source ftable

minh -- lowest harmonic number

maxh -- highest harmonic number

ref_sr (optional) -- *maxh* is scaled by (*sr* / *ref_sr*). The default value of *ref_sr* is *sr*. If *ref_sr* is zero or negative, it is now ignored.

interp (optional) -- if non-zero, allows changing the amplitude of the lowest and highest harmonic partial depending on the fractional part of *minh* and *maxh*. For example, if *maxh* is 11.3 then the 12th harmonic partial is added with 0.3 amplitude. This parameter is zero by default.

GEN30 does not support tables with an extended guard point (ie. table size = power of two + 1). Although such tables will work both for input and output, when reading source table(s), the guard point is ignored, and when writing the output table, guard point is simply copied from the first sample (table index = 0).

The reason of this limitation is that *GEN30* uses FFT, which requires power of two table size. *GEN32* allows using linear interpolation for resampling and phase shifting, which makes it possible to use any table size (however, for partials calculated with FFT, the power of two limitation still exists).

Credits

Author: Istvan Varga

New in version 4.16

GEN31

GEN31 — Mixes any waveform specified in an existing table.

Description

This routine is similar to GEN09, but allows mixing any waveform specified in an existing table.

Syntax

```
f # time size 31 src pna stra phsa pnb strb phsb ...
```

Performance

src -- source table number

pna, *pnb*, ... -- partial number, must be a positive integer

stra, *strb*, ... -- amplitude scale

phsa, *phsb*, ... -- start phase (0 to 1)

GEN31 does not support tables with an extended guard point (ie. table size = power of two + 1). Although such tables will work both for input and output, when reading source table(s), the guard point is ignored, and when writing the output table, guard point is simply copied from the first sample (table index = 0).

The reason of this limitation is that *GEN31* uses FFT, which requires power of two table size. *GEN32* allows using linear interpolation for resampling and phase shifting, which makes it possible to use any table size (however, for partials calculated with FFT, the power of two limitation still exists).

Credits

Author: Istvan Varga

New in version 4.15

GEN32

GEN32 — Mixes any waveform, resampled with either FFT or linear interpolation.

Description

This routine is similar to *GEN31*, but allows specifying source ftable for each partial. Tables can be resampled either with FFT, or linear interpolation.

Syntax

```
f # time size 32 srca pna stra phsa srcb pnb strb phsb ...
```

Performance

srca, *srcb* -- source table number. A negative value can be used to read the table with linear interpolation (by default, the source waveform is transposed and phase shifted using FFT); this is less accurate, but faster, and allows non-integer and negative partial numbers.

pna, *pnb*, ... -- partial number, must be a positive integer if source table number is positive (i.e. resample with FFT).

stra, *strb*, ... -- amplitude scale

phsa, *phsb*, ... -- start phase (0 to 1)

Examples

```
itmp    ftgen 1, 0, 16384, 7, 1, 16384, -1      ; sawtooth
itmp    ftgen 2, 0, 8192, 10, 1                ; sine
; mix tables
itmp    ftgen 5, 0, 4096, -32, -2, 1.5, 1.0, 0.25, 1, 2, 0.5, 0, \
        1, 3, -0.25, 0.5
; window
itmp    ftgen 6, 0, 16384, 20, 3, 1
; generate band-limited waveforms
inote   = 0
loop0:
icps    = 440 * exp(log(2) * (inote - 69) / 12)      ; one table for
inumh   = sr / (2 * icps)                             ; each MIDI note number
ift     = int(inote + 256.5)
itmp    ftgen ift, 0, 4096, -30, 5, 1, inumh
inote   = inote + 1
if (inote < 127.5) igoto loop0

instr 1

kcps    expon 20, p3, 16000
kft     = int(256.5 + 69 + 12 * log(kcps / 440) / log(2))
kft     = (kft > 383 ? 383 : kft)

a1      phasor kcps
a1      tableikt a1, kft, 1, 0, 1

out a1 * 10000

endin
instr 2

kcps    expon 20, p3, 16000
kft     = int(256.5 + 69 + 12 * log(kcps / 440) / log(2))
kft     = (kft > 383 ? 383 : kft)

kgdur   limit 10 / kcps, 0.1, 1
a1      grain2 kcps, 0.02, kgdur, 30, kft, 6, -0.5
```

```
      out a1 * 2000
    endin

-----
score:
-----

t 0 60
i 1 0 10
i 2 12 10
e
```

Credits

Author: Rasmus Ekman

Programmer: Istvan Varga

New in version 4.17

GEN33

GEN33 — Generate composite waveforms by mixing simple sinusoids.

Description

These routines generate composite waveforms by mixing simple sinusoids, similarly to *GEN09*, but the parameters of the partials are specified in an already existing table, which makes it possible to calculate any number of partials in the orchestra.

The difference between *GEN33* and *GEN34* is that *GEN33* uses inverse FFT to generate output, while *GEN34* is based on the algorithm used in *oscils* opcode. *GEN33* allows integer partials only, and does not support power of two plus 1 table size, but may be significantly faster with a large number of partials. On the other hand, with *GEN34*, it is possible to use non-integer partial numbers and extended guard point, and this routine may be faster if there is only a small number of partials (note that *GEN34* is also several times faster than *GEN09*, although the latter may be more accurate).

Syntax

```
f # time size 33 src nh scl [fmode]
```

Initialization

size -- number of points in the table. Must be power of two and at least 4.

src -- source table number. This table contains the parameters of each partial in the following format:

stra, pna, phsa, strb, pnb, phsb, ...

the parameters are:

- stra, strb, etc.: relative strength of partials. The actual amplitude depends on the value of scl, or normalization (if enabled).
- pna, pnb, etc.: partial number, or frequency, depending on fmode (see below); zero and negative values are allowed, however, if the absolute value of the partial number exceeds ($\text{size} / 2$), the partial will not be rendered. With *GEN33*, partial number is rounded to the nearest integer.
- phsa, phsb, etc.: initial phase, in the range 0 to 1.

Table length (not including the guard point) should be at least $3 * \text{nh}$. If the table is too short, the number of partials (nh) is reduced to $(\text{table length}) / 3$, rounded towards zero.

nh -- number of partials. Zero or negative values are allowed, and result in an empty table (silence). The actual number may be reduced if the source table (src) is too short, or some partials have too high frequency.

scl -- amplitude scale.

fmode (optional, default = 0) -- a non-zero value can be used to set frequency in Hz instead of partial numbers in the source table. The sample rate is assumed to be fmode if it is positive, or $-(\text{sr} * \text{fmode})$ if any negative value is specified.

Examples

```
; partials 1, 4, 7, 10, 13, 16, etc. with base frequency of 400 Hz

ibsfrq = 400
; estimate number of partials
inumh = int(1.5 + sr * 0.5 / (3 * ibsfrq))
; source table length
isrcln = int(0.5 + exp(log(2) * int(1.01 + log(inumh * 3) / log(2))))
; create empty source table
itmp   ftgen 1, 0, isrcln, -2, 0
ifpos  = 0
ifrq   = ibsfrq
inumh  = 0
11:
    tableiw ibsfrq / ifrq, ifpos, 1      ; amplitude
    tableiw ifrq, ifpos + 1, 1          ; frequency
    tableiw 0, ifpos + 2, 1             ; phase
ifpos  = ifpos + 3
ifrq   = ifrq + ibsfrq * 3
inumh  = inumh + 1
if (ifrq < (sr * 0.5)) igoto 11

; store output in ftable 2 (size = 262144)

itmp   ftgen 2, 0, 262144, -33, 1, inumh, 1, -1
```

See Also

GEN09, GEN34

Credits

Programmer: Istvan Varga
March 2002

New in version 4.19

GEN34

GEN34 — Generate composite waveforms by mixing simple sinusoids.

Description

These routines generate composite waveforms by mixing simple sinusoids, similarly to *GEN09*, but the parameters of the partials are specified in an already existing table, which makes it possible to calculate any number of partials in the orchestra.

The difference between *GEN33* and *GEN34* is that *GEN33* uses inverse FFT to generate output, while *GEN34* is based on the algorithm used in *oscils* opcode. *GEN33* allows integer partials only, and does not support power of two plus 1 table size, but may be significantly faster with a large number of partials. On the other hand, with *GEN34*, it is possible to use non-integer partial numbers and extended guard point, and this routine may be faster if there is only a small number of partials (note that *GEN34* is also several times faster than *GEN09*, although the latter may be more accurate).

Syntax

```
f # time size 34 src nh scl [fmode]
```

Initialization

size -- number of points in the table. Must be power of two or a power of two plus 1.

src -- source table number. This table contains the parameters of each partial in the following format:

stra, pna, phsa, strb, pnb, phsb, ...

the parameters are:

- stra, strb, etc.: relative strength of partials. The actual amplitude depends on the value of scl, or normalization (if enabled).
- pna, pnb, etc.: partial number, or frequency, depending on fmode (see below); zero and negative values are allowed, however, if the absolute value of the partial number exceeds ($\text{size} / 2$), the partial will not be rendered.
- phsa, phsb, etc.: initial phase, in the range 0 to 1.

Table length (not including the guard point) should be at least $3 * \text{nh}$. If the table is too short, the number of partials (nh) is reduced to $(\text{table length}) / 3$, rounded towards zero.

nh -- number of partials. Zero or negative values are allowed, and result in an empty table (silence). The actual number may be reduced if the source table (src) is too short, or some partials have too high frequency.

scl -- amplitude scale.

fmode (optional, default = 0) -- a non-zero value can be used to set frequency in Hz instead of partial numbers in the source table. The sample rate is assumed to be fmode if it is positive, or $-(\text{sr} * \text{fmode})$ if any negative value is specified.

Examples

```
; partials 1, 4, 7, 10, 13, 16, etc. with base frequency of 400 Hz

ibsfrq = 400
; estimate number of partials
inumh = int(1.5 + sr * 0.5 / (3 * ibsfrq))
; source table length
isrcln = int(0.5 + exp(log(2) * int(1.01 + log(inumh * 3) / log(2))))
; create empty source table
itmp   ftgen 1, 0, isrcln, -2, 0
ifpos  = 0
ifrq   = ibsfrq
inumh  = 0
11:
    tableiw ibsfrq / ifrq, ifpos, 1      ; amplitude
    tableiw ifrq, ifpos + 1, 1          ; frequency
    tableiw 0, ifpos + 2, 1             ; phase
ifpos  = ifpos + 3
ifrq   = ifrq + ibsfrq * 3
inumh  = inumh + 1
if (ifrq < (sr * 0.5)) igoto 11

; store output in ftable 2 (size = 262144)

itmp   ftgen 2, 0, 262144, -34, 1, inumh, 1, -1
```

See Also

GEN09, GEN33

Credits

Programmer: Istvan Varga
March 2002

New in version 4.19

GEN40

GEN40 — Generates a random distribution using a distribution histogram.

Description

Generates a continuous random distribution function starting from the shape of a user-defined distribution histogram.

Syntax

```
f # time size 40 shapetab
```

Performance

The shape of histogram must be stored in a previously defined table, in fact shapetab argument must be filled with the number of such table.

Histogram shape can be generated with any other GEN routines. Since no interpolation is used when GEN40 processes the translation, it is suggested that the size of the table containing the histogram shape to be reasonably big, in order to obtain more precision (however after the processing the shaping-table can be destroyed in order to re-gain memory).

This subroutine is designed to be used together with cusernd opcode (see cusernd for more information).

Credits

Author: Gabriel Maldonado

GEN41

GEN41 — Generates a random list of numerical pairs.

Description

Generates a discrete random distribution function by giving a list of numerical pairs.

Syntax

```
f # time size -41 value1 prob1 value2 prob2 value3 prob3 ... valueN probN
```

Performance

The first number of each pair is a value, and the second is the probability of that value to be chosen by a random algorithm. Even if any number can be assigned to the probability element of each pair, it is suggested to give it a percent value, in order to make it clearer for the user.

This subroutine is designed to be used together with `dusernd` and *urd* opcodes (see `dusernd` for more information).

Credits

Author: Gabriel Maldonado

GEN42

GEN42 — Generates a random distribution of discrete ranges of values.

Description

Generates a random distribution function of discrete ranges of values by giving a list of groups of three numbers.

Syntax

```
f # time size -42 min1 max1 prob1 min2 max2 prob2 min3 max3 prob3 ... minN maxN probN
```

Performance

The first number of each group is a the minimum value of the range, the second is the maximum value and the third is the probability of that an element belonging to that range of values can be chosen by a random algorithm. Even if any number can be assigned to the probability element of each group, it is suggested to give it a percent value, in order to make it clearer to the user.

This subroutine is designed to be used together with `dusernd` and `urd` opcodes (see `dusernd` for more information). Since both `dusernd` and `urd` do not use any interpolation, it is suggested to give a size reasonably big.

Credits

Author: Gabriel Maldonado

GEN43

GEN43 — Loads a PVOCEX file containing a PV analysis.

Description

This subroutine loads a PVOCEX file containing the PV analysis (amp-freq) of a soundfile and calculates the average magnitudes of all analysis frames of one or all audio channels. It then creates a table with these magnitudes for each PV bin.

Syntax

```
f # time size 43 filecod channel
```

Initialisation

size -- number of points in the table, power-of-two or power-of-two plus 1. GEN 43 does not make any distinction between these two sizes, but it requires the table to be at least the $\text{fftsize}/2$. PV bins cover the positive spectrum from 0Hz (table index 0) to the Nyquist (table index $\text{fftsize}/2+1$) in equal-size frequency increments (of size $\text{sr}/\text{fftsize}$).

filecod -- a pvocex file (which can be generated by pvanal).

channel -- audio channel number from which the magnitudes will be extracted; a 0 will average the magnitudes from all channels.

Reading stops at the end of the file.



Note

if p4 is positive, the table will be post-normalised. A negative p4 will cause post-normalisation to be skipped.

Examples

```
f1 0 512 43 "viola.pvx" 1
f1 0 -1024 -43 "noiseprint.pvx" 0
```

This table can be used as a masking table for pvstencil and pvsmaska. The first example uses a 1024-point FFT phase vocoder analysis file from which the first channel is used. The second uses all channels of a 2048-point file, without post-normalisation. For noise reduction applications with pvstencil, it is easiest to skip table normalisation (negative GEN code).

Credits

Author: Victor Lazzarini

GEN51

GEN51 — This subroutine fills a table with a fully customized micro-tuning scale, in the manner of Csound opcodes *cpstun*, *cpstuni* and *cpstmid*.

Description

This subroutine fills a table with a fully customized micro-tuning scale, in the manner of Csound opcodes *cpstun*, *cpstuni* et *cpstmid*.

Syntax

```
f # time size -51 numgrades interval basefreq basekey tuningRatio1 tuningRatio2 .... tuningRatioN
```

Performance

The first four parameters (i.e. p5, p6, p7 and p8) define the following generation directives:

p5 (numgrades) -- the number of grades of the micro-tuning scale

p6 (interval) -- the frequency range covered before repeating the grade ratios, for example 2 for one octave, 1.5 for a fifth etcetera

p7 (basefreq) -- the base frequency of the scale in cps

p8 (basekey) -- the integer index of the scale to which to assign basefreq unmodified

The other parameters define the ratios of the scale:

p9...pN (tuningRatio1...etc.) -- the tuning ratios of the scale

For example, for a standard 12-grade scale with the base-frequency of 261 cps assigned to the key-number 60, the corresponding f-statement in the score to generate the table should be:

```
;          numgrades      basefreq      tuning-ratios (eq.temp) .....  
;          interval      basekey  
f1 0 64 -51      12          2          261          60          1  1.059463 1.12246 1.18920 ..etc...
```

After the gen has been processed, the table f1 is filled with 64 different frequency values. The 60th element is filled with the frequency value of 261, and all other elements (preceding and subsequent) of the table are filled according to the tuning ratios

Another example with a 24-grade scale with a base frequency of 440 assigned to the key-number 48, and a repetition interval of 1.5:

```
;          numgrades      basefreq      tuning-ratios .....  
;          interval      basekey  
f1 0 64 -51      24          1.5        440          48          1  1.01  1.02  1.03  ..etc...
```

Credits

Author: Gabriel Maldonado

GEN52

GEN52 — Creates an interleaved multichannel table from the specified source tables, in the format expected by the *ftconv* opcode.

Description

GEN52 creates an interleaved multichannel table from the specified source tables, in the format expected by the *ftconv* opcode. It can also be used to extract a channel from a multichannel table and store it in a normal mono table, copy tables with skipping some samples, adding delay, or store in reverse order, etc.

Three parameters must be given for each channel to be processed. *fsrc* declares the source f-table number. The parameter *offset* specifies an offset for the source file. If different to 0, the source file is not read from the beginning, but the *offset* number of values are skipped. The *offset* is used to determine the channel number to be read from interleaved f-tables, e.g. for channel 2, *offset* must be 1. It can also be used to set a read offset on the source table. This parameter gives absolute values, so if a skip of 20 sample frames for a 2 channel f-table is desired, *offset* must be set to 40. The *srcchnls* parameter is used to declare the number of channels in the source f-table. This parameter sets the skip size when reading the source f-table.

When more than one channel (*nchannels* > 1) is given, source f-tables are interleaved in the newly created table.

If the source f-table is finished before the destination f-table is full, the remaining values are set to 0.

Syntax

```
f # time size 52 nchannels fsrc1 offset1 srcchnls1 [fsrc2 offset2 srcchnls2 ... fsrcN offsetN srcchnlsN]
```

Example

```
; source tables
f 1 0 16384 10 1
f 2 0 16384 10 0 1
; create 2 channel interleaved table
f 3 0 32768 -52 2 1 0 1 2 0 1
; extract first channel from table 3
f 4 0 16384 -52 1 3 0 2
; extract second channel from table 3
f 5 0 16384 -52 1 3 1 2
```

Credits

Author: Istvan Varga

The Utility Programs

Dan Ellis, MIT Media Lab

The Csound Utilities are *soundfile preprocessing* programs that return information on a soundfile or create some analyzed version of it for use by certain Csound generators. Though different in goals, they share a common soundfile access mechanism and are describable as a set. The Soundfile Utility programs can be invoked in two equivalent forms:

```
csound [-U utilname] [flags] [filenames]
```

```
utilname [flags] [filenames]
```

In the first, the utility is invoked as part of the Csound executable, while in the second it is called as a standalone program. The second is smaller by about 200K, but the two forms are identical in function. The first is convenient in not requiring the maintenance and use of several independent programs - one program does all. When using this form, a *-U flag* detected in the command line will cause all subsequent flags and names to be interpreted as per the named utility; i.e. Csound generation will not occur, and the program will terminate at the end of utility processing.

Directories.

Filenames are of two kinds, source soundfiles and resultant analysis files. Each has a hierarchical naming convention, influenced by the directory from which the Utility is invoked. Source soundfiles with a full pathname (begins with dot (.), slash (/), or for ThinkC includes a colon (:)), will be sought only in the directory named. Soundfiles without a path will be sought first in the current directory, then in the directory named by the SSDIR environment variable (if defined), then in the directory named by SFDIR. An unsuccessful search will return a "cannot open" error.

Resultant analysis files are written into the current directory, or to the named directory if a path is included. It is tidy to keep analysis files separate from sound files, usually in a separate directory known to the SADIR variable. Analysis is conveniently run from within the SADIR directory. When an analysis file is later invoked by a Csound generator it is sought first in the current directory, then in the directory defined by SADIR.

Soundfile Formats.

Csound can read and write audio files in a variety of formats. Write formats are described by Csound command flags. On reading, the format is determined from the soundfile header, and the data automatically converted to floating-point during internal processing. When Csound is installed on a host with local soundfile conventions (SUN, NeXT, Macintosh) it may conditionally include local packaging code which creates soundfiles not portable to other hosts. However, Csound on any host can always generate and read AIFF files, which is thus a portable format. Sampled sound libraries are typically AIFF, and the variable SSDIR usually points to a directory of such sounds. If defined, the SSDIR directory is in the search path during soundfile access. Note that some AIFF sampled sounds have an audio looping feature for sustained performance; the analysis programs will traverse any loop segment once only.

For soundfiles without headers, an SR value may be supplied by the *-r flag* (or its default). If both the *SR header* and the command-line flag are present, the flag value will override the header.

When sound is accessed by the audio Analysis programs, only a single channel is read. For stereo or quad files, the default is channel one; alternate channels may be obtained on request.

Analysis File Generation (ATSA, CVANAL, HETRO, LPANAL, PVANAL)

The following utilities exist for Soundfile analysis:

- *ATSA*: ATS analysis for use with the Csound *ATS Resynthesis* opcodes.
- *CVANAL*: Impulse Response Fourier Analysis for *convolve* operator.
- *HETRO*: Heterodyne analysis for the Csound *adsyn* generator.
- *LPANAL*: Linear predictive coding analysis for the Csound *Linear Predictive Coding (LPC) Resynthesis* opcodes.
- *PVANAL*: Phase vocoder analysis for the Csound *pvoc* generator.

atsa

atsa — Performs ATS analysis on a soundfile.

Description

ATS analysis for use with the Csound *ATS Resynthesis* opcodes.

Syntax

```
csound -U atsa [flags] infilename outfilename
```

Initialization

The following flags can be set for atsa (The default values are stated in parenthesis):

- b start (0.000000 seconds)
- e duration (0.000000 seconds or end)
- l lowest frequency (20.000000 Hertz)
- H highest frequency (20000.000000 Hertz)
- d frequency deviation (0.100000 of partial freq.)
- c window cycles (4 cycles)
- w window type (type: 1) (Options: 0=BLACKMAN, 1=BLACKMAN_H, 2=HAMMING, 3=VONHANN)
- h hop size (0.250000 of window size)
- m lowest magnitude (-60.000000)
- t track length (3 frames)
- s min. segment length (3 frames)
- g min. gap length (3 frames)
- T SMR threshold (30.000000 dB SPL)
- S min. segment SMR (60.000000 dB SPL)
- P last peak contribution (0.000000 of last peak's parameters)
- M SMR contribution (0.500000)
- F File Type (type: 4) (Options: 1=amp.and freq. only, 2=amp.,freq. and phase, 3=amp.,freq. and residual, 4=amp.,freq.,phase, and residual)

Parameters

ATS analysis was devised by Juan Pampin. For complete information on ATS visit: <http://www-ccrma.stanford.edu/~juan/ATS.html>.

Analysis parameters must be carefully tuned for the Analysis Algorithm (ATSA) to properly capture the nature of the signal to be analyzed. As there are a significant number of them, ATSH offers the possibility of Saving/Loading them in a Binary File carrying the extension "*.apf". The extension is not mandatory, but recommended. A brief explanation of each Analysis Parameters follows:

1. Start (secs.): the starting time of the analysis in seconds.
2. Duration (secs.): the duration time of the analysis in seconds. A zero means the whole duration of the input sound file.
3. Lowest Frequency (Hz.): this parameter will partially determine the size of the Analysis Window to be used. To compute the size of the Analysis Window, the period of the Lowest Frequency in samples (SR / LF) is multiplied by the number of cycles of it the user wants to fit in the Analysis Window (see parameter 6). This value is rounded to the next power of two to determine the size of the FFT for the analysis. The remaining samples are zero-padded. If the signal is a single, harmonic sound, then the value of the Lowest Frequency should be its fundamental frequency or a sub-harmonic of it. If it is not harmonic, then its lowest significant frequency component may be a good starting value.

4. Highest Frequency (Hz.): highest frequency to be taken into account for Peak Detection. Once it is determined that no relevant information is found beyond a certain frequency, the analysis may be faster and more accurate setting the Highest Frequency parameter to that value.
5. Frequency Deviation (Ratio): frequency deviation allowed for each peak in the Peak Continuation Algorithm, as a ratio of the frequency involved. For instance, considering a peak at 440 Hz and a Deviation of .1 will produce that the Peak Continuation Algorithm will only try to find candidates for its continuation between 396 and 484 Hz (10% above and below the frequency of the peak). A small value is likely to produce more trajectories whilst a large value will reduce them, but at the cost of rendering information difficult to be further processed.
6. Number of Cycles of Lowest Frequency to fit in Analysis Window: this will also partially determine the size of the Fourier Analysis Window to be used. See Parameter 3. For single harmonic signals, it is supposed to be more than one (typically 4).
7. Hop Size (Ratio): size of the gap between one Analysis Window and the next expressed as a ratio of the Window Size. For instance, a Hop Size value of .25 will "jump" by 512 samples (Windows will overlap for a 75% of their size). This parameter will also determine the size of the analysis frames obtained. Signals that change their spectra very fast (such as Speech sounds) may need a high frame rate in order to properly track their changes.
8. Amplitude Threshold (dB): the highest amplitude value to be taken into account for Peak Detection.
9. Window Type: the shape of the smoothing function to be used for the Fourier Analysis. There are four choices available at present: Blackman, Blackman-Harris, Von Hann, and Hanning. Precise specifications about them are easily found on D.S.P. bibliography.
- 10 Track Length (Frames): The Peak Continuation Algorithm will "look-back" by Length frames in order to do its job better, preventing frequency trajectories from curving too much and losing stability. However, a large value for this parameter will slow down the analysis significantly.
- 11 Minimal Segment Length (Frames): once the analysis is done, the spectral data can be further "cleaned" up during post-processing. Trajectories shorter than this value are suppressed if their average SMR is below Minimal Segment SMR (see parameters 16 and 14). This might help to avoid non-relevant sudden changes while keeping a high frame rate, reducing also the number of intermittent sinusoids during synthesis.
- 12 Minimal Gap Length (Frames): as parameter 11, this one is also used to clean up the data during post-processing. In this case, gaps (zero amplitude values, i.e. theoretical "silence") longer than Length frames are filled up with amplitude/frequency values obtained by linear interpolation of the adjacent active frames. This parameter prevents sudden interruptions of stable trajectories while keeping a high frame rate.
- 13 SMR Threshold (dB SPL): also a post-processing parameter, the SMR Threshold is used to eliminate partials with low averages.
- 14 Minimal Segment SMR (dB SPL): this parameter is used in combination with parameter 11. Short segments with SMR average below this value will be removed during post-processing.
- 15 Last Peak Contribution (0 to 1): as explained in Parameter 10, the Peak Continuation Algorithm "looks-back" several number of frames to do its job better. This parameter will help to weight the contribution of the first precedent peak over the others. A zero value means that all precedent peaks (to the size of Parameter 10) are equally taken in account.
- 16 SMR Contribution (0 to 1): In addition to the proximity in frequency of the peaks, the ATS Peak Continuation Algorithm may use psycho-acoustic information (the Signal-to-Mask-Ratio, or SMR) to improve the perceptual results. This parameter indicates how much the SMR information is used during tracking. For instance, a value of .5 makes the Peak Continuation Algorithm to use a 50% of SMR information and a 50% of Frequency Proximity information to decide which is the best candidate to continue a sinusoidal track.

Examples

The following command:

```
atsa -b0.1 -e1 -l100 -H10000 -w2 audiofile.wav audiofile.ats
```

Generates the ATS analysis file 'audiofile.ats' from the original 'audiofile.wav' file. It begins analysis from second 0.1 of the file and the analysis is performed for 1 second thereafter. The lowest frequency stored is 100 Hz and the highest is 10kHz. A Hamming window is used for each analysis frame.

cvanal

cvanal — Converts a soundfile into a single Fourier transform frame.

Description

Impulse Response Fourier Analysis for *convolve* operator

Syntax

```
csound -U cvanal [flags] infilename outfile
```

```
cvanal [flags] infilename outfile
```

Initialization

cvanal -- converts a soundfile into a single Fourier transform frame. The output file can be used by the *convolve* operator to perform Fast Convolution between an input signal and the original impulse response. Analysis is conditioned by the flags below. A space is optional between the flag and its argument.

-s rate -- sampling rate of the audio input file. This will over-ride the srate of the soundfile header, which otherwise applies. If neither is present, the default is 10000.

-c channel -- channel number sought. If omitted, the default is to process all channels. If a value is given, only the selected channel will be processed.

-b begin -- beginning time (in seconds) of the audio segment to be analyzed. The default is 0.0

-d duration -- duration (in seconds) of the audio segment to be analyzed. The default of 0.0 means to the end of the file.

Examples

```
cvanal asound cvfile
```

will analyze the soundfile "asound" to produce the file "cvfile" for the use with *convolve*.

To use data that is not already contained in a soundfile, a soundfile converter that accepts text files may be used to create a standard audio file, e.g., the .DAT format for SOX. This is useful for implementing FIR filters.

Files

The output file has a special *convolve* header, containing details of the source audio file. The analysis data is stored as "float", in rectangular (real/imaginary) form.



Note

The analysis file is *not* system independent! Ensure that the original impulse recording/data is retained. If/when required, the analysis file can be recreated.

Credits

Author: Greg Sullivan

Based on algorithm given in *Elements Of Computer Music*, by F. Richard Moore.

hetro

hetro — Decomposes an input soundfile into component sinusoids.

Description

Hetrodyne filter analysis for the Csound *adsyn* generator.

Syntax

```
csound -U hetro [flags] infilename outfilename
```

```
hetro [flags] infilename outfilename
```

Initialization

hetro takes an input soundfile, decomposes it into component sinusoids, and outputs a description of the components in the form of breakpoint amplitude and frequency tracks. Analysis is conditioned by the control flags below. A space is optional between flag and value.

-s srate -- sampling rate of the audio input file. This will over-ride the srate of the soundfile header, which otherwise applies. If neither is present, the default is 10000. Note that for *adsyn* synthesis the srate of the source file and the generating orchestra need not be the same.

-c channel -- channel number sought. The default is 1.

-b begin -- beginning time (in seconds) of the audio segment to be analyzed. The default is 0.0

-d duration -- duration (in seconds) of the audio segment to be analyzed. The default of 0.0 means to the end of the file. Maximum length is 32.766 seconds.

-f begfreq -- estimated starting frequency of the fundamental, necessary to initialize the filter analysis. The default is 100 (cps).

-h partials -- number of harmonic partials sought in the audio file. Default is 10, maximum is a function of memory available.

-M maxamp -- maximum amplitude summed across all concurrent tracks. The default is 32767.

-m minamp -- amplitude threshold below which a single pair of amplitude/frequency tracks is considered dormant and will not contribute to output summation. Typical values: 128 (48 db down from full scale), 64 (54 db down), 32 (60 db down), 0 (no thresholding). The default threshold is 64 (54 db down).

-n brkpts -- initial number of analysis breakpoints in each amplitude and frequency track, prior to thresholding (*-m*) and linear breakpoint consolidation. The initial points are spread evenly over the duration. The default is 256.

-l cutfreq -- substitute a 3rd order Butterworth low-pass filter with cutoff frequency *cutfreq* (in Hz), in place of the default averaging comb filter. The default is 0 (don't use).

Performance

As of Csound 4.08, *hetro* can write SDIF output files if the output file name ends with ".sdif" or ".SDIF". See the *sdif2ad* utility for more information about the Csound's SDIF support.

Examples

```
hetro -s44100 -b.5 -d2.5 -h16 -M24000 audiofile.test adsynfile7
```

This will analyze 2.5 seconds of channel 1 of a file "audiofile.test", recorded at 44.1 kHz, beginning .5 seconds from the start, and place the result in a file "adsynfile7". We request just the first 16 harmonics of the sound, with 256 initial breakpoint values per amplitude or frequency track, and a peak summation amplitude of 24000. The fundamental is estimated to begin at 100 Hz. Amplitude thresholding is at 54 db down.

The Butterworth LPF is not enabled.

File Format

The output file contains time-sequenced amplitude and frequency values for each partial of an additive complex audio source. The information is in the form of breakpoints (time, value, time, value,) using 16-bit integers in the range 0 - 32767. Time is given in milliseconds, and frequency in Hertz (cps). The breakpoint data is exclusively non-negative, and the values -1 and -2 uniquely signify the start of new amplitude and frequency tracks. A track is terminated by the value 32767. Before being written out, each track is data-reduced by amplitude thresholding and linear breakpoint consolidation.

A component partial is defined by two breakpoint sets: an amplitude set, and a frequency set. Within a composite file these sets may appear in any order (amplitude, frequency, amplitude; or amplitude, amplitude..., then frequency, frequency,...). During *adsyn* resynthesis the sets are automatically paired (amplitude, frequency) from the order in which they were found. There should be an equal number of each.

A legal *adsyn* control file could have following format:

```
-1 time1 value1 ... timeK valueK 32767 ; amplitude breakpoints for partial 1
-2 time1 value1 ... timeL valueL 32767 ; frequency breakpoints for partial 1
-1 time1 value1 ... timeM valueM 32767 ; amplitude breakpoints for partial 2
-2 time1 value1 ... timeN valueN 32767 ; frequency breakpoints for partial 2
-2 time1 value1 .....
-2 time1 value1 ..... ; pairable tracks for partials 3 and 4
-1 time1 value1 .....
-1 time2 value1 .....
```

Credits

Author: Tom Sullivan

1992

Author: John ffitich

1994

Author: Richard Dobson

2000

October 2002. Thanks to Rasmus Ekman, added a note about the SDIF format.

lpanal

lpanal — Performs both linear predictive and pitch-tracking analysis on a soundfile.

Description

Linear predictive analysis for the Csound *Linear Predictive Coding (LPC) Resynthesis* opcodes.

Syntax

```
csound -U lpanal [flags] infilename outfilename
```

```
lpanal [flags] infilename outfilename
```

Initialization

lpanal performs both lpc and pitch-tracking analysis on a soundfile to produce a time-ordered sequence of *frames* of control information suitable for Csound resynthesis. Analysis is conditioned by the control flags below. A space is optional between the flag and its value.

-a -- [alternate storage] asks *lpanal* to write a file with filter poles values rather than the usual filter coefficient files. When *lpread* / *lpreson* are used with pole files, automatic stabilization is performed and the filter should not get wild. (This is the default in the Windows GUI) - Changed by Marc Resibois.

-s srate -- sampling rate of the audio input file. This will over-ride the srate of the soundfile header, which otherwise applies. If neither is present, the default is 10000.

-c channel -- channel number sought. The default is 1.

-b begin -- beginning time (in seconds) of the audio segment to be analyzed. The default is 0.0

-d duration -- duration (in seconds) of the audio segment to be analyzed. The default of 0.0 means to the end of the file.

-p npoles -- number of poles for analysis. The default is 34, the maximum 50.

-h hopsize -- hop size (in samples) between frames of analysis. This determines the number of frames per second (srate / hopsize) in the output control file. The analysis framesize is hopsize * 2 samples. The default is 200, the maximum 500.

-C string -- text for the comments field of the lpf header. The default is the null string.

-P mincps -- lowest frequency (in Hz) of pitch tracking. -P0 means no pitch tracking.

-Q maxcps -- highest frequency (in Hz) of pitch tracking. The narrower the pitch range, the more accurate the pitch estimate. The defaults are -P70, -Q200.

-v verbosity -- level of terminal information during analysis.

- 0 = none
- 1 = verbose
- 2 = debug

The default is 0.

Examples

```
lpanal -a -p26 -d2.5 -P100 -Q400 audiofile.test lpfil22
```

will analyze the first 2.5 seconds of file "audiofile.test", producing sr/200 frames per second, each containing 26-pole filter coefficients and a pitch estimate between 100 and 400 Hertz. Stabilized (-a) output will be placed in "lpfil22" in the current directory.

File Format

Output is a file comprised of an identifiable header plus a set of frames of floating point analysis data. Each frame contains four values of pitch and gain information, followed by *npoles* filter coefficients. The file is readable by Csound's *lpread*.

lpanal is an extensive modification of Paul Lanksy's lpc analysis programs.

pvanal

pvanal — Converts a soundfile into a series of short-time Fourier transform frames.

Description

Fourier analysis for the Csound *pvoc* generator

Syntax

```
csound -U pvanal [flags] infilename outfile
```

```
pvanal [flags] infilename outfile
```

Pvanal extension to create a PVOC-EX file.

The standard Csound utility program *pvanal* has been extended to enable a PVOC-EX format file to be created, using the existing interface. To create a PVOC-EX file, the file name must be given the required extension, “.pvx”, e.g “test.pvx”. The requirement for the FFT size to be a power of two is here relaxed, and any positive value is accepted; odd numbers are rounded up internally. However, power-of-two sizes are still to be preferred for all normal applications.

The channel select flags are ignored, and all source channels will be analysed and written to the output file, up to a compiler-set limit of eight channels. The analysis window size (*iwinsize*) is set internally to double the FFT size.

Initialization

pvanal converts a soundfile into a series of short-time Fourier transform (STFT) frames at regular timepoints (a frequency-domain representation). The output file can be used by *pvoc* to generate audio fragments based on the original sample, with timescales and pitches arbitrarily and dynamically modified. Analysis is conditioned by the flags below. A space is optional between the flag and its argument.

-s srate -- sampling rate of the audio input file. This will over-ride the *srate* of the soundfile header, which otherwise applies. If neither is present, the default is 10000.

-c channel -- channel number sought. The default is 1.

-b begin -- beginning time (in seconds) of the audio segment to be analyzed. The default is 0.0

-d duration -- duration (in seconds) of the audio segment to be analyzed. The default of 0.0 means to the end of the file.

-n frmsiz -- STFT frame size, the number of samples in each Fourier analysis frame. Must be a power of two, in the range 16 to 16384. For clean results, a frame must be larger than the longest pitch period of the sample. However, very long frames result in temporal "smearing" or reverberation. The bandwidth of each STFT bin is determined by sampling rate / frame size. The default framesize is the smallest power of two that corresponds to more than 20 milliseconds of the source (e.g. 256 points at 10 kHz sampling, giving a 25.6 ms frame).

-w windfact -- Window overlap factor. This controls the number of Fourier transform frames per second. Csound's *pvoc* will interpolate between frames, but too few frames will generate audible distortion; too many frames will result in a huge analysis file. A good compromise for *windfact* is 4, meaning that each input point occurs in 4 output windows, or conversely that the offset between successive STFT frames is *framesize*/4. The default value is 4. Do not use this flag with *-h*.

-h hopsize -- STFT frame offset. Converse of above, specifying the increment in samples between successive frames of analysis (see also *lpanal*). Do not use with *-w*.

-H -- Use a Hamming window instead of the default von Hann window.

-K -- Use a Kaiser window instead of the default von Hann window. The Kaiser parameter default is 6.8, but can be set with the *-B* option.

-B beta -- Set the beta parameter for any Kaiser window used to the floating point value beta.

Examples

```
pvanal asound pvfile
```

will analyze the soundfile "asound" using the default frmsiz and windfact to produce the file "pvfile" suitable for use with pvoc.

Files

The output file has a special *pvoc* header containing details of the source audio file, the analysis frame rate and overlap. Frames of analysis data are stored as float, with the magnitude and “frequency” (in Hz) for the first $N/2 + 1$ Fourier bins of each frame in turn. “Frequency” encodes the phase increment in such a way that for strong harmonics it gives a good indication of the true frequency. For low amplitude or rapidly moving harmonics it is less meaningful.

Diagnostics

Prints total number of frames, and frames completed every 20th frame.

Credits

Author: Dan Ellis

MIT Media Lab

Cambridge, Massachussetts

1990

File Queries (SNDINFO)

The following utilities exist for Soundfile query:

- *SNDINFO*: Displays information about a soundfile.

sndinfo

sndinfo — Displays information about a soundfile.

Description

Get basic information about one or more soundfiles.

Syntax

```
csound -U sndinfo [options] soundfilenames ...
```

```
sndinfo [options] soundfilenames ...
```

Initialization

sndinfo will attempt to find each named file, open it for reading, read in the soundfile header, then print a report on the basic information it finds. The order of search across soundfile directories is as described above. If the file is of type AIFF, some further details are listed first.

There are two option types:

1. *-i* or *-il* will print instrument information, which includes looping. The option continues until a *-i0* option.
2. The other option is *-b* which prints the broadcast information for WAV files. It can similarly be negated with *-b0*.

Examples

```
csound -U sndinfo test Bosendorfer/"BOSEN mf A0 st" foo foo2
```

where the environment variables SFDIR = /u/bv/sound, and SSDIR = /so/bv/Samples, might produce the following:

```
util  SNDINFO:
      /u/bv/sound/test:
        srate 22050, monaural, 16 bit shorts, 1.10 seconds
        headersiz 1024, datasiz 48500 (24250 sample frames)

      /so/bv/Samples/Bosendorfer/BOSEN mf A0 st:  AIFF, 197586 stereo samples, base Frq 261.6 (MIDI 6
        AIFF soundfile, looping with modes 1, 0
        srate 44100, stereo, 16 bit shorts, 4.48 seconds

        headersiz 402, datasiz 790344 (197586 sample frames)

      /u/bv/sound/foo:
        no recognizable soundfile header

      /u/bv/sound/foo2:
        couldn't find
```

File Conversion (HET_IMPORT, HET_EXPORT, PVLOOK, PV_EXPORT, PV_IMPORT, SDIF2AD, SR-CONV)

The following utilities exist for file conversion:

- *HET_EXPORT*: Exports a .het (produced by *HETRO*) to a comma separated text file.
- *HET_IMPORT*: Generates a .het (in the format produced by *HETRO*) from a comma separated text file for usage with the *adsyn* generator.
- *PVLOOK*: View formatted text output of STFT analysis files.
- *PV_EXPORT*: Converts a file generated by *PVANAL* to a text file.
- *PV_IMPORT*: Converts a text file (in the format generated by *PV_EXPORT*) to a *PVANAL* format file to be used with the *pvoc* opcode.
- *SDIF2AD*: Converts SDIF files to files usable by *adsynt*.
- *SRCONV*: Converts the sample rate of an audio file.

dnoise

dnoise — Reduces noise in a file.

Description

This is a noise reduction scheme using frequency-domain noise-gating.

Syntax

```
dnoise [flags] -i noise_ref_file -o output_soundfile input_soundfile
```

Initialization

Dnoise specific flags:

- *(no flag)* input soundfile to be denoised
- *-i fname* input reference noise soundfile
- *-o fname* output soundfile
- *-N fnum* # of bandpass filters (default: 1024)
- *-w fovlp* filter overlap factor: {0,1,(2),3} DON'T USE *-w* AND *-M*
- *-M awlen* analysis window length (default: N-1 unless *-w* is specified)
- *-L swlen* synthesis window length (default: M)
- *-D dfac* decimation factor (default: M/8)
- *-b btim* begin time in noise reference soundfile (default: 0)
- *-B smpst* starting sample in noise reference soundfile (default: 0)
- *-e etim* end time in noise reference soundfile (default: end of file)
- *-E smpend* final sample in noise reference soundfile (default: end of file)
- *-t thr* threshold above noise reference in dB (default: 30)
- *-S gfact* sharpness of noise-gate turnoff, range: 1 to 5 (default: 1)
- *-n numfrm* number of FFT frames to average over (default: 5)
- *-m mingain* minimum gain of noise-gate when off in dB (default: -40)

Soundfile format options:

- *-A* AIFF format output
- *-W* WAV format output
- *-J* IRCAM format output
- *-h* skip soundfile header (not valid for AIFF/WAV output)
- *-8* 8-bit unsigned char sound samples

- *-c* 8-bit signed_char sound samples
- *-a* alaw sound samples
- *-u* ulaw sound samples
- *-s* short_int sound samples
- *-l* long_int sound samples
- *-f* float sound samples. Floats also supported for WAV files. (New in Csound 3.47.)

Additional options:

- *-R* verbose - print status info
- *-H [N]* print a heartbeat character at each soundfile write.
- *-- fname* output to log file fname
- *-V* verbose - print status info



Note

DNOISE also looks at the environment variable SFOUTYP to determine soundfile output format.

The *-i* flag is used for a reference noise file (normally created from a short section of the denoised file, where only noise is audible). The input soundfile to be denoised can be given anywhere on the command line, without a flag.

Performance

This is a noise reduction scheme using frequency-domain noise-gating. This should work best in the case of high signal-to-noise with hiss-type noise.

The algorithm is that suggested by Moorer & Berger in “Linear-Phase Bandsplitting: Theory and Applications” presented at the 76th Convention 1984 October 8-11 New York of the Audio Engineering Society (preprint #2132) except that it uses the Weighted Overlap-Add formulation for short-time Fourier analysis-synthesis in place of the recursive formulation suggested by Moorer & Berger. The gain in each frequency bin is computed independently according to

$$\text{gain} = g0 + (1-g0) * [\text{avg} / (\text{avg} + th*th*nref)] ^ sh$$

where *avg* and *nref* are the mean squared signal and noise respectively for the bin in question. (This is slightly different than in Moorer & Berger.)

The critical parameters *th* and *g0* are specified in dB and internally converted to decimal values. The *nref* values are computed at the start of the program on the basis of a noise_soundfile (specified in the command line) which contains noise without signal.

The *avg* values are computed over a rectangular window of *m* FFT frames looking both ahead and behind the current time. This corresponds to a temporal extent of *m***D*/*R* (which is typically (*m***N*/8)/*R*). The default settings of *N*, *M*, and *D* should be appropriate for most uses. A higher sample rate than 16 KHz might indicate a higher *N*.

Credits

Author: Mark Dolson

August 26, 1989

Author: John ffitch

December 30, 2000

Updated by Rasmus Ekman on March 11, 2002.

het_export

het_export — Converts a .het file to a comma separated text file.

Syntax

```
het_export het_file ctext_file
```

```
csound -U het_export het_file ctext_file
```

Initialization

het_file - Name of the input .het file.

ctext_file - Name of the output comma-separated text file.

The *het_export* utility generates a comma-separated text file for manual editing of a .het file produced by the *HETRO* utility. It can be used in combination with *het_import* to produce data for the *adsyn* generator.

Credits

Author: John ffitch

1995

het_import

het_import — Converts a comma-separated text file to a .het file.

Syntax

```
het_import ctext_file het_file
```

```
csound -U het_import ctext_file het_file
```

Initialization

ctext_file - Name of the input comma-separated text file.

het_file - Name of the output .het file.

The *het_import* utility generates a *.het* file usable with the *adsyn* generator. It can be used in combination with *het_export* to modify sound analysis made by the *HETRO* utility.

Credits

Author: John ffitch

1995

pvlook

pvlook — View formatted text output of STFT analysis files.

Description

View formatted text output of STFT analysis files created with *pvanal*.

Syntax

csound -U **pvlook** [flags] infilename

pvlook [flags] infilename

Initialization

plook reads a file, and frequency and amplitude trajectories for each of the analysis bins, in readable text form. The file is assumed to be an STFT analysis file created by *pvanal*. By default, the entire file is processed.

`-bb n` -- begin at analysis bin number n , numbered from 1. Default is 1.

`-eb n` -- end at analysis bin number n . Defaults to the highest.

-bf n -- begin at analysis frame number *n*, numbered from 1. Defaults to 1.

`-efn` -- end at analysis frame number n . Defaults to the highest.

`-i` -- prints values as integers. Defaults to floating point.

Examples

[illegible]

0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000	0.000
Bin 1	Amps.	0.180	0.066	0.252	0.248	0.245	0.246	0.246	0.249
0.252	0.251	0.250	0.248	0.244	0.245	0.248	0.250	0.254	0.251
0.248	0.247	0.244	0.246	0.249	0.250	0.253	0.251	0.247	0.246
0.245	0.246	0.250	0.251	0.252	0.250	0.247	0.245	0.246	0.247
0.251	0.252	0.250	0.249	0.246	0.245	0.248	0.249	0.252	0.253
0.249	0.248	0.245	0.245	0.249	0.251	0.252	0.252	0.249	0.246
0.246	0.245	0.249	0.252	0.252	0.251	0.249	0.245	0.246	0.248
0.250	0.253	0.251	0.249	0.247	0.244	0.247	0.249	0.250	0.253
0.251	0.248	0.247	0.245	0.247	0.250	0.252	0.252	0.251	0.247
0.246	0.246	0.247	0.251	0.252	0.251	0.249	0.246	0.245	0.248
0.249	0.252	0.252	0.249	0.248	0.246	0.245	0.249	0.250	0.252
0.252	0.249	0.247	0.246	0.246	0.249	0.252	0.252	0.251	0.248
0.245	0.246	0.247	0.249	0.253	0.251	0.249	0.247	0.245	0.246
0.248	0.250	0.253	0.251	0.248	0.247	0.244	0.246	0.250	0.251
0.252	0.250	0.247	0.246	0.246	0.248	0.251	0.252	0.251	0.250
0.246	0.245	0.247	0.248	0.251	0.252	0.250	0.248	0.246	0.245
0.248	0.249	0.252	0.252	0.248	0.247	0.245	0.245	0.249	0.251
0.251	0.251	0.248	0.246	0.246	0.247	0.250	0.252	0.251	0.250
0.248	0.244	0.246	0.248	0.250	0.253	0.251	0.248	0.247	0.245
0.247	0.249	0.250	0.252	0.250	0.247	0.246	0.245	0.247	0.251
0.252	0.251	0.250	0.246	0.245	0.247	0.248	0.252	0.252	0.249
0.248	0.245	0.245	0.248	0.249	0.251	0.252	0.248	0.247	0.245
0.245	0.249	0.250	0.251	0.251	0.248	0.246	0.245	0.246	0.249
0.252	0.251	0.250	0.247	0.244	0.246	0.247	0.249	0.252	0.251
0.249	0.247	0.244	0.247	0.249	0.250	0.252	0.250	0.247	0.246
0.245	0.247	0.250	0.251	0.251	0.250	0.246	0.245	0.246	0.248
0.251	0.252	0.250	0.249	0.245	0.245	0.247	0.248	0.251	0.252
0.249	0.247	0.245	0.245	0.248	0.250	0.251	0.251	0.247	0.246
0.245	0.245	0.249	0.251	0.251	0.250	0.247	0.245	0.246	0.246
0.249	0.252	0.251	0.249	0.247	0.244	0.247	0.248	0.250	0.252
0.250	0.247	0.246	0.245	0.247	0.250	0.251	0.252	0.249	0.246
0.245	0.245	0.247	0.251	0.251	0.250	0.249	0.246	0.245	0.247
0.248	0.251	0.251	0.249	0.248	0.245	0.245	0.248	0.249	0.251
0.251	0.248	0.246	0.245	0.245	0.249	0.251	0.251	0.251	0.247
0.245	0.245	0.246	0.249						

3.294 3.292 3.293 3.295 3.291 3.292 3.293 3.290 3.294 3.295
3.292 3.294 3.291 3.289 3.293 3.291 3.293 3.296 3.292 3.293
3.293 3.288 3.292 3.293 3.292 3.296 3.293 3.291 3.294 3.289
3.292 3.295 3.291 3.294 3.293 3.289 3.292 3.291 3.290 3.295
3.293 3.292 3.294 3.289 3.291 3.293 3.290 3.295 3.294 3.290
3.293 3.290 3.289 3.294 3.291 3.293 3.295 3.290 3.292 3.292
3.289 3.293 3.293 3.292 3.295 3.291 3.289 3.292 3.290 3.292
3.295 3.291 3.293 3.292 3.288 3.292 3.291 3.291 3.295 3.291
3.291 3.292 3.289 3.291 3.294 3.291 3.294 3.292 3.289 3.292
3.290 3.290 3.295 3.292 3.293 3.294 3.289 3.291 3.292 3.290
3.294 3.293 3.291 3.293 3.289 3.290 3.293 3.291 3.294 3.295
3.290 3.292 3.291 3.289 3.294 3.293 3.292 3.294 3.290 3.290
3.292 3.289 3.293 3.294 3.291 3.293 3.291 3.289 3.292 3.291
3.291 3.295 3.291 3.291 3.292 3.288 3.292 3.293 3.291 3.295
3.292 3.290 3.292 3.289 3.291 3.294 3.291 3.293 3.292 3.288
3.291 3.291 3.290 3.295 3.292 3.291 3.293 3.289 3.290 3.292
3.290 3.294 3.293 3.290 3.292 3.290 3.289 3.293 3.291 3.292
3.294 3.290 3.290 3.291 3.289 3.293 3.293 3.291 3.293 3.290
3.288 3.291 3.290 3.292 3.294 3.290 3.292 3.291 3.288 3.291
3.291 3.291 3.294 3.291 3.290 3.291 3.288 3.291 3.293 3.291
3.293 3.292 3.288 3.291 3.290 3.290 3.294 3.291 3.291 3.292
3.288 3.290 3.291 3.290 3.294 3.293 3.290 3.292 3.289 3.289
3.293 3.290 3.292 3.293 3.289 3.291 3.290 3.289 3.293 3.292
3.291 3.293 3.289 3.289 3.291 3.289 3.292 3.293 3.290 3.292
3.290 3.288 3.292 3.291 3.291 3.294 3.290 3.290 3.291 3.288
3.291 3.292 3.291 3.293 3.291 3.288 3.291 3.289 3.290 3.293
3.290 3.292 3.292 3.288 3.291 3.291 3.290 3.293 3.291 3.290
3.292 3.288 3.289 3.292 3.290 3.292 3.293 3.289 3.291 3.289
3.288 3.293 3.291 3.291 3.292 3.288 3.289 3.290 3.288 3.292
3.293 3.290 3.292 3.289 3.288 3.291 3.290 3.291 3.293 3.289
3.290 3.290 3.287 3.291 3.291 3.290 3.293 3.290 3.288 3.290
3.288 3.290 3.293 3.291 3.292 3.291 3.288 3.290 3.289 3.289
3.293 3.290 3.290 3.291 3.287 3.289 3.291 3.289 3.292 3.291
3.288 3.290 3.288 3.288 3.292 3.290 3.291 3.292 3.288 3.289
3.290 3.288 3.292 3.292 3.290 3.292 3.289 3.288 3.291 3.289
3.291 3.293 3.289 3.291 3.290 3.287 3.291 3.290 3.290 3.293
3.289 3.289 3.290 3.287 3.290 3.292 3.290 3.292 3.290 3.287
3.290 3.289 3.289 3.292 3.290 3.290 3.291 3.287 3.289 3.290
3.289 3.292 3.291 3.289 3.291 3.288

etc...

Credits

Author: Richard Karpen

Seattle, Wash

1993 (New in Csound version 3.57)

pv_export

pv_export — Converts a .pvx file to a comma separated text file.

Syntax

```
pv_export pv_file ctext_file
```

```
csound -U pv_export pv_file ctext_file
```

Initialization

pv_file - Name of the input .pvx file.

ctest_file - Name of the output comma-separated text file.

The *pv_export* utility generates a comma-separated text file for manual editing of a .pvx file produced by the *PVANAL* utility. It can be used in combination with *pv_import* to produce data for the *pvoc* generator.

Credits

Author: John ffitch

1995

pv_import

pv_import — Converts a comma-separated text file to a .pvx file.

Syntax

```
pv_import ctext_file pv_file
```

```
csound -U pv_import ctext_file pv_file
```

Initialization

ctest_file - Name of the input comma-separated text file.

pv_file - Name of the output .pvx file.

The *pv_import* utility generates a .pvx file usable with the *pvoc* generator. It can be used in combination with *pv_export* to modify sound analysis made by the *PVANAL* utility.

Credits

Author: John ffitch

1995

sdif2ad

sdif2ad — Converts SDIF files to files usable by adsyn.

Description

Convert files Sound Description Interchange Format (SDIF) to the format usable by Csound's *adsyn* opcode. As of Csound version 4.10, *sdif2ad* was available only as a standalone program for Windows console and DOS.

Syntax

```
csound -U sdif2ad [flags] infilename outfilename
```

Initialization

Flags:

- *-sN* -- apply amplitude scale factor N
- *-pN* -- keep only the first N partials. Limited to 1024 partials. The source partial track indices are used directly to select internal storage. As these can be arbitrary values, the maximum of 1024 partials may not be realized in all cases.
- *-r* -- byte-reverse output file data. The byte-reverse option is there to facilitate transfer across platforms, as Csound's *adsyn* file format is not portable.

If the filename passed to *hetro* has the extension “.sdif”, data will be written in SDIF format as 1TRC frames of additive synthesis data. The utility program *sdif2ad* can be used to convert any SDIF file containing a stream of 1TRC data to the Csound *adsyn* format. *sdif2ad* allows the user to limit the number of partials retained, and to apply an amplitude scaling factor. This is often necessary, as the SDIF specification does not, as of the release of *sdif2ad*, require amplitudes to be within a particular range. *sdif2ad* reports information about the file to the console, including the frequency range.

The main advantages of SDIF over the *adsyn* format, for Csound users, is that SDIF files are fully portable across platforms (data is “big-endian”), and do not have the duration limit of 32.76 seconds imposed by the 16 bit *adsyn* format. This limit is necessarily imposed by *sdif2ad*. Eventually, SDIF reading will be incorporated directly into *adsyn*, thus enabling files of any length (subject to system memory limits) to be analysed and processed.

Users should remember that the SDIF formats are still under development. While the 1TRC format is now fairly well established, it can still change.

For detailed information on the Sound Description Interchange Format, refer to the CNMAT website: <http://cnmat.CNMAT.Berkeley.EDU/SDIF>

Some other SDIF resources (including a viewer) are available via the NC_DREAM website: <http://www.bath.ac.uk/~masjpf/NCD/dreamhome.html>

Credits

Author: Richard Dobson

Somerset, England

August, 2000

New in Csound version 4.08

srconv

srconv — Converts the sample rate of an audio file.

Description

Converts the sample rate of an audio file at sample rate R_{in} to a sample rate of R_{out} . Optionally the ratio (R_{in} / R_{out}) may be linearly time-varying according to a set of (time, ratio) pairs in an auxiliary file.

Syntax

```
srconv [flags] infile
```

Initialization

Flags:

- *-P num* = pitch transposition ratio ($srate / r$) [don't specify both *P* and *r*]
- *-P num* = pitch transposition ratio ($srate / r$) [don't specify both *P* and *r*]
- *-Q num* = quality factor (1, 2, 3, or 4: default = 2)
- *-i filnam* = auxiliary breakpoints file (no breakpoint by default. i.e. No ratio change)
- *-r num* = output sample rate (must be specified)
- *-o fnam* = sound output filename
- *-A* = create an AIFF format output soundfile
- *-J* = create an IRCAM format output soundfile
- *-W* = create a WAV format output soundfile
- *-h* = no header on output soundfile
- *-c* = 8-bit signed_char sound samples
- *-a* = alaw sound samples
- *-8* = 8-bit unsigned_char sound samples
- *-u* = ulaw sound samples
- *-s* = short_int sound samples
- *-l* = long_int sound samples
- *-f* = float sound samples
- *-r N* = orchestra srate override
- *-K* = Do not generate PEAK chunks
- *-R* = continually rewrite header while writing soundfile (WAV/AIFF)
- *-H#* = print a heartbeat style 1, 2 or 3 at each soundfile write
- *-N* = notify (ring the bell) when score or miditrack is done

- `--fnam` = log output to file

This program performs arbitrary sample-rate conversion with high fidelity. The method is to step through the input at the desired sampling increment, and to compute the output points as appropriately weighted averages of the surrounding input points. There are two cases to consider:

1. sample rates are in a small-integer ratio - weights are obtained from table.
2. sample rates are in a large-integer ratio - weights are linearly interpolated from table.

Calculate increment: if decimating, then window is impulse response of low-pass filter with cutoff frequency at half of output sample rate; if interpolating, then window is impulse response of low-pass filter with cutoff frequency at half of input sample rate.

Credits

Author: Mark Dolson

August 26, 1989

Author: John ffitich

December 30, 2000

Other Csound Utilities (CS, CSB64ENC, ENVEXT, EXTRACTOR, MAKECSD, MIXER, SCALE)

The following miscellaneous utilities are available:

- *CS*: Starts Csound with a set of options that can be controlled by environment variables, and input and output files determined by the specified filename stem.
- *CSB64ENC*: Converts a binary file to a Base64 encoded text file.
- *ENVEXT*: Extract the envelope of a file to a text list.
- *EXTRACTOR*: Extract a section of audio from an audio file.
- *MAKECSD*: Creates a CSD file from the specified input files.
- *MIXER*: Mixes together a number of soundfiles.
- *SCALE*: Scale the amplitude of a sound file.

CS

cs — Starts Csound with a set of options that can be controlled by environment variables, and input and output files determined by the specified filename stem.

Description

Starts Csound with a set of options that can be controlled by environment variables, and input and output files determined by the specified filename stem.

Syntax

```
cs [-OPTIONS] <name> [CSOUND OPTIONS ... ]
```

Initialization

Flags:

- **- *OPTIONS* = OPTIONS** is a sequence of alphabetic characters that can be used for selecting the Csound executable to be run, as well as the command line flags (see below). There is a default for the option 'r' (selects real-time output), but it can be overridden.
- **<name>** = this is the filename stem for selecting input files; it may contain a path. Files that have .csd, .orc, or .sco extension are searched, and either a CSD or an orc/sco pair that matches <name> the best are selected. MIDI files with a .mid extension are also searched, and if one that matches <name> at least as close as the CSD or orc/sco pair, it is used with the -F flag.



NOTE

The MIDI file is not used if any -M or -F flag is specified by the user - new in version 4.24.0) Unless there is any option (-n or -o) related to audio output, an output file name with the appropriate extension is automatically generated (based on the name of selected input files and format options). The output file is always written to the current directory.



NOTE

file name extensions are not case sensitive.

- **[*CSOUND OPTIONS ...*]** = any number of additional options for Csound that are simply copied to the final command line to be executed.

The command line that is executed is generated from four parts:

1. Csound executable (possibly with options). This is exactly one of the following (the last one has the highest precedence):
 - a built-in default
 - the value of the CSOUND environment variable
 - environment variables with a name in the format of CSOUND_x where x is an uppercase letter selected by characters of the -OPTIONS string. Thus, if the -dcba option is used, and the environment variables CSOUND_B and CSOUND_C are defined, the value of CSOUND_B will take effect.

2. Any number of option lists, added in the following order:
 - either some built-in defaults, or the value of the CSFLAGS environment variable if it is defined.
 - environment variables with a name in the format of CSFLAGS_x where x is an uppercase letter selected by characters of the -OPTIONS string. Thus, if the -dcba option is used, and the environment variables CSFLAGS_A and CSFLAGS_C are defined as '-M 1 -o dac' and '-m231 -H0', respectively, the string '-m231 -H0 -M 1 -o dac' will be added.
3. The explicit options of [CSOUND OPTIONS ...].
4. Any options and file names generated from <name>.



NOTE

Quoted options that contain spaces are allowed.

Examples

Assuming the following environment variables:

```
CSOUND      = csoundfltk.exe -W
CSOUND_D    = csound64.exe -J
CSOUND_R    = csoundfltk.exe -h

CSFLAGS     = -d -m135 -H1 -s
CSFLAGS_D   = -f
CSFLAGS_R   = -m0 -H0 -o dac1 -M "MIDI Yoke NT: 1" -b 200 -B 6000
```

And a directory that contains:

```
foo.orc      piano.csd
foo.sco      piano.mid
im.csd       piano2.mid
ImproSculpt2_share.csd  foobar.csd
```

The following commands will execute as shown:

```
cs foo      => csoundfltk.exe -W -d -m135 -H1 -s -o foo.wav \
foo.orc foo.sco

cs foob     => csoundfltk.exe -W -d -m135 -H1 -s          \
-o foobar.wav foobar.csd

cs -r imp -i adc => csoundfltk.exe -h -d -m135 -H1 -s -m0 -H0 \
-o dac1 -M "MIDI Yoke NT: 1" \
-b 200 -B 6000 -i adc \
ImproSculpt2_share.csd

cs -d im     => csound64.exe -J -d -m135 -H1 -s -f -o im.sf \
im.csd

cs piano    => csoundfltk.exe -W -d -m135 -H1 -s          \
-F piano.mid -o piano.wav \
piano.csd

cs piano2    => csoundfltk.exe -W -d -m135 -H1 -s          \
-F piano2.mid -o piano2.wav \
piano.csd
```

Credits

Author: Istvan Varga

Jan 2003

csb64enc

csb64enc — Converts a binary file to a Base64 encoded text file.

Description

The *csb64enc* utility generates a Base64 encoded text file from a binary file, such as a standard MIDI file (.mid) or any type of audio file. It is useful to convert a file in the format accepted by the *<CsFileB>* section of a csd file, to include the file within it.

Syntax

```
csb64enc [OPTIONS ... ] infile1 [ infile2 [ ... ]]
```

Initialization

Flags:

- *-w n* = set line width of the output file to n (default: 72)
- *-o fname* = output file name (default: stdout)

Examples

```
csb64enc -w 78 -o file.txt file.mid
```

This command produces a Base64 encoded text file from the standard MIDI file *file.mid*. This file can now be pasted within a csd file's *<CsFileB>* section.

See also

makecsd

Credits

Author: Istvan Varga

Jan 2003

envext

envext — Extracts the envelope of a file to a text file.

Syntax

```
envext [-flags] soundfile
```

```
csound -U envext [-flags] soundfile
```

Initialization

soundfile - Name of the input soundfile.

The following flags are available for envext (The default values are stated in parenthesis):

-o *fnam* Name of output filename (newenv)
-w size (in seconds) of analysis window (0.25)

The *envext* utility generates a text file containing time and amplitude pairs by finding the absolute peak within each window.

Example

Using the command (while in the manual directory):

```
csound -U envext examples/mary.wav
```

will produce the a text file containing the following:

```
0.000 0.000
0.000 0.000
0.250 0.000
0.500 0.000
0.750 0.000
1.249 0.170
1.499 0.269
1.530 0.307
1.872 0.263
2.056 0.304
2.294 0.241
2.570 0.216
2.761 0.178
3.077 0.011
3.251 0.001
3.500 0.000
```

Which shows the time for the peak amplitude within each measured window.

Credits

Author: John ffitch

1995

extractor

extractor — Extract a section of audio from an audio file.

Description

Extract a section of audio, by time or sample, from an existing sound file.

Syntax

```
extractor [OPTIONS ... ] infile
```

Initialization

Flags:

- *-S integer* = Start the extract at the given sample number.
- *-Z integer* = End the extract at the given sample number.
- *-Q integer* = Extract given number of samples.
- *-T fptime* = Start the extract at the given time in seconds.
- *-E fptime* = End the extract at the given time in seconds.
- *-D fptime* = Extract given time in seconds.
- *-v* = Verbose mode.
- *-R* = Continually rewrite the header while writing soundfile (WAV/AIFF).
- *-H integer* = Show a "heart-beat" to indicate progress, in style 1, 2 or 3.
- *-N* = Alert call (usually ringing the bell) when finished.
- *-v* = Verbose mode.
- *-o fname* = output file name (default: test.wav)

Examples

The default values are

```
extractor -S 0 -Z end-of-file -otest
```

For example

```
extractor -S 10234 -D 2.13 in.aiff -o out.wav
```

This creates a new sound file taken from sample 10234 and lasting 2.13 seconds.

Credits

Author: John ffitch

1994

makecsd

makecsd — Creates a CSD file from the specified input files.

Description

Creates a CSD file from the specified input files. The first input file that has a .orc extension (case is not significant) is put to the <CsInstruments> section, and the first input file that has a .sco extension becomes <CsScore>. Any remaining files are Base64 encoded and added as <CsFileB> tags. An empty <CsOptions> section is always added.

Some text filtering is performed on the orchestra and score file:

- newlines are converted to the native format of the system on which makecsd is being run.
- blank lines are removed from the beginning and end of files.
- any trailing whitespace is removed from the end of lines.
- optionally, tabs can be expanded to spaces with an user specified tabstop size.

Syntax

```
makecsd [OPTIONS ... ] infile1 [ infile2 [ ... ]]
```

Initialization

Flags:

- - *t n* = expand tabs to spaces using tabstop size *n* (default: disabled). This applies only to the orchestra and score file.
- - *w n* = set Base64 line width to *n* (default: 72). Note: the orchestra and score are not wrapped.
- - *o fname* = output file name (default: stdout)

Examples

```
makecsd -t 6 -w 78 -o file.csd file.mid file.orc file.sco sample.aif
```

This creates a CSD from file.orc and file.sco (tabs are expanded to spaces assuming a tabstop size of 6 characters), and file.mid and sample.aif are added as <CsFileB> tags containing Base64 encoded data with a line width of 78 characters. The output file is file.csd.

Credits

Author: Istvan Varga

Jan 2003

mixer

mixer — Mixes together a number of soundfiles.

Description

Mixes together a number of soundfiles, starting at different times and with individual channel selection from the input files.

Syntax

```
mixer [OPTIONS ... ] infile [[OPTIONS... ] infile] ...
```

Initialization

Flags:

- *-A* = Generate an AIFF output file.
- *-W* = Generate an WAV output file.
- *-h* = Generate an output file with no header.
- *-c* = Generate 8-bit signed_char sound samples.
- *-a* = Generate alaw sound samples.
- *-u* = Generate ulaw sound samples.
- *-s* = Generate short integer sound samples.
- *-l* = Generate long (32 bit) integer sound samples.
- *-f* = Generate floating point samples.
- *-F arg* = Specifies the gain to be applied to the following input file. If arg is a floating point number that gain is applied uniformly to the input. Alternatively it could be a file name which specifies a breakpoint file for varying the gain for different periods.
- *-S integer* = Indicate at which sample to start to mix in the next input file.
- *-T fpnum* = Indicate at which time (in seconds) to start to mix in the next input file.
- *-1* = Mix in channel 1 from next sound file.
- *-2* = Mix in channel 2 from next sound file.
- *-3* = Mix in channel 3 from next sound file.
- *-4* = Mix in channel 4 from next sound file.
- *-^ intx inty* = Mix in channel x from next sound file as channel y in the output.
- *-v* = Verbose mode.
- *-R* = Continually rewrite the header while writing soundfile (WAV/AIFF).
- *-H integer* = Show a "heart-beat" to indicate progress, in style 1, 2 or 3.
- *-N* = Alert call (usually ringing the bell) when finished.

- *-o fname* = output file name (default: test.wav)

Examples

The default values are

```
mixer -s -otest -F 1.0 -S 0
```

For example

```
mixer -F 0.96 in1.wav -S 300 -2 in2.aiff -S 300 -^4 1 in3.wav -o out.wav
```

This creates a new sound file with a constant gain of 0.96 from in1.wav with the second channel of in2.aiff mixed in after 300 samples and channel 4 of in3.wav output as channel 1 after 300 samples.

Credits

Author: John ffitch

1994

scale

scale — Scale the amplitude of a sound file.

Description

Takes a sound file and scales it by applying a gain, either constant or variable. The scale can be specified as a multiplier, a maximum or a percentage of 0db.

Syntax

```
scale [OPTIONS ... ] infile
```

Initialization

Flags:

- *-A* = Generate an AIFF outout file.
- *-W* = Generate an WAV outout file.
- *-h* = Generate an outout file with no header.
- *-c* = Generate 8-bit signed_char sound samples.
- *-a* = Generate alaw sound samples.
- *-u* = Generate ulaw sound samples.
- *-s* = Generate short integer sound samples.
- *-l* = Generate long (32 bit) integer sound samples.
- *-f* = Generate floating point samples.
- *-F arg* = Specifies the gain to be applied. If arg is a floating point number that gain is applied uniformly to the input. Alternatively it could be a file name which specifies a breakpoint file for varying the gain for different periods.
- *-M fpnum* = Scales the input so the maximum absolute displacement is the value given.
- *-P fpnum* = Scales the input so the maximum absolute displacement is the percentage given of 0db.
- *-R* = Continually rewrite the header while writing soundfile (WAV/AIFF).
- *-H integer* = Show a "heart-beat" to indicate progress, in style 1, 2 or 3.
- *-N* = Alert call (usually ringing the bell) when finished.
- *-o fname* = output file name (default: test.wav)

Examples

```
scale -s -W -F 0.96 -o out.wav sound.wav
```

This creates a new sound file with a constant gain of 0.96. It is particularly useful if the input file is in floating point format.

Credits

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1994

Credits

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Cscore

Cscore is an API (application programming interface) for generating and manipulating numeric score files. It is a part of the larger Csound API and includes a number of functions that can be called by a user-designed program written in the C language. *Cscore* can be invoked either as a standalone score preprocessor, or as part of a Csound performance by including the -C flag in its arguments:

```
cscore [scorefilein] [> scorefileout]
```

(where *cscore* is the name of your user-written program), or

```
csound [-C] [otherflags] [orchname] [scorename]
```

The available API functions augment the C language library functions; they can read either standard numeric scores or pre-sorted score files, can massage and expand the data in various ways, then make it available for performance by a Csound orchestra.

The user-written control program is written in C, and is compiled and linked to the Csound library (or the *csound* commandline program) by the user. It is not essential to know the C language well to write this program, since the function calls have a simple syntax, and are powerful enough to do most of the complicated work. Additional power can come from C later as the need arises.

The following sections explain all of the steps needed to make use of *Cscore*:

- *Events, Lists, and Operations* - Explains the syntax of *Cscore* functions and data structures.
- *Writing a Cscore Control Program* - Illustrates by example how to write your own control program.
- *Compiling a Cscore Program* - Outlines the steps for compiling and linking with the Csound library.
- *More Advanced Examples* - Addresses advanced issues such as multiple input scores and the details of running *Cscore* inside of a Csound performance.

Events, Lists, and Operations

An event in *Cscore* is equivalent to one statement of a *standard numeric score* or a time-warped score (the format in which Csound writes a sorted score -- see any *score.srt*), and is stored internally in time-warped format. It is important to note that when *Cscore* is used in standalone-mode, it cannot understand any of the non-numeric "conveniences" that Csound allows in the input score format. Therefore, scores making use of features such as carry, ramp, expressions, and others will have to either be sorted first with the *scsort* utility or used with a modified *Csound* executable that contains the user's *Cscore* program. Score opcodes with macro arguments (r, m, n, and {}) are not understood.

Score events are each read in from an existing score file and stored in a C structure. The structures main components are an opcode and an array of pfield values. *Cscore* handles reading the events and storing them in memory for you. The format of the structure starts as follows:

```
typedef struct {
    CSHDR h;          /* space-managing header */
    char *strarg;      /* address of optional string argument */
    char op;           /* opcode-t, w, f, i, a, s or e */
    short pcnt;
```

```
MYFLT p2orig; /* unwarped p2, p3 */
MYFLT p3orig;
MYFLT p[1]; /* array of pfields p0, p1, p2 ... */
} EVENT;
```

MYFLT is either the C type *float* or *double* depending on how your copy of the Csound library was compiled. You should just declare any floating-point variables as MYFLT in your user program for compatibility.

Any *Cscore* function that creates, reads, or copies an event will return a pointer to the storage structure holding the event data. The event pointer can be used to access any component of the structure, in the form of *e->op* or *e->p[n]*. Each newly stored event will give rise to a new pointer, and a sequence of new events will generate a sequence of distinct pointers that must themselves be stored. Groups of event pointers are stored in an event list, which has its own structure:

```
typedef struct {
    CSHDR h;
    int nslots; /* max events in this event list */
    int nevents; /* number of events present */
    EVENT *e[1]; /* array of event pointers e0, e1, e2.. */
} EVLIST;
```

Any *Cscore* function that creates or modifies a list will return a pointer to the new list. The list pointer can be used to access any of its component event pointers, in the form of *a->e[n]*. Event pointers and list pointers are thus primary tools for manipulating the data of a score file. Pointers and lists of pointers can be copied and reordered without modifying the data values they refer to. This means that notes and phrases can be copied and manipulated from a high level of control. Alternatively, the data within an event or group of events can be modified without changing the event or list pointers. The *Cscore* API functions enable scores to be created and manipulated in this way.

With Csound 5, the names of all of the *Cscore* API functions have changed to be more explicit. In addition, each function now requires a pointer to a CSOUND object as its first argument. The structure of the CSOUND object is unimportant (and indeed cannot be modified in a user program). How to obtain this CSOUND pointer will be shown in the next section. The *Cscore* functions and data structures are available in the `cscore.h` header file, which you must include in your program code before you can use them.

The names of the *Cscore* functions specify whether they operate on single events or event lists. In the following summary of available function calls, some simple naming conventions are used:

```
The symbol cs is a pointer to a CSOUND object (CSOUND *);
The symbols e, f are pointers to events (notes);
The symbols a, b are pointers to lists (arrays) of such events;
The symbol n is an integer parameter of type int;
"..." indicates a string parameter (either a constant or variable of type char *);
The symbol fp is a score input stream file pointer (FILE *);
```

calling syntax	description
-----	-----
/* Functions for working with single events */	
e = cscoreCreateEvent(cs, n);	create a blank event with n pfields
e = cscoreDefineEvent(cs, "...");	defines an event as per the character string ...
e = cscoreCopyEvent(cs, f);	make a new copy of event f
e = cscoreGetEvent(cs);	read the next event in the score input file
cscorePutEvent(cs, e);	write event e to the score output file
cscorePutString(cs, "...");	write the string-defined event to score output
/* Functions for working with event lists */	
a = cscoreListCreate(cs, n);	create an empty event list with n slots
a = cscoreListAppendEvent(cs, a, e);	append event e to list a
a = cscoreListAppendStringEvent(cs, a, "...");	append a string-defined event to list a;
a = cscoreListCopy(cs, b);	copy the list b (but not the events)
a = cscoreListCopyEvents(cs, b);	copy the events of b, making a new list

<pre>a = cscoreListGetSection(cs); a = cscoreListGetNext(cs, nbeats); a = cscoreListGetUntil(cs, beatno); a = cscoreListSeparateF(cs, b); a = cscoreListSeparateTWF(cs, b); a = cscoreListAppendList(cs, a, b); a = cscoreListConcatenate(cs, a, b); cscoreListSort(cs, a); n = cscoreListCount(cs, a); a = cscoreListExtractInstruments(cs, b, "..."); a = cscoreListExtractTime(cs, b, from, to); cscoreListPut(cs, a); cscoreListPlay(cs, a); /* Functions for reclaiming memory */ cscoreFreeEvent(cs, e); cscoreListFree(cs, a); cscoreListFreeEvents(cs, a); /* Functions for working with multiple input score files */ fp = cscoreFileGetCurrent(cs); fp = cscoreFileOpen(cs, "filename"); cscoreFileSetCurrent(cs, fp); cscoreFileClose(cs, fp);</pre>	<pre>read all events from score input, up to next s or e read next nbeats beats from score input (nbeats is read all events from score input up to beat beatno separate the f statements from list b into list a separate the t,w & f statements from list b into li append the list b onto the list a concatenate (append) the list b onto the list a (sa sort the list a into chronological order by p[2] returns the number of events in list a extract notes of instruments ... (no new events) extract notes of time-span, creating new events (fr write the events of list a to the score output file send events of list a to the Csound orchestra for immediate performance (or print events if no orches release the space of event e release the space of list a (but not the events) release the events of list a, and the list space get the currently active input scorefile pointer (initially finds the command-line input scorefile p open another input scorefile (maximum of 5) make fp the currently active scorefile pointer close the scorefile relating to FILE *fp</pre>
--	---

Under Csound 4, the function names and parameters were as follows:

calling syntax	description
-----	-----
<pre>e = createv(n);</pre>	create a blank event with n pfields
<pre>e = defev("...");</pre>	defines an event as per the character string ...
<pre>e = copyev(f);</pre>	make a new copy of event f
<pre>e = getev();</pre>	read the next event in the score input file
<pre>putev(e);</pre>	write event e to the score output file
<pre>putstr("...");</pre>	write the string-defined event to score output
<pre>a = lcreat(n);</pre>	create an empty event list with n slots
<pre> int n;</pre>	
<pre>a = lappev(a,e);</pre>	append event e to list a
<pre>a = lappstrev(a,"...");</pre>	append a string-defined event to list a;
<pre>a = lcopy(b);</pre>	copy the list b (but not the events)
<pre>a = lcopyev(b);</pre>	copy the events of b, making a new list
<pre>a = lget();</pre>	read all events from score input, up to next s or e
<pre>a = lgetnext(nbeats);</pre>	read next nbeats beats from score input
<pre> float nbeats;</pre>	
<pre>a = lgetuntil(beatno);</pre>	read all events from score input up to beat beatno
<pre> float beatno;</pre>	
<pre>a = lsepf(b);</pre>	separate the f statements from list b into list a
<pre>a = lseptwf(b);</pre>	separate the t,w & f statements from list b into list a
<pre>a = lcat(a,b);</pre>	concatenate (append) the list b onto the list a
<pre>lsort(a);</pre>	sort the list a into chronological order by p[2]
<pre>a = lxins(b,"...");</pre>	extract notes of instruments ... (no new events)
<pre>a = lxtimev(b,from,to);</pre>	extract notes of time-span, creating new events
<pre> float from, to;</pre>	
<pre>lput(a);</pre>	write the events of list a to the score output file
<pre>lplay(a);</pre>	send events of list a to the Csound orchestra for immediate performance (or print events if no orchestra)
<pre>releev(e);</pre>	release the space of event e
<pre>lrel(a);</pre>	release the space of list a (but not the events)
<pre>lreleev(a);</pre>	release the events of list a, and the list space
<pre>fp = getcurfp();</pre>	get the currently active input scorefile pointer (initially finds the command-line input scorefile pointer)
<pre>fp = filopen("filename");</pre>	open another input scorefile (maximum of 5)
<pre>setcurfp(fp);</pre>	make fp the currently active scorefile pointer
<pre>filclose(fp);</pre>	close the scorefile relating to FILE *fp

Writing a Cscore Control Program

The general format for a *Cscore* control program is:

```
#include "cscore.h"
void cscore(CSOUND *cs)
```

```
{
/*  VARIABLE DECLARATIONS  */
/*  PROGRAM BODY  */
}
```

The *include* statement will define the event and list structures and all of the *Cscore* API functions for the program. The name of the user function needs to be *cscore* if it will be linked with the standard main program in *cscormai.c* or linked as the internal *Cscore* routine for a personal Csound executable. This *cscore()* function receives one argument from *cscormai.c* or Csound -- *CSOUND* **cs* -- which is a pointer to a Csound object. The pointer *cs* must be passed as the first parameter to every *Cscore* API function that the program calls.

The following C program will read from a *standard numeric score*, up to (but not including) the first *s* or *e* statement, then write that data (unaltered) as output.

```
#include "cscore.h"
void cscore(CSOUND *cs)
{
    EVLIST *a;
    a = cscoreListGetSection(cs); /* a is allowed to point to an event list */
    cscoreListPut(cs, a); /* read events in, return the list pointer */
    cscorePutString(cs, "e"); /* write these events out (unchanged) */
    cscorePutString(cs, "e"); /* write the string e to output */
}
```

After execution of *cscoreListGetSection()*, the variable *a* points to a list of event addresses, each of which points to a stored event. We have used that same pointer to enable another list function -- *cscoreListPut()* -- to access and write out all of the events that were read. If we now define another symbol *e* to be an event pointer, then the statement

```
e = a->e[4];
```

will set it to the contents of the 4th slot in the *EVLIST* structure, *a*. The contents is a pointer to an event, which is itself comprised of an *array* of parameter field values. Thus the term *e->p[5]* will mean the value of parameter field 5 of the 4th event in the *EVLIST* denoted by *a*. The program below will multiply the value of that *pfield* by 2 before writing it out.

```
#include "cscore.h"
void cscore(CSOUND *cs)
{
    EVENT *e;
    EVLIST *a;
    a = cscoreListGetSection(cs); /* a pointer to an event */
    e = a->e[4]; /* read a score as a list of events */
    e->p[5] *= 2; /* point to event 4 in event list a */
    cscoreListPut(cs, a); /* find pfield 5, multiply its value by 2 */
    cscorePutString(cs, "e"); /* write out the list of events */
    cscorePutString(cs, "e"); /* add a "score end" statement */
}
```

Now consider the following score, in which *p[5]* contains frequency in Hz.

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
i 1 1 3 0 440 10000
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000
e
```


If this score were given to the preceding main program, the resulting output would look like this:

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
i 1 1 3 0 440 10000
i 1 4 3 0 512 10000      ; p[5] has become 512 instead of 256.
i 1 7 3 0 880 10000
e
```

Note that the 4th event is in fact the second note of the score. So far we have not distinguished between notes and function table setup in a numeric score. Both can be classed as events. Also note that our 4th event has been stored in *e[4]* of the structure. For compatibility with Csound *pfield* notation, we will ignore *p[0]* and *e[0]* of the event and list structures, storing *p1* in *p[1]*, event 1 in *e[1]*, etc. The *Cscore* functions all adopt this convention.

As an extension to the above, we could decide to use the same pointers *a* and *e* to examine each of the events in the list. Note that *e* was not set to the numeral 4, but to the location of the 4th slot in the list. To inspect *p5* of the previous event in the list, we need only redefine *e* with the assignment

```
e = a->e[3];
```

and reference the 5th slot of the *pfield* array using the expression

```
e->p[5]
```

More generally, we can use an integer variable as an index to the array *e[]*, and access each event in sequence by using a loop and incrementing the index. The number of events stored in an *EVLIST* is contained in the *nevents* member of the struct.

```
int index;      /* start with e[1] because e[0] is not used */
for (index = 1; index <= a->nevents; index++)
{
    e = a->e[index];
    /* do something with e */
}
```

The above example starts with *e[1]* and increases the index each time through the loop (*index++*) until it is greater than *a->nevents*, the index of the last event in the list. The statements inside the *for* loop do execute a final time when *index* equals *a->nevents*.

In the following program we will use the same input score. This time we will separate the *ftable* statements from the *note* statements. We will next write the three note-events stored in the list *a* to the output, then create a second score section consisting of the original pitch set and a transposed version of itself. This will bring about an octave doubling.

Here, our index to the array is *n* and we increment *n* as part of a *for* block which iterates *nevents* times, allowing one statement to act upon the same *pfield* of each successive event.

```
#include "cscore.h"
void cscore(CSOUND *cs)
{
    EVENT *e, *f;
    EVLIST *a, *b;
    int n;

    a = cscoreListGetSection(cs);      /* read score into event list "a" */
```

```
b = cscoreListSeparateF(cs, a);          /* separate f statements */
cscoreListPut(cs, b);                   /* write f statements out to score */
cscoreListFreeEvents(cs, b);            /* and release the spaces used */
e = cscoreDefineEvent(cs, "t 0 120");   /* define event for tempo statement */
cscorePutEvent(cs, e);                  /* write tempo statement to score */
cscoreListPut(cs, a);                   /* write the notes */
cscorePutString(cs, "s");                /* section end */
cscorePutEvent(cs, e);                  /* write tempo statement again */
b = cscoreListCopyEvents(cs, a);         /* make a copy of the notes in "a" */
for (n = 1; n <= b->nevents; n++)       /* iterate the following lines nevents times: */
{
    f = b->e[n];
    f->p[5] *= 0.5;                      /* transpose pitch down one octave */
}
a = cscoreListAppendList(cs, a, b);     /* now add these notes to original pitches */
cscoreListPut(cs, a);
cscorePutString(cs, "e");
}
```

The output of this program is:

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
t 0 120
i 1 1 3 0 440 10000
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000
s
t 0 120
i 1 1 3 0 440 10000
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000
i 1 1 3 0 220 10000
i 1 4 3 0 128 10000
i 1 7 3 0 440 10000
e
```

If the output is only being written to a file, then the unsorted order of the events is not a problem. The output is written to a file (or standard output) whenever the function *cscoreListPut()* is used. However, if this program were to be called during a Csound performance and the function *cscoreListPlay()* replaced *cscoreListPut()*, then the events would be sent to the orchestra instead of to a file and they should then be sorted beforehand by calling the function *cscoreListSort()*. The details of score output and playing when using *Cscore* from within Csound are described in the next section.

Next we extend the above program by using the *for* loop to look at *p[5]* and *p[6]*. In the original score *p[6]* denotes amplitude. To create a diminuendo in the added lower octave, which is independent from the original set of notes, a variable called *dim* will be used.

```
#include "cscore.h"
void cscore(CSOUND *cs)
{
    EVENT *e, *f;
    EVLIST *a, *b;
    int n, dim;                                /* declare two integer variables */

    a = cscoreListGetSection(cs);
    b = cscoreListSeparateF(cs, a);
    cscoreListPut(cs, b);
    cscoreListFreeEvents(cs, b);
    e = cscoreDefineEvent(cs, "t 0 120");
    cscorePutEvent(cs, e);
    cscoreListPut(cs, a);
    cscorePutString(cs, "s");
    cscorePutEvent(cs, e);
    b = cscoreListCopyEvents(cs, a);
    dim = 0;                                   /* initialize dim to 0 */
    for (n = 1; n <= b->nevents; n++)
    {
        f = b->e[n];
```

```
        f->p[6] -= dim;          /* subtract current value of dim */
        f->p[5] *= 0.5;          /* transpose pitch down one octave */
        dim += 2000;            /* increase dim for each note */
    }
    a = cscoreListAppendList(cs, a, b);    /* now add these notes to original pitches */
    cscoreListPut(cs, a);
    cscorePutString(cs, "e");
}
```

Using the same input score again, the output from this program is:

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
t 0 120
i 1 1 3 0 440 10000
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000
s
t 0 120
i 1 1 3 0 440 10000      ; Three original notes at
i 1 4 3 0 256 10000      ; beats 1,4 and 7 with no dim.
i 1 7 3 0 880 10000
i 1 1 3 0 220 10000      ; three notes transposed down one octave
i 1 4 3 0 128 8000       ; also at beats 1,4 and 7 with dim.
i 1 7 3 0 440 6000
e
```

In the following program the same three-note sequence will be repeated at various time intervals. The starting time of each group is determined by the values of the array *cue*. This time the *dim* will occur for each group of notes rather than each note. Note the position of the statement which increments the variable *dim* outside the inner *for* loop.

```
#include "cscore.h"
int cue[3] = {0,10,17};          /* declare an array of 3 integers */
void cscore(CSOUND *cs)
{
    EVENT *e, *f;
    EVLIST *a, *b;
    int n, dim, cuecount;          /* declare new variable cuecount */

    a = cscoreListGetSection(cs);
    b = cscoreListSeparateF(cs, a);
    cscoreListPut(cs, b);
    cscoreListFreeEvents(cs, b);
    e = cscoreDefineEvent(cs, "t 0 120");
    cscorePutEvent(cs, e);
    dim = 0;
    for (cuecount = 0; cuecount <= 2; cuecount++) /* elements of cue are numbered 0, 1, 2 */
    {
        for (n = 1; n <= a->nevents; n++)
        {
            f = a->e[n];
            f->p[6] -= dim;
            f->p[2] += cue[cuecount];    /* add values of cue */
        }
        printf("; diagnostic: cue = %d\n", cue[cuecount]);
        dim += 2000;
        cscoreListPut(cs, a);
    }
    cscorePutString(cs, "e");
}
```

Here the inner *for* loop looks at the events of list *a* (the notes) and the outer *for* loop looks at each *repetition* of the events of list *a* (the pitch group "cues"). This program also demonstrates a useful trouble-shooting device with the *printf* function. The *semi-colon* is first in the character string to produce a comment statement in the resulting score file. In this case the value of *cue* is being printed in the output to insure that the program is taking the proper *array* member at the proper time. When output data is wrong or error messages are encountered, the *printf* function can help to pinpoint the

problem.

Using the same input file, the C program above will generate the following score. Can you determine why the last set of notes starts at the wrong time and how to correct the problem?

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
t 0 120
; diagnostic: cue = 0
i 1 1 3 0 440 10000
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000
; diagnostic: cue = 10
i 1 11 3 0 440 8000
i 1 14 3 0 256 8000
i 1 17 3 0 880 8000
; diagnostic: cue = 17
i 1 28 3 0 440 4000
i 1 31 3 0 256 4000
i 1 34 3 0 880 4000
e
```

Compiling a Cscore Program

A *Cscore* program can be invoked either as a *standalone program* or as part of Csound in between sorting the score and performing the score with the orchestra:

```
cscore [scorefilein] [> scorefileout]
```

or

```
csound [-C] [otherflags] [orchname] [scorename]
```

Before trying to compile your own *Cscore* program, you will most likely want to obtain a copy of the Csound source code. Either download the latest source distribution for your platform or check out a copy of the csound5 module from Sourceforge CVS. There are several files in the sources that will help you. Within the `examples/cscore/` directory are a number of examples of *Cscore* control programs, including all of the examples contained in this manual. And in the `frontends/cscore/` directory are the two files *cscoremain.c* and *cscore.c*. *cscoremain.c* contains a simple *main* function that performs all of the initialization that a standalone *Cscore* program needs to do before it calls your control function. This main “stub” initializes Csound, reads the commandline arguments, opens the input and output score files, and then calls a function *cscore()*. As described above, it is expected that you will write the *cscore()* function and provide it in another file. The file `frontends/cscore/cscore.c` shows the simplest example of a *cscore()* function that reads in a score of any length and writes it to the output unchanged.

So, to create a standalone program, write a control program as shown in the previous section. Let's assume that you saved this program in a file named “*mycscore.c*”. Next, you need to compile and link this program with the Csound library and *cscoremain.c* in order to create an executable by following the set of directions below that apply to your operating system. It will be helpful to already have some familiarity with the C compiler on your computer since the information below cannot be complete for all possible systems.

Linux and Unix

The following commands assume that you have copied your file *mycscore.c* into the same directory as *cscoremain.c*, that you have opened a terminal to that same directory, and that you have previously installed a binary distribution of Csound that placed a library *libcsound.a* or *libcsound.so* into `/usr/local/lib` and the header files for the Csound API into `/usr/local/include/csound`.

To compile and link:

```
gcc mycscore.c cscoremain.c -o cscore -lcsound -L/usr/local/lib -I/usr/local/include/csound
```

To run (sending the results to standard output):

```
./cscore test.sco
```

It is possible that on some Unix systems, the C compiler will be named *cc* or something else other than *gcc*.

Windows

Csound is usually compiled on Windows using the MinGW environment that makes GCC -- the same compiler used on Linux -- available using a Unix-like command shell (MSYS). Since pre-compiled libraries for Csound on Windows are built in this way, you may need to use MinGW as well to link to them. If you have built Csound using another compiler, then you should be able to build *Cscore* with that compiler as well.

Compiling standalone *Cscore* programs using MinGW should be similar to the procedure for Linux above with library and header paths changed appropriately for where Csound is installed on the Windows system. *(Please feel free to contribute more detailed instructions here as the editor has been unable to test Cscore on a Windows machine).*

OS X

The following commands assume that you have copied your file *mycscore.c* into the same directory as *cscoremain.c* and that you have opened a terminal to that same directory. In addition, the Apple-supplied developer tools (including the GCC compiler) should be installed on your system and you should have previously installed a binary distribution of Csound that placed the CsoundLib framework into */Library/Frameworks*.

Use this command compile and link. (You may get a warning about "multiple definitions of symbol *_cscore*").

```
gcc cscore.c cscoremain.c -o cscore -framework CsoundLib -I/Library/Frameworks/CsoundLib.framework/
```

To run (sending the results to standard output):

```
./cscore test.sco
```

MacOS 9

You will need CodeWarrior or some other development environment installed on your computer (MPW may work). Download the source code distribution for OS 9 (it will have a name like *Csound5.05_OS9_src.smi.bin*).

If using CodeWarrior, find and open the project file "Cscore5.cw8.mcp" in the folder "Csound5.04-OS9-source:macintosh:Csound5Library:". This project file is configured to use the source files *cscore.c* and *cscoremain_MacOS9.c* from the csound5 source tree and the Csound5Lib shared library produced by compiling Csound with the "Csound5.cw8.mcp" project file. You should substitute your own *Cscore* program file for *cscore.c* and either compile Csound5Lib first or substitute a copy of the library in the project from the binary distribution of Csound for OS 9. The file

cscoremain_MacOS9.c contains specialized code for configuring CodeWarrior's SIOUX console library and allows commandline arguments to be entered before the program is run.

Once you have the proper files included in the project window, click the "Make" button and CodeWarrior should produce an application named "Cscore". When you run this application, it first displays a window allowing you to type in the arguments to the main function. You only need to type in the filename or pathname to the input score -- do not type in "cscore". The input file should be in the same folder as the application or else you will need to type a full or relative pathname to the file. Output will be displayed in the console window. You can use the *Save* command from the *File* menu before quitting if you wish. Alternatively, in the commandline dialog, you can choose to redirect the output to a file by clicking on the *File* button on the right side of the dialog. (Note that the console window can only display about 32,000 characters, so writing to a file is necessary for long scores).

Making Cscore usable from within Csound

To operate from Csound, first follow the instructions for compiling Csound (see *Building Csound*) according to the operating system that you are using. Once you have successfully built an unmodified Csound system, then substitute your own *cscore()* function for the one in the file *Top/cscore_internal.c*, and rebuild Csound.

The resulting executable is your own special Csound, usable as above. The *-C* flag will invoke your *Cscore* program after the input score is sorted into "score.srt". The details of what happens when you run Csound with the *-C* flag are given in the next section.

Csound 5 also provides an additional way to run your own *Cscore* program from within Csound. Using the API, a host application can set a *Cscore callback function*, which is a function that Csound will call instead of using the built-in *cscore()* function. One advantage of this approach is that it is not necessary to recompile the entirety of Csound. Another benefit is that the host application can select at runtime from more than one *Cscore* function to designate as the callback. The disadvantage is that you need to write a host application.

A simple approach to using a *Cscore* callback via the API would be to modify the standard Csound main program -- which is a simple Csound host -- contained in the file *frontends/csound/csound_main.c*. Adding a call to *csoundSetCscoreCallback()* after the call to *csoundCreate()* but before the call to *csoundCompile()* should do the job. Recompiling this file and linking to an existing Csound library will make a commandline version of Csound that works similarly to the one described above. Don't forget to use the *-C* flag.

Notes about score formats and run-time behavior

As stated previously, the input files to *Cscore* may be in original or time-warped and pre-sorted form; this modality will be preserved (section by section) in reading, processing, and writing scores. Standalone processing will most often use unwarped sources and create unwarped new files. When running from within Csound, the input score will arrive already warped and sorted, and can thus be sent directly (normally section by section) to the orchestra. One advantage of this method of using *Cscore* is that all of the syntactical conveniences of the full Csound score language may be used -- macros, arithmetic expressions, carry, ramp, etc. -- since the score will go through the "Carry, Tempo, Sort" phases of score processing before being passed to the user-supplied *Cscore* program.

When running within Csound, a list of events can be conveyed to a Csound orchestra using *cscoreListPlay()*. There may be any number of *cscoreListPlay()* calls in a *Cscore* program. Each list so conveyed can be either time-warped or not, but each list must be in strict *p2*-chronological order (either from presorting or using *cscoreListSort()*). If there is no *cscoreListPlay()* in a *Cscore* module run from within Csound, all events written out (via *cscorePutEvent()*, *cscorePutString()*, or *cscoreListPut()*) are written to a new score in the current directory with the name "cscore.out". Csound then invokes the score sorter again before sending this new score to the orchestra for performance. The final, sorted, output score is written to a file named "cscore.srt".

A standalone *Cscore* program will normally use the "put" commands to write into its output file. If a standalone *Cscore* program calls *cscoreListPlay()*, the events thus intended for performance will be sent to the output in the same way as if *cscoreListPut()* had been called instead.

A note list sent by *cscoreListPlay()* for performance should be temporally distinct from subsequent note lists. No note-end should extend past the next list's start time, since *cscoreListPlay()* will complete each list before starting the next (i.e. like a Section marker that doesn't reset local time to zero). This is important when using *cscoreListGetNext()* or *cscoreListGetUntil()* to fetch and process score segments prior to performance, because these functions may only read part of an unsorted section.

More Advanced Examples

The following program demonstrates reading from two different input files. The idea is to switch between two 2-section scores, and write out the interleaved sections to a single output file.

```
#include "cscore.h"                /* CSCORE_SWITCH.C */
cscore(CSOUND* cs)                /* callable from either Csound or standalone cscore */
{
    EVLIST *a, *b;
    FILE *fp1, *fp2;               /* declare two scorefile stream pointers */
    fp1 = cscoreFileGetCurrent(cs); /* this is the command-line score */
    fp2 = cscoreFileOpen(cs, "score2.srt"); /* this is an additional score file */
    a = cscoreListGetSection(cs);    /* read section from score 1 */
    cscoreListPut(cs, a);            /* write it out as is */
    cscorePutString(cs, "s");
    cscoreFileSetCurrent(cs, fp2);
    b = cscoreListGetSection(cs);    /* read section from score 2 */
    cscoreListPut(cs, b);            /* write it out as is */
    cscorePutString(cs, "s");
    cscoreListFreeEvents(cs, a);     /* optional to reclaim space */
    cscoreListFreeEvents(cs, b);
    cscoreFileSetCurrent(cs, fp1);
    a = cscoreListGetSection(cs);    /* read next section from score 1 */
    cscoreListPut(cs, a);            /* write it out */
    cscorePutString(cs, "s");
    cscoreFileSetCurrent(cs, fp2);
    b = cscoreListGetSection(cs);    /* read next sect from score 2 */
    cscoreListPut(cs, b);            /* write it out */
    cscorePutString(cs, "e");
}
```

Finally, we show how to take a literal, uninterpreted score file and imbue it with some expressive timing changes. The theory of composer-related metric pulses has been investigated at length by Manfred Clynes, and the following is in the spirit of his work. The strategy here is to first create an *array* of new *onset* times for every possible sixteenth-note onset, then to index into it so as to adjust the start and duration of each note of the input score to the interpreted time-points. This also shows how a Csound orchestra can be invoked repeatedly from a run-time score generator.

```
#include "cscore.h"                /* CSCORE_PULSE.C */

/* program to apply interpretive durational pulse to */
/* an existing score in 3/4 time, first beats on 0, 3, 6 ... */

static float four[4] = { 1.05, 0.97, 1.03, 0.95 }; /* pulse width for 4's */
static float three[3] = { 1.03, 1.05, .92 };       /* pulse width for 3's */

cscore(CSOUND* cs)                /* This example should be called from Csound */
{
    EVLIST *a, *b;
    EVENT *e, **ep;
    float pulse16[4*4*4*4*3*4]; /* 16th-note array, 3/4 time, 256 measures */
    float acc16, acc1, inc1, acc3, inc3, acc12, inc12, acc48, inc48, acc192, inc192;
    float *p = pulse16;
    int n16, n1, n3, n12, n48, n192;

    /* fill the array with interpreted ontimes */
    for (acc192=0., n192=0; n192<4; acc192+=192.*inc192, n192++)
        for (acc48=acc192, inc192=four[n192], n48=0; n48<4; acc48+=48.*inc48, n48++)
            for (acc12=acc48, inc48=inc192*four[n48], n12=0; n12<4; acc12+=12.*inc12, n12++)
                for (acc3=acc12, inc12=inc48*four[n12], n3=0; n3<4; acc3+=3.*inc3, n3++)
                    for (acc1=acc3, inc3=inc12*four[n3], n1=0; n1<3; acc1+=inc1, n1++)
```

```
        for (acc16=acc1,inc1=inc3*three[n1],n16=0; n16<4; acc16+=.25*inc1*four[n16],n16++)
            *p++ = acc16;

/* for (p = pulsel6, n1 = 48; n1--; p += 4) /* show vals & diffs */
/*   printf("%g %g %g %g %g %g %g %g\n", *p, *(p+1), *(p+2), *(p+3),
/*   *(p+1)-*p, *(p+2)-*(p+1), *(p+3)-*(p+2), *(p+4)-*(p+3)); */

a = cscoreListGetSection(cs); /* read sect from tempo-warped score */
b = cscoreListSeparateTWF(cs, a); /* separate warp & fn statements */
cscoreListPlay(cs, b); /* and send these to performance */
a = cscoreListAppendStringEvent(cs, a, "s"); /* append a sect statement to note list */
cscoreListPlay(cs, a); /* play the note-list without interpretation */
for (ep = &a->e[1], n1 = a->nevents; n1--; ) { /* now pulse-modifiy it */
    e = *ep++;
    if (e->op == 'i') {
        e->p[2] = pulsel6[(int)(4. * e->p2orig)];
        e->p[3] = pulsel6[(int)(4. * (e->p2orig + e->p3orig))] - e->p[2];
    }
}

cscoreListPlay(cs, a); /* now play modified list */
}
```

Extending Csound

Adding Unit Generators

If the existing Csound unit generators do not suit your needs, it is relatively easy to extend Csound by writing new unit generators in C or C++. The translator, loader, and run-time monitor will treat your module just like any other, provided you follow some conventions.

Historically, this has been done with builtin unit generators, that is, with code that is statically linked with the rest of the Csound executable.

Today, the preferred method is to create plugin unit generators. These are dynamic link libraries (DLLs) on Windows, and loadable modules (shared libraries that are `dlopened`) on Linux. Csound searches for and loads these plugins at run time. The advantage of this method, of course, is that plugins created by any developer at any time can be used with already existing versions of Csound.

Creating a Builtin Unit Generator

You need a structure defining the inputs, outputs and workspace, plus some initialization code and some perf-time code. Let's put an example of these in two new files, `newgen.h` and `newgen.c`. The examples given are for Csound 5. For earlier versions, all opcode functions omit the first parameter (`CSOUND *csound`).

```
/* newgen.h - define a structure */

/* Declares Csound structures and functions. */
#include "csoundCore.h"

typedef struct
{
    OPDS h;
    MYFLT *result, *istrt, *incr, *itime, *icontin; /* required header */
    MYFLT curval, vincr; /* addr outarg, inargs */
    long countdown; /* private dataspace */
} RMP; /* ditto */

/* newgen.c - init and perf code */
/* Declares Csound structures and functions. */
#include "csoundCore.h"
/* Declares RMP structure. */
#include "newgen.h"

int rampset (CSOUND *csound, RMP * p) /* at note initialization: */
{
    if (*p->icontin == FL(0.0))
        p->curval = *p->istrt; /* optionally get new start value */
    p->vincr = *p->incr / csound->esr; /* set s-rate increment per sec. */
    p->countdown = *p->itime * csound->esr; /* counter for itime seconds */
    return OK;
}

int ramp (CSOUND *csound, RMP * p) /* during note performance: */
{
    MYFLT *rsltp = p->result; /* init an output array pointer */
    int nn = csound->ksmps; /* array size from orchestra */
    do
    {
        *rsltp++ = p->curval; /* copy current value to output */
        if (--p->countdown > 0) /* for the first itime seconds, */
            p->curval += p->vincr; /* ramp the value */
    }
    while (--nn);
    return OK;
}
```

Now we add this module to the translator table in `entry1.c`, under the opcode name `rampt`:

```
#include "newgen.h"

int rampset(CSOUND *, RMP *), ramp(CSOUND *, RMP *);

/*  opname      dsblksiz  thread   outtypes  intypes  iopadr   kopadr   aopadr  */
{ "rampt",  S(RMP),   5,         "a",      "iiio",   (SUBR) rampset, (SUBR) NULL, (SUBR) ramp  },
```

Finally you must relink Csound with the new module. Add the name of the C file to the `libCsoundSources` list in the `SConstruct` file:

```
libCsoundSources = Split('''
Engine/auxfd.c
...
OOps/newgen.c
...
Top/utility.c
''')
```

Run `scons` just as you would for any other Csound build, and the new module will be built into your Csound.

The above actions have added a new generator to the Csound language. It is an audio-rate linear ramp function which modifies an input value at a user-defined slope for some period. A ramp can optionally continue from the previous note's last value. The Csound manual entry would look like:

```
ar rampt istart, islope, itime [, icontin]
```

istart -- beginning value of an audio-rate linear ramp. Optionally overridden by a continue flag.

islope -- slope of ramp, expressed as the y-interval change per second.

itime -- ramp time in seconds, after which the value is held for the remainder of the note.

icontin (optional) -- continue flag. If zero, ramping will proceed from input *istart* . If non-zero, ramping will proceed from the last value of the previous note. The default value is zero.

The file `newgen.h` includes a one-line list of output and input parameters. These are the ports through which the new generator will communicate with the other generators in an instrument. Communication is by *address*, not *value*, and this is a list of pointers to values of type `MYFLT` (which is double if the macro `USE_DOUBLE` is defined, and float otherwise). There are no restrictions on names, but the input-output argument types are further defined by character strings in `entry1.c` (inargs, outargs). Inarg types are commonly *x*, *a*, *k*, and *i*, in the normal Csound manual conventions; also available are *o* (optional, defaulting to 0), *p* (optional, defaulting to 1). Outarg types include *a*, *k*, *i* and *s* (asig or ksig). It is important that all listed argument names be assigned a corresponding argument type in `entry1.c`. Also, *i*-type args are valid only at initialization time, and other-type args are available only at perf time. Subsequent lines in the `RMP` structure declare the work space needed to keep the code re-entrant. These enable the module to be used multiple times in multiple instrument copies while preserving all data.

The file `newgen.c` contains two subroutines, each called with a pointer to the Csound instance and a pointer to the uniquely allocated `RMP` structure and its data. The subroutines can be of three types: note initialization, *k*-rate signal generation, *a*-rate signal generation. A module normally requires two of these: initialization, and either *k*-rate or *a*-rate subroutines which become inserted in various threaded lists of runnable tasks when an instrument is activated. The thread-types appear in `entry1.c` in two forms: *isub*, *ksub* and *asub* names; and a threading index which is the sum of *isub*=1, *ksub*=2, *asub*=4. The code itself may reference (but should only read) public members of the `CSOUND` structure defined in `csoundCore.h`, the most useful of which are:

OPARMS	*oparms		
MYFLT	esr	user-defined	sampling rate
MYFLT	ekr	user-defined	control rate
int	ksmps	user-defined	ksmps
int	nchnls	user-defined	nchnls
int	oparms->odebug	command-line	-v flag
int	oparms->msglevel	command-line	-m level
MYFLT	tpidsr	2 * PI /	esr

Function tables

To access stored function tables, special help is available. The newly defined structure should include a pointer

```
FUNC          *ftp;
```

initialized by the statement

```
ftp = csound->FTFind(csound, p->ifuncno);
```

where MYFLT *ifuncno is an i-type input argument containing the ftable number. The stored table is then at ftp->ftable, and other data such as length, phase masks, cps-to-incr converters, are also accessed from this pointer. See the FUNC structure in csoundCore.h, the csoundFTFind() code in fgens.c, and the code for oscset() and koscil() in OOps/ugens2.c.

Additional Space

Sometimes the space requirement of a module is too large to be part of a structure (upper limit 65279 bytes, due to the unsigned short dsblksiz parameter and reserved codes $\geq 0xFF00$), or it is dependent on an i-arg value which is not known until initialization. Additional space can be dynamically allocated and properly managed by including the line

```
AUXCH          auxch;
```

in the defined structure (*p), then using the following style of code in the init module:

```
csound->AuxAlloc(csound, npoints * sizeof(MYFLT), &p->auxch);
```

The address of this auxiliary space is kept in a chain of such spaces belonging to this instrument, and is automatically managed while the instrument is being duplicated or garbage-collected during performance. The assignment

```
void *auxp = p->auxch.auxp;
```

will find the allocated space for init-time and perf-time use. See the LINSEG structure in ugens1.h and the code for lsgset() and klnseg() in OOps/ugens1.c.

File Sharing

When accessing an external file often, or doing it from multiple places, it is often efficient to read

the entire file into memory. This is accomplished by including the line

```
MEMFIL      *mfp;
```

in the defined structure (*p), then using the following style of code in the init module:

```
p->mfp = csound->ldmemfile(csound, filename);
```

where char *filename is a string name of the file requested. The data read will be found between

```
(char *) p->mfp->beginp; and (char *) p->mfp->endp;
```

Loaded files do not belong to a particular instrument, but are automatically shared for multiple access. See the ADSYN structure in ugens3.h and the code for adset() and adsyn() in OOps/ugens3.c.

String arguments

To permit a string input argument (MYFLT *ifilnam, say) in our defined structure (*p), assign it the argtype *S* in entry1.c, and include the following code in the init module:

```
strcpy(filename, (char*) p->ifilnam);
```

See the code for adset() in OOps/ugens3.c, lprdset() in OOps/ugens5.c, and pvset() in OOps/ugens8.c.

Adding a Plugin Unit Generator

The procedure for creating a plugin unit generator is very similar to the procedure for creating a builtin. The actual unit generator code would normally be identical. The differences are as follows.

Again supposing that your unit generator is named newgen, perform the following steps:

1. Write your newgen.c and newgen.h file as you would for a builtin unit generator. Put these files in the csound5/Opcodes directory.
2. #include "csdl.h" in your unit generator sources, instead of csoundCore.h.
3. Add your OENTRY records and unit generator registration functions at the bottom of your C file. Example (but you can have as many unit generators in one plugin as you like):

```
#define S sizeof
static OENTRY localops[] = {
{
    { "rampt", S(RMP), 5, "a", "iiio", (SUBR) rampset, (SUBR) NULL, (SUBR) ramp },
};
/*
 * The following macro from csdl.h defines
 * the "csound_opcode_init()" opcode registration
 * function for the localops table.
 */
LINKAGE
```

4. Add your plugin as a new target in the plugin opcodes section of the SConstruct build file:

```
pluginEnvironment.SharedLibrary('newgen',
    Split('' 'Opcodes/newgen.c
    Opcodes/another_file_used_by_newgen.c
    Opcodes/yet_another_file_used_by_newgen.c'''))
```

5. Run the Csound 5 build in the regular way.

OENTRY Reference

The OENTRY structure (see `H/csoundCore.h`, `Engine/entry1.c`, and `Engine/rdorch.c`) contains the following public fields:

`opname`, `dsblksiz`, `thread`, `outtypes`, `intypes`, `iopadr`, `kopadr`, `aopadr`

dsblksiz There are two types of opcodes, polymorphic and non-polymorphic. For non-polymorphic opcodes, the `dsblksiz` flag specifies the size of the opcode structure in bytes, and arguments are always passed to the opcode at the same rate. Polymorphic opcodes can accept arguments at different rates, and those arguments are actually dispatched to other opcodes as determined by the `dsblksiz` flag and the following naming convention (note: the following list is not complete, see `Engine/entry1.c` for all possible special `dsblksiz` codes):

`0xffff` The type of the first output argument determines which unit generator function is actually called: `xxx -> xxx.a`, `xxx.i`, or `xxx.k`.

`0xfffe` The types of the first two input arguments determine which unit generator function is actually called: `xxx -> xxx.aa`, `xxx.ak`, `xxx.ka`, or `xxx.kk`, as in the `oscil` unit generator.

`0xfffd` Refers to one input argument of type `a` or `k`, as in the `peak` unit generator.

thread Specifies the rate(s) at which the unit generator's functions are called, as follows:

Table 18. Rate at which ugens are called according to thread parameter

0	i-rate <i>or</i> k-rate (B out only)
1	i-rate
2	k-rate
3	i-rate <i>and</i> k-rate
4	a-rate
5	i-rate <i>and</i> a-rate
7	i-rate <i>and</i> (k-rate <i>or</i> a-rate)

outtypes Lists the return values of the unit generator functions, if any. The types allowed are (note: the following list is not complete, see `Engine/entry1.c` for all possible output types):

Table 19. List of out types for ugens

i	i-rate scalar
k	k-rate scalar

a	a-rate vector
x	k-rate scalar or a-rate vector
f	f-rate streaming pvoc fsig type
m	multiple a-rate output arguments

`intypes` Lists the arguments the unit generator functions take, if any. The types allowed are (note: the following list is not complete, see `Engine/entry1.c` for all possible input types):

Table 20. List of in types ofr ugens

i	i-rate scalar
k	k-rate scalar
a	a-rate vector
x	a-rate scalar or a-rate vector
f	f-rate streaming pvoc fsig type
S	String
B	
l	
m	Begins an indefinite list of i-rate arguments (any count)
M	Begins an indefinite list of arguments (any rate, any count)
N	Begins an indefinite list of (optional a-, k-, i-, or S-rate)-rate arguments (any odd count)
n	Begins an indefinite list of i-rate arguments (any odd count)
O	Optional k-rate, defaulting to 0
o	Optional i-rate, defaulting to 0
p	Optional i-rate, defaulting to 1
q	Optional i-rate, defaulting to 10
v	Optional k-rate, defaulting to 0.5
v	Optional i-rate, defaulting to 0.5
j	Optional i-rate, defaulting to -1
h	Optional i-rate, defaulting to 127
y	Begins an indefinite list of a-rate arguments (any count)
z	Begins an indefinite list of k-rate arguments (any count)
Z	Begins an indefinite list of alternating k-rate and a-rate arguments (kaka...) (any count)

`iopadr` The address of the unit generator function (of type `int (*SUBR)(CSOUND *, void *)`) that is called at i-time, or NULL for no function.

`kopadr` The address of the unit generator function (of type `int (*SUBR)(CSOUND *, void *)`) that is called at k-rate, or NULL for no function.

`aopadr` The address of the unit generator function (of type `int (*SUBR)(CSOUND *, void`

) that is called at a-rate, or NULL for no function.

Part IV. Opcode Quick Reference

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Opcode Quick Reference

Orchestra Syntax:Header.

```
0dbfs = iarg
```

```
0dbfs
```

```
kr = iarg
```

```
ksmps = iarg
```

```
nchnls = iarg
```

```
sr = iarg
```

Orchestra Syntax:Block Statements.

```
endin
```

```
endop
```

```
instr i, j, ...
```

```
opcode name, outtypes, intypes
```

Orchestra Syntax:Macros.

```
#define NAME # replacement text #
```

```
#define NAME(a' b' c') # replacement text #
```

```
$NAME
```

```
#ifdef NAME
```

```
....
```

```
#else
```

```

    ....

#end

#ifdef NAME

    ....

#else

    ....

#end

#include "filename"

#undef NAME

```

Signal Generators:Additive Synthesis/Resynthesis.

```

ares adsyn kamod, kfmod, ksmod, ifilcod

ares adsynt kamp, kcps, iwfn, ifreqfn, iampfn, icnt [, iphs]

ar adsynt2 kamp, kcps, iwfn, ifreqfn, iampfn, icnt [, iphs]

ares hsboscil kamp, ktone, kbrite, ibasfreq, iwfn, ioctfn \
    [, ioctcnt] [, iphs]

```

Signal Generators:Basic Oscillators.

```

kres lfo kamp, kcps [, itype]

ares lfo kamp, kcps [, itype]

ares oscbnk kcps, kamd, kfmd, kpmd, iovrlap, iseed, kl1minf, kl1maxf, \
    kl2minf, kl2maxf, ilfomode, kegminf, kegmaxf, kegminl, kegmaxl, \
    kegming, kegmaxg, iegmode, kfn [, il1fn] [, il2fn] [, iegffn] \
    [, ieglfn] [, iegqfn] [, itabl] [, ioutfn]

ares oscil xamp, xcps, ifn [, iphs]

kres oscil kamp, kcps, ifn [, iphs]

ares oscil3 xamp, xcps, ifn [, iphs]

```

```
kres oscil3 kamp, kcps, ifn [, iphs]

ares oscili xamp, xcps, ifn [, iphs]

kres oscili kamp, kcps, ifn [, iphs]

ares oscilikt xamp, xcps, kfn [, iphs] [, istor]

kres oscilikt kamp, kcps, kfn [, iphs] [, istor]

ares osciliktp kcps, kfn, kphs [, istor]

ares oscilikts xamp, xcps, kfn, async, kphs [, istor]

ares osciln kamp, ifrq, ifn, itimes

ares oscils iamp, icps, iphs [, iflg]

ares poscil aamp, acps, ifn [, iphs]

ares poscil aamp, kcps, ifn [, iphs]

ares poscil kamp, acps, ifn [, iphs]

ares poscil kamp, kcps, ifn [, iphs]

ires poscil kamp, kcps, ifn [, iphs]

kres poscil kamp, kcps, ifn [, iphs]

ares poscil3 kamp, kcps, ifn [, iphs]

kres poscil3 kamp, kcps, ifn [, iphs]

kout vibr kAverageAmp, kAverageFreq, ifn

kout vibrato kAverageAmp, kAverageFreq, kRandAmountAmp, \
    kRandAmountFreq, kAmpMinRate, kAmpMaxRate, kcpsMinRate, \
    kcpsMaxRate, ifn [, iphs]
```

Signal Generators:Dynamic Spectrum Oscillators.

```
ares buzz xamp, xcps, knh, ifn [, iphs]

ares gbuzz xamp, xcps, knh, klh, kmul, ifn [, iphs]
```

ares mpulse kamp, kintvl [, ioffset]

ares vco xamp, xcps, iwave, kpw [, ifn] [, imaxd] [, ileak] [, inyx] \
[, iphs] [, iskip]

ares vco2 kamp, kcps [, imode] [, kpw] [, kphs] [, inyx]

kfn vco2ft kcps, iwave [, inyx]

ifn vco2ift icps, iwave [, inyx]

ifn vco2init iwave [, ibasfn] [, ipmul] [, iminsiz] [, imaxsiz] [, isrcft]

Signal Generators:FM Synthesis.

ares fmb3 kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, \
ifn4, ivfn

ares fmbell kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, \
ifn4, ivfn

ares fmmetal kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, \
ifn4, ivfn

ares fmpercfl kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, \
ifn3, ifn4, ivfn

ares fmrhode kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, \
ifn3, ifn4, ivfn

ares fmvoice kamp, kfreq, kvowel, ktilt, kvibamt, kvibrate, ifn1, \
ifn2, ifn3, ifn4, ivibfn

ares fmwurlie kamp, kfreq, kc1, kc2, kvdepth, kvrate, ifn1, ifn2, ifn3, \
ifn4, ivfn

ares foscil xamp, kcps, xcar, xmod, kndx, ifn [, iphs]

ares foscili xamp, kcps, xcar, xmod, kndx, ifn [, iphs]

Signal Generators:Granular Synthesis.

asig diskgrain Sfname, kamp, kfreq, kpitch, kgrsize, kprate, \

```

    ifun, iolaps[, ioffset, imaxgrsize]

ares fof xamp, xfund, xform, koct, kband, kris, kdur, kdec, iolaps, \
    ifna, ifnb, itotdur [, iphs] [, ifmode] [, iskip]

ares fof2 xamp, xfund, xform, koct, kband, kris, kdur, kdec, iolaps, \
    ifna, ifnb, itotdur, kphs, kgliss [, iskip]

ares fog xamp, xdens, xtrans, aspd, koct, kband, kris, kdur, kdec, \
    iolaps, ifna, ifnb, itotdur [, iphs] [, itmode] [, iskip]

ares grain xamp, xpitch, xdens, kampoff, kpitchoff, kgdur, igfn, \
    iwfn, imgdur [, igrnd]

ares grain2 kcps, kfmd, kgdur, iovrlp, kfn, iwfn [, irpow] \
    [, iseed] [, imode]

ares grain3 kcps, kphs, kfmd, kpmd, kgdur, kdens, imaxovr, kfn, iwfn, \
    kfrpow, kprpow [, iseed] [, imode]

ares granule xamp, ivoice, iratio, imode, ithd, ifn, ipshift, igskip, \
    igskip_os, ilength, kgap, igap_os, kgsz, igsize_os, iatt, idec \
    [, iseed] [, ipitch1] [, ipitch2] [, ipitch3] [, ipitch4] [, ifnenv]

a1 [, a2, a3, a4, a5, a6, a7, a8] partikkel agrainfreq, \
    kdistribution, idisttab, async, kenv2amt, ienv2tab, ienv_attack, \
    ienv_decay, ksustain_amount, ka_d_ratio, kduration, kamp, igainmasks, \
    kwavfreq, ksweepshape, iwavfreqstarttab, iwavfreqendtab, awavfm, \
    ifmampstab, kfmenv, icosine, ktraincps, knumpartials, kchroma, \
    ichannelmasks, krandommask, kwaveform1, kwaveform2, kwaveform3, \
    kwaveform4, iwaveamptab, asamplepos1, asamplepos2, asamplepos3, \
    asamplepos4, kwavekey1, kwavekey2, kwavekey3, kwavekey4, imax_grains \
    [, iopcode_id]

async [,aphase] partikkelsync iopcode_id

ares [, ac] sndwarp xamp, xtimewarp, xresample, ifn1, ibeg, iwsz, \
    irandw, ioverlap, ifn2, itimemode

ar1, ar2 [,ac1] [, ac2] sndwarpst xamp, xtimewarp, xresample, ifn1, \
    ibeg, iwsz, irandw, ioverlap, ifn2, itimemode

asig syncgrain kamp, kfreq, kpitch, kgrsize, kprate, ifun1, \
    ifun2, iolaps

asig syncloop kamp, kfreq, kpitch, kgrsize, kprate, klstart, \
    klend, ifun1, ifun2, iolaps[,istart, iskip]

ar vosim kamp, kFund, kForm, kDecay, kPulseCount, kPulseFactor, ifn [, iskip]

```

Signal Generators:Hyper Vectorial Synthesis.

```
hvs1 kx, inumParms, inumPointsX, iOutTab, iPositionsTab, iSnapTab [, iConfigTab]
```

```
hvs2 kx, ky, inumParms, inumPointsX, iOutTab, iPositionsTab, iSnapTab [, iConfigTab]
```

```
hvs3 kx, ky, kz, inumParms, inumPointsX, iOutTab, iPositionsTab, iSnapTab [, iConfigTab]
```

Signal Generators:Linear and Exponential Generators.

```
kout expcurve kindex, ksteepness
```

```
ares expon ia, idur, ib
```

```
kres expon ia, idur, ib
```

```
ares expseg ia, idur1, ib [, idur2] [, ic] [...]
```

```
kres expseg ia, idur1, ib [, idur2] [, ic] [...]
```

```
ares expsega ia, idur1, ib [, idur2] [, ic] [...]
```

```
ares expsegr ia, idur1, ib [, idur2] [, ic] [...], irel, iz
```

```
kres expsegr ia, idur1, ib [, idur2] [, ic] [...], irel, iz
```

```
kout gainslider kindex
```

```
ares jspline xamp, kcpsMin, kcpsMax
```

```
kres jspline kamp, kcpsMin, kcpsMax
```

```
ares line ia, idur, ib
```

```
kres line ia, idur, ib
```

```
ares linseg ia, idur1, ib [, idur2] [, ic] [...]
```

```
kres linseg ia, idur1, ib [, idur2] [, ic] [...]
```

```
ares linsegr ia, idur1, ib [, idur2] [, ic] [...], irel, iz
```

```
kres linsegr ia, idur1, ib [, idur2] [, ic] [...], irel, iz
```

kout logcurve kindex, ksteepness

ksig loopseg kfreq, ktrig, ktime0, kvalue0 [, ktime1] [, kvalue1] \
[, ktime2] [, kvalue2] [...]

ksig loopsegg kphase, kvalue0, kdur0, kvalue1 \
[, kdurl, ... , kdurN-1, kvalueN]

ksig lpshold kfreq, ktrig, ktime0, kvalue0 [, ktime1] [, kvalue1] \
[, ktime2] [, kvalue2] [...]

ksig lpsholdp kphase, ktrig, ktime0, kvalue0 [, ktime1] [, kvalue1] \
[, ktime2] [, kvalue2] [...]

ares rspline xrangeMin, xrangeMax, kcpsMin, kcpsMax

kres rspline krangeMin, krangeMax, kcpsMin, kcpsMax

kscl scale kinput, kmax, kmin

ares transeg ia, idur, itype, ib [, idur2] [, itype] [, ic] ...

kres transeg ia, idur, itype, ib [, idur2] [, itype] [, ic] ...

Signal Generators:Envelope Generators.

ares adsr iatt, idec, islev, irel [, idel]

kres adsr iatt, idec, islev, irel [, idel]

ares envlpx xamp, irise, idur, idec, ifn, iatss, iatdec [, ixmod]

kres envlpx kamp, irise, idur, idec, ifn, iatss, iatdec [, ixmod]

ares envlpxr xamp, irise, idec, ifn, iatss, iatdec [, ixmod] [,irind]

kres envlpxr kamp, irise, idec, ifn, iatss, iatdec [, ixmod] [,irind]

ares linen xamp, irise, idur, idec

kres linen kamp, irise, idur, idec

ares linenr xamp, irise, idec, iatdec

kres linenr kamp, irise, idec, iatdec

ares madsr iatt, idec, islev, irel [, idel] [, ireltim]

kres madsr iatt, idec, islev, irel [, idel] [, ireltim]

ares mxadsr iatt, idec, islev, irel [, idel] [, ireltim]

kres mxadsr iatt, idec, islev, irel [, idel] [, ireltim]

ares xadsr iatt, idec, islev, irel [, idel]

kres xadsr iatt, idec, islev, irel [, idel]

Signal Generators:Models and Emulations.

ares bamboo kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] \
[, ifreq1] [, ifreq2]

ares barmodel kbcL, kbcR, iK, ib, kscan, iT30, ipos, ivel, iwid

ares cabasa iamp, idettack [, inum] [, idamp] [, imaxshake]

aI3, aV2, aV1 chuap kL, kR0, kC1, kG, kGa, kGb, kE, kC2, iI3, iV2, iV1, ktime_step

ares crunch iamp, idettack [, inum] [, idamp] [, imaxshake]

ares dripwater kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] \
[, ifreq1] [, ifreq2]

ares gogobel kamp, kfreq, ihrd, ipos, imp, kvibf, kvamp, ivfn

ares guiro kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] [, ifreq1]

ax, ay, az lorenz ksv, krv, kbv, kh, ix, iy, iz, iskip [, iskipinit]

kiter, koutrig mandel ktrig, kx, ky, kmaxIter

ares mandol kamp, kfreq, kpluck, kdetune, kgain, ksize, ifn [, iminfreq]

ares marimba kamp, kfreq, ihrd, ipos, imp, kvibf, kvamp, ivibfn, idec \
[, idoubles] [, itriples]

ares moog kamp, kfreq, kfiltq, kfiltrate, kvibf, kvamp, iafn, iwfn, ivfn

```
ax, ay, az planet kmass1, kmass2, ksep, ix, iy, iz, ivx, ivy, ivz, idelta \
    [, ifriction] [, iskip]

ares prepiano ifreq, iNS, iD, iK, \
    iT30, iB, kbcl, kbcr, imass, ifreq, iinit, ipos, ivel, isfreq, \
    isspread[, irattles, irubbers]

al, ar prepiano ifreq, iNS, iD, iK, \
    iT30, iB, kbcl, kbcr, imass, ifreq, iinit, ipos, ivel, isfreq, \
    isspread[, irattles, irubbers]

ares sandpaper iamp, idettack [, inum] [, idamp] [, imaxshake]

ares sekere iamp, idettack [, inum] [, idamp] [, imaxshake]

ares shaker kamp, kfreq, kbeans, kdamp, ktimes [, idecay]

ares sleighbells kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] \
    [, ifreq1] [, ifreq2]

ares stix iamp, idettack [, inum] [, idamp] [, imaxshake]

ares tambourine kamp, idettack [, inum] [, idamp] [, imaxshake] [, ifreq] \
    [, ifreq1] [, ifreq2]

ares vibes kamp, kfreq, ihrd, ipos, imp, kvibf, kvamp, ivibfn, idec

ares voice kamp, kfreq, kphoneme, kform, kvibf, kvamp, ifn, ivfn
```

Signal Generators:Phasors.

```
ares phasor xcps [, iphs]

kres phasor kcps [, iphs]

ares phasorbnk xcps, kndx, icnt [, iphs]

kres phasorbnk kcps, kndx, icnt [, iphs]

aphase, asyncout syncphasor xcps, asyncin, [, iphs]
```

Signal Generators:Random (Noise) Generators.

```
ares betarand krange, kalpha, kbeta
```

ires betarand krange, kalpha, kbeta

kres betarand krange, kalpha, kbeta

ares bexprnd krange

ires bexprnd krange

kres bexprnd krange

ares cauchy kalpha

ires cauchy kalpha

kres cauchy kalpha

aout cuserrnd kmin, kmax, ktableNum

iout cuserrnd imin, imax, itableNum

kout cuserrnd kmin, kmax, ktableNum

aout dusernd ktableNum

iout dusernd itableNum

kout dusernd ktableNum

ares exprand klambda

ires exprand klambda

kres exprand klambda

ares gauss krange

ires gauss krange

kres gauss krange

kout jitter kamp, kcpsMin, kcpsMax

kout jitter2 ktotamp, kamp1, kcps1, kamp2, kcps2, kamp3, kcps3

ares linrand krange

ires linrand krange

kres linrand krange

ares noise xamp, kbeta

ares pcauchy kalpha

ires pcauchy kalpha

kres pcauchy kalpha

ares pinkish xin [, imethod] [, inumbands] [, iseed] [, iskip]

ares poisson klambda

ires poisson klambda

kres poisson klambda

ares rand xamp [, iseed] [, isel] [, ioffset]

kres rand xamp [, iseed] [, isel] [, ioffset]

ares randh xamp, xcps [, iseed] [, isize] [, ioffset]

kres randh kamp, kcps [, iseed] [, isize] [, ioffset]

ares randi xamp, xcps [, iseed] [, isize] [, ioffset]

kres randi kamp, kcps [, iseed] [, isize] [, ioffset]

ares random kmin, kmax

ires random imin, imax

kres random kmin, kmax

ares randomh kmin, kmax, acps

kres randomh kmin, kmax, kcps

ares randomi kmin, kmax, acps

kres randomi kmin, kmax, kcps

ax rnd31 kscl, krpow [, iseed]

ix rnd31 iscl, irpow [, iseed]

kx rnd31 kscl, krpow [, iseed]

seed ival

kout trandom ktrig, kmin, kmax

ares trirand krange

ires trirand krange

kres trirand krange

ares unirand krange

ires unirand krange

kres unirand krange

aout = urd(ktableNum)

iout = urd(itableNum)

kout = urd(ktableNum)

ares weibull ksigma, ktau

ires weibull ksigma, ktau

kres weibull ksigma, ktau

Signal Generators:Sample Playback.

a1 bbcutm asource, ibps, isubdiv, ibarlength, iphrasebars, inumrepeats \
[, istutterspeed] [, istutterchance] [, ienvchoice]

a1,a2 bbcuts asource1, asource2, ibps, isubdiv, ibarlength, iphrasebars, \
inumrepeats [, istutterspeed] [, istutterchance] [, ienvchoice]

asig flooper kamp, kpitch, istart, idur, ifad, ifn

asig flooper2 kamp, kpitch, kloopstart, kloopend, kcrossfade, ifn \
[, istart, imode, ifenv, iskip]

aleft, aright fluidAllOut

fluidCCi iEngineNumber, iChannelNumber, iControllerNumber, iValue

fluidCCK iEngineNumber, iChannelNumber, iControllerNumber, kValue

fluidControl ienginenum, kstatus, kchannel, kdata1, kdata2

ienginenum fluidEngine [iReverbEnabled] [, iChorusEnabled] [, iNumChannels] [, iPolyphony]

isfnum fluidLoad soundfont, ienginenum[, ilistpresets]

fluidNote ienginenum, ichannelnum, imidikey, imidivel

aleft, aright fluidOut ienginenum

fluidProgramSelect ienginenum, ichannelnum, isfnum, ibanknum, ipresetnum

fluidSetInterpMethod ienginenum, ichannelnum, iInterpMethod

ar1 [,ar2] loscil xamp, kcps, ifn [, ibas] [, imod1] [, ibeg1] [, iend1] \
[, imod2] [, ibeg2] [, iend2]

ar1 [,ar2] loscil3 xamp, kcps, ifn [, ibas] [, imod1] [, ibeg1] [, iend1] \
[, imod2] [, ibeg2] [, iend2]

ar1 [, ar2, ar3, ar4, ar5, ar6, ar7, ar8, ar9, ar10, ar11, ar12, ar13, ar14, \
ar15, ar16] loscilx xamp, kcps, ifn \
[, iwsiz, ibas, istr, imod1, ibeg1, iend1]

ares lphasor xtrns [, ilps] [, ilpe] [, imode] [, istr] [, istor]

ares lposcil kamp, kfregratio, kloop, kend, ifn [, iphs]

ares lposcil3 kamp, kfregratio, kloop, kend, ifn [, iphs]

ar lposcila aamp, kfregratio, kloop, kend, ift [,iphs]

ar1, ar2 lposcilsa aamp, kfregratio, kloop, kend, ift [,iphs]

ar1, ar2 lposcilsa2 aamp, kfregratio, kloop, kend, ift [,iphs]

sfilist ifilhandle

ar1, ar2 sfinstr ivel, inotenum, xamp, xfreq, instrnum, ifilhandle \
[, iflag] [, ioffset]

ar1, ar2 sfinstr3 ivel, inotenum, xamp, xfreq, instrnum, ifilhandle \
[, iflag] [, ioffset]

```
ares sfinstr3m ivel, inotenum, xamp, xfreq, instrnum, ifilhandle \  
    [, iflag] [, ioffset]
```

```
ares sfinstrm ivel, inotenum, xamp, xfreq, instrnum, ifilhandle \  
    [, iflag] [, ioffset]
```

```
ir sfload "filename"
```

```
ar1, ar2 sflooper ivel, inotenum, kamp, kpitch, ipreindex, kloopstart, kloopend, kcrossfade, ifn \  
    [, istart, imode, ifenv, iskip]
```

```
sfpassign istartindex, ifilhandle[, imsgs]
```

```
ar1, ar2 sfplay ivel, inotenum, xamp, xfreq, ipreindex [, iflag] [, ioffset] [, ienv]
```

```
ar1, ar2 sfplay3 ivel, inotenum, xamp, xfreq, ipreindex [, iflag] [, ioffset] [, ienv]
```

```
ares sfplay3m ivel, inotenum, xamp, xfreq, ipreindex [, iflag] [, ioffset] [, ienv]
```

```
ares sfplaym ivel, inotenum, xamp, xfreq, ipreindex [, iflag] [, ioffset] [, ienv]
```

```
sfplist ifilhandle
```

```
ir sfpreset iprog, ibank, ifilhandle, ipreindex
```

```
asig, krec sndloop ain, kpitch, ktrig, idur, ifad
```

```
ares waveset ain, krep [, ilen]
```

Signal Generators:Scanned Synthesis.

```
scanhammer isrc, idst, ipos, imult
```

```
ares scans kamp, kfreq, ifn, id [, iorder]
```

```
aout scantable kamp, kpch, ipos, imass, istiff, idamp, ivel
```

```
scanu init, irate, ifnvel, ifnmass, ifnstif, ifncentr, ifndamp, kmass, \  
    kstif, kcentr, kdamp, ileft,  iright, kpos, kstrngth, ain, idisp, id
```

```
kpos, kvel xscanmap iscan, kamp, kvamp [, iwhich]
```


itabx, itaby

Signal Generators:Waveguide Physical Modeling.

ares pluck kamp, kcps, icps, ifn, imeth [, iparm1] [, iparm2]

ares repluck iplk, kamp, icps, kpick, krefl, axcite

ares streson asig, kfr, ifdbgain

ares wgbow kamp, kfreq, kpres, krat, kvibf, kvamp, ifn [, iminfreq]

ares wgbowedbar kamp, kfreq, kpos, kbowpres, kgain [, iconst] [, itvel] \
[, ibowpos] [, ilow]

ares wgbrass kamp, kfreq, ktens, iatt, kvibf, kvamp, ifn [, iminfreq]

ares wgclar kamp, kfreq, kstiff, iatt, idetk, kngain, kvibf, kvamp, ifn \
[, iminfreq]

ares wgflute kamp, kfreq, kjet, iatt, idetk, kngain, kvibf, kvamp, ifn \
[, iminfreq] [, ijetrf] [, iendrf]

ares wgpluck icps, iamp, kpick, iplk, idamp, ifilt, axcite

ares wgpluck2 iplk, kamp, icps, kpick, krefl

Signal I/O:File I/O.

dumpk ksig, ifilename, iformat, iprd

dumpk2 ksig1, ksig2, ifilename, iformat, iprd

dumpk3 ksig1, ksig2, ksig3, ifilename, iformat, iprd

dumpk4 ksig1, ksig2, ksig3, ksig4, ifilename, iformat, iprd

ficlose ihandle

ficlose Sfilename

`fin ifilename, iskipframes, iformat, ain1 [, ain2] [, ain3] [...]`

`fini ifilename, iskipframes, iformat, in1 [, in2] [, in3] [, ...]`

`fink ifilename, iskipframes, iformat, kin1 [, kin2] [, kin3] [...]`

`ihandle fiopen ifilename, imode`

`fout ifilename, iformat, aout1 [, aout2, aout3,...,aoutN]`

`fouti ihandle, iformat, iflag, iout1 [, iout2, iout3,...,ioutN]`

`foutir ihandle, iformat, iflag, iout1 [, iout2, iout3,...,ioutN]`

`foutk ifilename, iformat, kout1 [, kout2, kout3,...,koutN]`

`fprintks "filename", "string", [, kval1] [, kval2] [...]`

`fprints "filename", "string" [, ival1] [, ival2] [...]`

`kres readk ifilname, iformat, iprd`

`kr1, kr2 readk2 ifilname, iformat, iprd`

`kr1, kr2, kr3 readk3 ifilname, iformat, iprd`

`kr1, kr2, kr3, kr4 readk4 ifilname, iformat, iprd`

Signal I/O:Signal Input.

`ar1 [, ar2 [, ar3 [, ... ar24]]] disk in ifilcod, kpitch [, iskiptim] \
[, iwraparound] [, iformat] [, iskipinit]`

`a1[, a2[, ... a24]] disk in2 ifilcod, kpitch[, iskiptim \
[, iwrap[, iformat [, iwsizel[, ibufsize[, iskipinit]]]]]`

`ar1 in`

`ar1, ar2, ar3, ar4, ar5, ar6, ar7, ar8, ar9, ar10, ar11, ar12, ar13, ar14, \
ar15, ar16, ar17, ar18, ar19, ar20, ar21, ar22, ar23, ar24, ar25, ar26, \
ar27, ar28, ar29, ar30, ar31, ar32 in32`

arl inch ksigl

arl, ar2, ar3, ar4, ar5, ar6 inh

arl, ar2, ar3, ar4, ar5, ar6, ar7, ar8 ino

arl, ar2, ar3, a4 inq

inrg kstart, ain1 [,ain2, ain3, ..., ainN]

arl, ar2 ins

kvalue invalue "channel name"

Sname invalue "channel name"

arl, ar2, ar3, ar4, ar5, ar6, ar7, ar8, ar9, ar10, ar11, ar12, \
ar13, ar14, ar15, ar16 inx

inz ksigl

arl[, ar2[, ar3[, ... a24]]] soundin ifilcod [, iskptim] [, iformat] \
[, iskipinit] [, ibufsize]

Signal I/O:Signal Output.

mdelay kstatus, kchan, kd1, kd2, kdelay

aout1 [,aout2 ... aoutX] monitor

out asig

out32 asig1, asig2, asig3, asig4, asig5, asig6, asig7, asig8, asig10, \
asig11, asig12, asig13, asig14, asig15, asig16, asig17, asig18, \
asig19, asig20, asig21, asig22, asig23, asig24, asig25, asig26, \
asig27, asig28, asig29, asig30, asig31, asig32

outc asig1 [, asig2] [...]

outch ksig1, asig1 [, ksig2] [, asig2] [...]

outh asig1, asig2, asig3, asig4, asig5, asig6

outo asig1, asig2, asig3, asig4, asig5, asig6, asig7, asig8

outq asig1, asig2, asig3, asig4

outq1 asig

outq2 asig

outq3 asig

outq4 asig

outrg kstart, aout1 [,aout2, aout3, ..., aoutN]

outs asig1, asig2

outs1 asig

outs2 asig

outvalue "channel name", kvalue

outvalue "channel name", "string"

outx asig1, asig2, asig3, asig4, asig5, asig6, asig7, asig8, \
asig9, asig10, asig11, asig12, asig13, asig14, asig15, asig16

outz ksig1

soundout asig1, ifilcod [, iformat]

soundouts asig1, asigr, ifilcod [, iformat]

Signal I/O:Software Bus.

kval chani kchan

aval chani kchan

chano kval, kchan

chano aval, kchan

chn_k Sname, imode[, itype, idflt, imin, imax]

chn_a Sname, imode

chn_S Sname, imode

chnclear Sname

gival chnexport Sname, imode[, itype, idflt, imin, imax]

gkval chnexport Sname, imode[, itype, idflt, imin, imax]

gaval chnexport Sname, imode

gSval chnexport Sname, imode

ival chnget Sname

kval chnget Sname

aval chnget Sname

Sval chnget Sname

chnmix aval, Sname

itype, imode, ictltype, idflt, imin, imax chnparams

chnset ival, Sname

chnset kval, Sname

chnset aval, Sname

chnset Sval, Sname

setksmps iksmps

xinarg1 [, xinarg2] ... [xinargN] xin

xout xoutarg1 [, xoutarg2] ... [, xoutargN]

Signal I/O:Printing and Display.

dispfft xsig, iprd, iwsiz [, iwtyp] [, idbout] [, iwtflg]

display xsig, iprd [, inprds] [, iwtflg]

flashtxt iwhich, String

print iarg [, iarg1] [, iarg2] [...]

printf_i Sfmt, itrigger, [iarg1[, iarg2[, ...]]]

printf Sfmt, ktrig, [xarg1[, xarg2[, ...]]]

printk itime, kval [, ispace]

printk2 kvar [, inumspaces]

printks "string", itime [, kval1] [, kval2] [...]

prints "string" [, kval1] [, kval2] [...]

Signal I/O:Soundfile Queries.

ir filelen ifilcod, [iallowraw]

ir filenchnls ifilcod [, iallowraw]

ir filepeak ifilcod [, ichnl]

ir filesr ifilcod [, iallowraw]

Signal Modifiers:Amplitude Modifiers.

ares balance asig, acomp [, ihp] [, iskip]

ares clip asig, imeth, ilimit [, iarg]

ar compress aasig, acsig, kthresh, kloknee, khiknee, kratio, katt, krel, ilook

ares dam asig, kthreshold, icomp1, icomp2, irtime, iftime

ares gain asig, krms [, ihp] [, iskip]


```
ares multitap asig [, itime1] [, igain1] [, itime2] [, igain2] [...]
```

```
ares vdelay asig, adel, imaxdel [, iskip]
```

```
ares vdelay3 asig, adel, imaxdel [, iskip]
```

```
aout vdelayx ain, adl, imd, iws [, ist]
```

```
aout1, aout2, aout3, aout4 vdelayxq ain1, ain2, ain3, ain4, adl, imd, iws [, ist]
```

```
aout1, aout2 vdelayxs ain1, ain2, adl, imd, iws [, ist]
```

```
aout vdelayxw ain, adl, imd, iws [, ist]
```

```
aout1, aout2, aout3, aout4 vdelayxwq ain1, ain2, ain3, ain4, adl, \
    imd, iws [, ist]
```

```
aout1, aout2 vdelayxws ain1, ain2, adl, imd, iws [, ist]
```

Signal Modifiers:Panning and Spatialization.

```
ao1, ao2 bformdec isetup, aw, ax, ay, az [, ar, as, at, au, av \
    [, abk, al, am, an, ao, ap, aq]]
```

```
ao1, ao2, ao3, ao4 bformdec isetup, aw, ax, ay, az [, ar, as, at, \
    au, av [, abk, al, am, an, ao, ap, aq]]
```

```
ao1, ao2, ao3, ao4, ao5 bformdec isetup, aw, ax, ay, az [, ar, as, \
    at, au, av [, abk, al, am, an, ao, ap, aq]]
```

```
ao1, ao2, ao3, ao4, ao5, ao6, ao7, ao8 bformdec isetup, aw, ax, ay, az \
    [, ar, as, at, au, av [, abk, al, am, an, ao, ap, aq]]]
```

```
ao1, ao2 bformdec1 isetup, aw, ax, ay, az [, ar, as, at, au, av \
    [, abk, al, am, an, ao, ap, aq]]
```

```
ao1, ao2, ao3, ao4 bformdec1 isetup, aw, ax, ay, az [, ar, as, at, \
    au, av [, abk, al, am, an, ao, ap, aq]]
```

```
ao1, ao2, ao3, ao4, ao5 bformdec1 isetup, aw, ax, ay, az [, ar, as, \
    at, au, av [, abk, al, am, an, ao, ap, aq]]
```

```
ao1, ao2, ao3, ao4, ao5, ao6, ao7, ao8 bformdec1 isetup, aw, ax, ay, az \
    [, ar, as, at, au, av [, abk, al, am, an, ao, ap, aq]]]
```

```
aw, ax, ay, az bformenc asig, kalpha, kbeta, kord0, kord1
```

```
aw, ax, ay, az, ar, as, at, au, av bformenc asig, kalpha, kbeta, \
    kord0, kord1 , kord2
```


aw, ax, ay, az, ar, as, at, au, av, ak, al, am, an, ao, ap, aq bformenc \
asig, kalpha, kbeta, kord0, kord1, kord2, kord3

aw, ax, ay, az bformenc1 asig, kalpha, kbeta

aw, ax, ay, az, ar, as, at, au, av bformenc1 asig, kalpha, kbeta

aw, ax, ay, az, ar, as, at, au, av, ak, al, am, an, ao, ap, aq bformenc1 \
asig, kalpha, kbeta

aleft, aright hrtfer asig, kaz, kelev, "HRTFcompact"

aleft, aright hrtfmove asrc, kAz, kElev, ifile1, ifiler [, imode, ifade, isr]

aleft, aright hrtfmove2 asrc, kAz, kElev, ifile1, ifiler [,ioverlap, iradius, isr]

aleft, aright hrtfstat asrc, iAz, iElev, ifile1, ifiler [,iradius, isr]

a1, a2 locsend

a1, a2, a3, a4 locsend

a1, a2 locsig asig, kdegree, kdistance, kreverbsend

a1, a2, a3, a4 locsig asig, kdegree, kdistance, kreverbsend

a1, a2, a3, a4 pan asig, kx, ky, ifn [, imode] [, ioffset]

a1, a2 pan2 asig, xp [, imode]

a1, a2, a3, a4 space asig, ifn, ktime, kreverbsend, kx, ky

aW, aX, aY, aZ spat3d ain, kX, kY, kZ, idist, ift, imode, imdel, iovr [, istor]

aW, aX, aY, aZ spat3di ain, iX, iY, iZ, idist, ift, imode [, istor]

spat3dt ioutft, iX, iY, iZ, idist, ift, imode, irlen [, iftnocl]

k1 spdist ifn, ktime, kx, ky

a1, a2, a3, a4 spsend

ar1, ..., ar16 vbap16 asig, kazim [, kelev] [, kspread]

```
ar1, ..., ar16 vbap16move asig, idur, ispread, ifldnum, ifld1 \
    [, ifld2] [...]
```

```
ar1, ar2, ar3, ar4 vbap4 asig, kazim [, kelev] [, kspread]
```

```
ar1, ar2, ar3, ar4 vbap4move asig, idur, ispread, ifldnum, ifld1 \
    [, ifld2] [...]
```

```
ar1, ..., ar8 vbap8 asig, kazim [, kelev] [, kspread]
```

```
ar1, ..., ar8 vbap8move asig, idur, ispread, ifldnum, ifld1 \
    [, ifld2] [...]
```

```
vbaplsinit idim, ilsnum [, idir1] [, idir2] [...] [, idir32]
```

```
vbapz inumchnls, istartndx, asig, kazim [, kelev] [, kspread]
```

```
vbapzmove inumchnls, istartndx, asig, idur, ispread, ifldnum, ifld1, \
    ifld2, [...]
```

Signal Modifiers:Reverberation.

```
ares alpass asig, krvt, ilpt [, iskip] [, insmps]
```

```
a1, a2 babo asig, ksrcx, ksrcy, ksrcz, irx, iry, irz [, idiff] [, ifno]
```

```
ares comb asig, krvt, ilpt [, iskip] [, insmps]
```

```
aoutL, aoutR freeverb ainL, ainR, kRoomSize, kHFDamp[, iSRate[, iSkip]]
```

```
ares nestedap asig, imode, imaxdel, idel1, igain1 [, idel2] [, igain2] \
    [, idel3] [, igain3] [, istor]
```

```
ares nreverb asig, ktime, khdif [, iskip] [,inumCombs] [, ifnCombs] \
    [, inumAlpas] [, ifnAlpas]
```

```
ares reverb asig, krvt [, iskip]
```

```
ares reverb2 asig, ktime, khdif [, iskip] [,inumCombs] \
    [, ifnCombs] [, inumAlpas] [, ifnAlpas]
```

```
aoutL, aoutR reverbsc ainL, ainR, kfblvl, kfco[, israte[, ipitchm[, iskip]]]
```

ares valpass asig, krvt, xlpt, imaxlpt [, iskip] [, insmps]

ares vcomb asig, krvt, xlpt, imaxlpt [, iskip] [, insmps]

Signal Modifiers:Sample Level Operators.

denorm a1[, a2[, a3[, ...]]]

ares diff asig [, iskip]

kres diff ksig [, iskip]

kres downsamp asig [, iwlen]

ares fold asig, kincr

ares integ asig [, iskip]

kres integ ksig [, iskip]

ares interp ksig [, iskip] [, imode]

ares ntrpol asig1, asig2, kpoint [, imin] [, imax]

ires ntrpol isig1, isig2, ipoint [, imin] [, imax]

kres ntrpol ksig1, ksig2, kpoint [, imin] [, imax]

a(x) (control-rate args only)

i(x) (control-rate args only)

k(x) (i-rate args only)

ares samphold asig, agate [, ival] [, ivstor]

kres samphold ksig, kgate [, ival] [, ivstor]

ares upsamp ksig

kval vaget kndx, avar

vaset kval, kndx, avar

Signal Modifiers:Signal Limiters.

ares limit asig, klow, khigh

ires limit isig, ilow, ihigh

kres limit ksig, klow, khigh

ares mirror asig, klow, khigh

ires mirror isig, ilow, ihigh

kres mirror ksig, klow, khigh

ares wrap asig, klow, khigh

ires wrap isig, ilow, ihigh

kres wrap ksig, klow, khigh

Signal Modifiers:Special Effects.

ar distort asig, kdist, ifn[, ihp, istor]

ares distort1 asig, kpregain, kpostgain, kshape1, kshape2[, imode]

ares flanger asig, adel, kfeedback [, imaxd]

ares harmon asig, kestfrq, kmaxvar, kgenfreq1, kgenfreq2, imode, \
 iminfrq, iprd

ares harmon2 asig, koct, kfrq1, kfrq2, icpsmode, ilowest[, ipolarity]

ares harmon3 asig, koct, kfrq1, \
 kfrq2, kfrq3, icpsmode, ilowest[, ipolarity]

ares harmon4 asig, koct, kfrq1, \
 kfrq2, kfrq3, kfrq4, icpsmode, ilowest[, ipolarity]

ares phaser1 asig, kfreq, kord, kfeedback [, iskip]

ares phaser2 asig, kfreq, kq, kord, kmode, ksep, kfeedback

Signal Modifiers:Standard Filters.

ares atone asig, khp [, iskip]

ares atonex asig, khp [, inumlayer] [, iskip]

ares biquad asig, kb0, kb1, kb2, ka0, ka1, ka2 [, iskip]

ares biquada asig, ab0, ab1, ab2, aa0, aa1, aa2 [, iskip]

ares butbp asig, kfreq, kband [, iskip]

ares butbr asig, kfreq, kband [, iskip]

ares buthp asig, kfreq [, iskip]

ares butlp asig, kfreq [, iskip]

ares butterbp asig, kfreq, kband [, iskip]

ares butterbr asig, kfreq, kband [, iskip]

ares butterhp asig, kfreq [, iskip]

ares butterlp asig, kfreq [, iskip]

ares clfilt asig, kfreq, itype, inpol [, ikind] [, ipbr] [, isba] [, iskip]

aout mode ain, kfreq, kQ [, iskip]

ares tone asig, khp [, iskip]

ares tonex asig, khp [, inumlayer] [, iskip]

Signal Modifiers:Standard Filters:Resonant.

ares areson asig, kcf, kbw [, iscl] [, iskip]

ares bqrez asig, xfco, xres [, imodel] [, iskip]

ares lowpass2 asig, kcf, kq [, iskip]

ares lowres asig, kcutoff, kresonance [, iskip]

ares lowresx asig, kcutoff, kresonance [, inumlayer] [, iskip]

ares lpfl8 asig, kfco, kres, kdist

asig moogladder ain, kcf, kres[, istor]

ares moogvcf asig, xfco, xres [,iscale, iskip]

ares moogvcf2 asig, xfco, xres [,iscale, iskip]

ares reson asig, kcf, kbw [, iscl] [, iskip]

ares resonr asig, kcf, kbw [, iscl] [, iskip]

ares resonx asig, kcf, kbw [, inumlayer] [, iscl] [, iskip]

ares resony asig, kbf, kbw, inum, ksep [, isepmode] [, iscl] [, iskip]

ares resonz asig, kcf, kbw [, iscl] [, iskip]

ares rezzy asig, xfco, xres [, imode, iskip]

ahp,alp,abp,abr statevar ain, kcf, kq [, iosamps, istor]

alow, ahigh, aband svfilter asig, kcf, kq [, iscl]

ares tbvcf asig, xfco, xres, kdist, kasym [, iskip]

ares vlowres asig, kfco, kres, iord, ksep

Signal Modifiers:Standard Filters:Control.

kres aresonk ksig, kcf, kbw [, iscl] [, iskip]

kres atonek ksig, khp [, iskip]

kres lineto ksig, ktime

kres port ksig, ihtim [, isig]

kres portk ksig, khtim [, isig]

kres resonk ksig, kcf, kbw [, iscl] [, iskip]

kres resonxk ksig, kcf, kbw[, inumlayer, iscl, istor]

kres tlineto ksig, ktime, ktrig

kres tonek ksig, khp [, iskip]

Signal Modifiers:Specialized Filters.

ares dcblock ain [, igain]

ares dcblock2 ain [, iorder] [, iskip]

asig eqfil ain, kcf, kbw, kgain[, istor]

ares filter2 asig, iM, iN, ib0, ib1, ..., ibM, ial, ia2, ..., iaN

kres filter2 ksig, iM, iN, ib0, ib1, ..., ibM, ial, ia2, ..., iaN

asig fofilter ain, kcf, kris, kdec[, istor]

ar1, ar2 hilbert asig

ares nlfilt ain, ka, kb, kd, kC, kL

ares pareq asig, kc, kv, kq [, imode] [, iskip]

ar rbjeq asig, kfco, klvl, kQ, kS[, imode]

ares zfilter2 asig, kdamp, kfreq, iM, iN, ib0, ib1, ..., ibM, \
ial,ia2, ..., iaN

Signal Modifiers:Waveguides.

ares wguidel asig, xfreq, kcutoff, kfeedback

```
ares wguide2 asig, xfreq1, xfreq2, kcutoff1, kcutoff2, \
    kfeedback1, kfeedback2
```

Signal Modifiers:Waveshaping.

```
aout chebyshevpoly ain, k0 [, k1 [, k2 [...]]]

aout pdclip ain, kWidth, kCenter [, ibipolar [, ifullscale]]

aout pdhalf ain, kShapeAmount [, ibipolar [, ifullscale]]

aout pdhalfy ain, kShapeAmount [, ibipolar [, ifullscale]]

aout powershape ain, kShapeAmount [, ifullscale]
```

Signal Modifiers:Comparators and Accumulators.

```
amax max ain1 [, ain2] [, ain3] [, ain4] [...]

kmax max kin1 [, kin2] [, kin3] [, kin4] [...]

knumkout max_k asig, ktrig, itype

amax maxabs ain1 [, ain2] [, ain3] [, ain4] [...]

kmax maxabs kin1 [, kin2] [, kin3] [, kin4] [...]

maxabsaccum aAccumulator, aInput

maxaccum aAccumulator, aInput

amin min ain1 [, ain2] [, ain3] [, ain4] [...]

kmin min kin1 [, kin2] [, kin3] [, kin4] [...]

amin minabs ain1 [, ain2] [, ain3] [, ain4] [...]

kmin minabs kin1 [, kin2] [, kin3] [, kin4] [...]

minabsaccum aAccumulator, aInput

minaccum aAccumulator, aInput
```


Instrument Control:Clock Control.

clockoff inum

clockon inum

Instrument Control:Conditional Values.

(a == b ? v1 : v2)

(a >= b ? v1 : v2)

(a > b ? v1 : v2)

(a <= b ? v1 : v2)

(a < b ? v1 : v2)

(a != b ? v1 : v2)

Instrument Control:Duration Control.

ihold

turnoff

turnoff2 kinsno, kmode, krelease

turnon insnum [, itime]

Instrument Control:Invocation.

event "scorechar", kinsnum, kdelay, kdur, [, kp4] [, kp5] [, ...]

event "scorechar", "insname", kdelay, kdur, [, kp4] [, kp5] [, ...]

event_i "scorechar", iinsnum, idelay, idur, [, ip4] [, ip5] [, ...]

```
event_i "scorechar", "insname", idelay, idur, [, ip4] [, ip5] [, ...]
```

```
mute insnum [, iswitch]
```

```
mute "insname" [, iswitch]
```

```
remove insnum
```

```
schedkwhen ktrigger, kmintim, kmaxnum, kinsnum, kwhen, kdur \  
[, ip4] [, ip5] [...]
```

```
schedkwhen ktrigger, kmintim, kmaxnum, "insname", kwhen, kdur \  
[, ip4] [, ip5] [...]
```

```
schedkwhennamed ktrigger, kmintim, kmaxnum, "name", kwhen, kdur \  
[, ip4] [, ip5] [...]
```

```
schedule insnum, iwhen, idur [, ip4] [, ip5] [...]
```

```
schedule "insname", iwhen, idur [, ip4] [, ip5] [...]
```

```
schedwhen ktrigger, kinsnum, kwhen, kdur [, ip4] [, ip5] [...]
```

```
schedwhen ktrigger, "insname", kwhen, kdur [, ip4] [, ip5] [...]
```

```
scoreline Sin, ktring
```

```
scoreline_i Sin
```

Instrument Control:Program Flow Control.

```
cggoto condition, label
```

```
cigoto condition, label
```

```
ckgoto condition, label
```

```
cngoto condition, label
```

```
else
```

```
elseif xa R xb then
```

```
endif
```

goto label

if ia R ib igoto label

if ka R kb kgoto label

if xa R xb goto label

if xa R xb then

igoto label

kgoto label

loop_ge indx, idecr, imin, label

loop_ge kndx, kdecr, kmin, label

loop_gt indx, idecr, imin, label

loop_gt kndx, kdecr, kmin, label

loop_le indx, incr, imax, label

loop_le kndx, kncr, kmax, label

loop_lt indx, incr, imax, label

loop_lt kndx, kncr, kmax, label

tigoto label

timeout istrtr, idur, label

Instrument Control:Realtime Performance Control.

ir active insnum

kres active kinsnum

cpuprc insnum, ipercent

exitnow

jacktransport icommand [, ilocation]

maxalloc insnum, icount

prealloc insnum, icount

prealloc "insname", icount

Instrument Control:Initialization and Reinitialization.

ares = xarg

ires = iarg

kres = karg

ares init iarg

ires init iarg

kres init iarg

insno nstrnum "name"

p(x)

pset icon1 [, icon2] [...]

reinit label

rigoto label

rireturn

ir tival

Instrument Control:Sensing and Control.

kres button knum

ktrig changed kvar1 [, kvar2,..., kvarN]

kres checkbox knum

kres control knum

ares follow asig, idt

ares follow2 asig, katt, krel

Svalue getcfg iopt

ktrig metro kfreq [, initphase]

ksig miditempo

icount pcount

kres peak asig

kres peak ksig

ivalue pindex ipfieldIndex

koct, kamp pitch asig, iupdt, ilo, ihi, idbthresh [, ifrqs] [, iconf] \
[, istrtr] [, iocts] [, iq] [, inptls] [, irolloff] [, iskip]

kcps, krms pitchamdf asig, imincps, imaxcps [, icps] [, imedi] \
[, idowns] [, iexcps] [, irmsmedi]

kcps, kamp ptrack asig, ihopsize[,ipeaks]

rewindscore

kres rms asig [, ihp] [, iskip]

kres[, kkeydown] sensekey

ktrig_out seqtime ktime_unit, kstart, kloop, kinitndx, kfn_times

ktrig_out seqtime2 ktrig_in, ktime_unit, kstart, kloop, kinitndx, kfn_times

setctrl inum, ival, itype

setscorepos ipos

splitrig ktrig, kndx, imaxtics, ifn, kout1 [,kout2,...,koutN]

ktemp tempest kin, iprd, imindur, imemdur, ihp, ithresh, ihtim, ixfdbak, \
istartempo, ifn [, idisprd] [, itweek]

tempo ktempo, istartempo

kres tempoval

ktrig timedseq ktimpnt, ifn, kp1 [,kp2, kp3, ...,kpN]

kout trigger ksig, kthreshold, kmode

trigseq ktrig_in, kstart, kloop, kinitndx, kfn_values, kout1 [, kout2] [...]

kx, ky xyin iprd, ixmin, ixmax, iymmin, iymax [, ixinit] [, iyinit]

Instrument Control:Stacks.

xval1, [xval2, ... , xval31] pop

ival1, [ival2, ... , ival31] pop

fsig pop_f

push xval1, [xval2, ... , xval31]

push ival1, [ival2, ... , ival31]

push_f fsig

stack iStackSize

Instrument Control:Subinstrument Control.

a1, [...] [, a8] subinstr instrnum [, p4] [, p5] [...]

a1, [...] [, a8] subinstr "insname" [, p4] [, p5] [...]

subinstrinit instrnum [, p4] [, p5] [...]

subinstrinit "insname" [, p4] [, p5] [...]

Instrument Control:Time Reading.

ir date

Sir dates [itime]

ir readclock inum

ires rtclock

kres rtclock

kres timeinstk

kres timeinsts

kres timeinsts

ires timek

kres timek

ires times

kres times

Table Control.

ftfree ifno, iwhen

gir ftgen ifn, itime, isize, igen, iarga [, iargb] [...]

ifno ftgentmp ip1, ip2dummy, isize, igen, iarga, iargb, ...

sndload Sfname[, ifmt[, ichns[, isr[, ibas[, iamp[, istrtr \

[, ilpmod[, ilps[, ilpe]]]]]]]]]]

Table Control:Table Queries.

ftchnls(x) (init-rate args only)

ftlen(x) (init-rate args only)

ftlptim(x) (init-rate args only)

ftsr(x) (init-rate args only)

nsamp(x) (init-rate args only)

ires tablen g ifn

kres tablen g kfn

tb0_init ifn

tb1_init ifn

tb2_init ifn

tb3_init ifn

tb4_init ifn

tb5_init ifn

tb6_init ifn

tb7_init ifn

tb8_init ifn

tb9_init ifn

tb10_init ifn

tb11_init ifn

tb12_init ifn

tb13_init ifn

tb14_init ifn

tb15_init ifn

iout = tb0(iIndex)

kout = tb0(kIndex)

iout = tbl(iIndex)
kout = tbl(kIndex)
iout = tb2(iIndex)
kout = tb2(kIndex)
iout = tb3(iIndex)
kout = tb3(kIndex)
iout = tb4(iIndex)
kout = tb4(kIndex)
iout = tb5(iIndex)
kout = tb5(kIndex)
iout = tb6(iIndex)
kout = tb6(kIndex)
iout = tb7(iIndex)
kout = tb7(kIndex)
iout = tb8(iIndex)
kout = tb8(kIndex)
iout = tb9(iIndex)
kout = tb9(kIndex)
iout = tbl0(iIndex)
kout = tbl0(kIndex)
iout = tbl1(iIndex)
kout = tbl1(kIndex)
iout = tbl2(iIndex)
kout = tbl2(kIndex)
iout = tbl3(iIndex)
kout = tbl3(kIndex)
iout = tbl4(iIndex)
kout = tbl4(kIndex)
iout = tbl5(iIndex)

kout = tbl5(kIndex)

Table Control:Dynamic Selection.

ares tableikt xndx, kfn [, ixmode] [, ixoff] [, iwrap]

kres tableikt kndx, kfn [, ixmode] [, ixoff] [, iwrap]

ares tablekt xndx, kfn [, ixmode] [, ixoff] [, iwrap]

kres tablekt kndx, kfn [, ixmode] [, ixoff] [, iwrap]

ares tablexkt xndx, kfn, kwrap, iwsiz [, ixmode] [, ixoff] [, iwrap]

Table Control:Read/Write Opreations.

ftload "filename", iflag, ifn1 [, ifn2] [...]

ftloadk "filename", ktrig, iflag, ifn1 [, ifn2] [...]

ftsave "filename", iflag, ifn1 [, ifn2] [...]

ftsavek "filename", ktrig, iflag, ifn1 [, ifn2] [...]

tablecopy kdft, ksft

tablegpw kfn

tableicopy idft, isft

tableigpw ifn

tableimix idft, idoff, ilen, is1ft, isloff, islg, is2ft, is2off, is2g

tableiw isig, indx, ifn [, ixmode] [, ixoff] [, iwgmde]

tablemix kdft, kdoff, klen, ks1ft, ksloff, kslg, ks2ft, ks2off, ks2g

ares tablera kfn, kstart, koff

```
tablew asig, andx, ifn [, ixmode] [, ixoff] [, iwgmode]

tablew isig, indx, ifn [, ixmode] [, ixoff] [, iwgmode]

tablew ksig, kndx, ifn [, ixmode] [, ixoff] [, iwgmode]

kstart tablewa kfn, asig, koff

tablewkt asig, andx, kfn [, ixmode] [, ixoff] [, iwgmode]

tablewkt ksig, kndx, kfn [, ixmode] [, ixoff] [, iwgmode]

kout tabmorph kindex, kweightpoint, ktabnum1, ktabnum2, \
    ifn1, ifn2 [, ifn3, ifn4, ... ifnN]

aout tabmorpha aindex, aweightpoint, atabnum1, atabnum2, \
    ifn1, ifn2 [, ifn3, ifn4, ... ifnN]

aout tabmorphak aindex, kweightpoint, ktabnum1, ktabnum2, \
    ifn1, ifn2 [, ifn3, ifn4, ... ifnN]

kout tabmorphi kindex, kweightpoint, ktabnum1, ktabnum2, \
    ifn1, ifn2 [, ifn3, ifn4, ... ifnN]

tabplay ktrig, knumtics, kfn, kout1 [,kout2,..., koutN]

tabrec ktrig_start, ktrig_stop, knumtics, kfn, kin1 [,kin2,...,kinN]
```

FLTK:Containers.

```
FLgroup "label", iwidth, iheight, ix, iy [, iborder] [, image]

FLgroupEnd

FLpack iwidth, iheight, ix, iy, itype, ispace, iborder

FLpackEnd

FLpanel "label", iwidth, iheight [, ix] [, iy] [, iborder] [, ikbdcapture] [, iclose]

FLpanelEnd

FLscroll iwidth, iheight [, ix] [, iy]
```

FLscrollEnd

FLtabs iwidth, iheight, ix, iy

FLtabsEnd

FLTK:Valuators.

```
kout, ihandle FLcount "label", imin, imax, istep1, istep2, itype, \
      iwidth, iheight, ix, iy, iopcode [, kp1] [, kp2] [, kp3] [...] [, kpN]
```

```
koutx, kouty, ihandlex, ihandley FLjoy "label", iminx, imaxx, iminy, \
      imaxy, iexp, iexpy, idisp, idispy, iwidth, iheight, ix, iy
```

```
kout, ihandle FLknob "label", imin, imax, iexp, itype, idisp, iwidth, \
      ix, iy [, icursorsize]
```

```
kout, ihandle FLroller "label", imin, imax, istep, iexp, itype, idisp, \
      iwidth, iheight, ix, iy
```

```
kout, ihandle FLslider "label", imin, imax, iexp, itype, idisp, iwidth, \
      iheight, ix, iy
```

```
kout, ihandle FLtext "label", imin, imax, istep, itype, iwidth, \
      iheight, ix, iy
```

FLTK:Other.

```
ihandle FLbox "label", itype, ifont, isize, iwidth, iheight, ix, iy [, image]
```

```
kout, ihandle FLbutBank itype, inumx, inumy, iwidth, iheight, ix, iy, \
      iopcode [, kp1] [, kp2] [, kp3] [, kp4] [, kp5] [....] [, kpN]
```

```
kout, ihandle FLbutton "label", ion, ioff, itype, iwidth, iheight, ix, \
      iy, iopcode [, kp1] [, kp2] [, kp3] [, kp4] [, kp5] [....] [, kpN]
```

```
ihandle FLcloseButton "label", iwidth, iheight, ix, iy
```

```
ihandle FLEXecButton "command", iwidth, iheight, ix, iy
```

```
inumsnap FLgetsnap index [, igroup]
```

```
ihandle FLhvsBox inumlinesX, inumlinesY, iwidth, iheight, ix, iy [, image]
```

FLhvsBox kx, ky, ihandle

kascii FLkeyIn [ifn]

FLloadsnap "filename" [, igroup]

kx, ky, kb1, kb2, kb3 FLmouse [, imode]

FLprintk itime, kval, idisp

FLprintk2 kval, idisp

FLrun

FLsavesnap "filename" [, igroup]

inumsnap, inumval FLsetsnap index [, ifn, igroup]

FLsetSnapGroup igroup

FLsetVal ktrig, kvalue, ihandle

FLsetVal_i ivalue, ihandle

FLslidBnk "names", inumsliders [, ioutable] [, iwidth] [, iheight] [, ix] \
[, iy] [, itypetable] [, iexptable] [, istart_index] [, iminmaxtable]

FLslidBnk2 "names", inumsliders, ioutable, iconfigtable [,iwidth, iheight, ix, iy, istart_index]

FLslidBnk2Set ihandle, ifn [, istartIndex, istartSlid, inumSlid]

FLslidBnk2Setk ktrig, ihandle, ifn [, istartIndex, istartSlid, inumSlid]

ihandle FLslidBnkGetHandle

FLslidBnkSet ihandle, ifn [, istartIndex, istartSlid, inumSlid]

FLslidBnkSetk ktrig, ihandle, ifn [, istartIndex, istartSlid, inumSlid]

FLupdate

`ihandle FLvalue "label", iwidth, iheight, ix, iy`

`FLvkeybd "keyboard.map", iwidth, iheight, ix, iy`

`FLvslidBnk "names", inumsliders [, ioutable] [, iwidth] [, iheight] [, ix] \
[, iy] [, itypetable] [, iexptable] [, istart_index] [, iminmaxtable]`

`FLvslidBnk2 "names", inumsliders, ioutable, iconfigtable [,iwidth, iheight, ix, iy, istart_index]`

`koutx, kouty, kinside FLxyin ioutx_min, ioutx_max, iouty_min, iouty_max, \
iwindx_min, iwindx_max, iwindy_min, iwindy_max [, iexpx, iexpy, ioutx, iouty]`

`vphaseseg kphase, ioutab, ielems, itab1,idist1,itab2 \
[,idist2,itab3, ... ,idistN-1,itabN]`

FLTK:Appearance.

`FLcolor ired, igreen, iblue [, ired2, igreen2, iblue2]`

`FLcolor2 ired, igreen, iblue`

`FLhide ihandle`

`FLlabel isize, ifont, ialign, ired, igreen, iblue`

`FLsetAlign ialign, ihandle`

`FLsetBox itype, ihandle`

`FLsetColor ired, igreen, iblue, ihandle`

`FLsetColor2 ired, igreen, iblue, ihandle`

`FLsetFont ifont, ihandle`

`FLsetPosition ix, iy, ihandle`

`FLsetSize iwidth, iheight, ihandle`

`FLsetText "itext", ihandle`

FLsetTextColor ired, iblue, igreen, ihandle

FLsetTextSize isize, ihandle

FLsetTextType itype, ihandle

FLshow ihandle

Mathematical Operations:Arithmetic and Logic Operations.

$a + b$ (no rate restriction)

a / b (no rate restriction)

$a \% b$ (no rate restriction)

$a * b$ (no rate restriction)

$a \&\& b$ (logical AND; not audio-rate)

$a \& b$ (bitwise AND)

$\sim a$ (bitwise NOT)

$a | b$ (bitwise OR)

$a \ll b$ (bitshift left)

$a \gg b$ (bitshift left)

$a \# b$ (bitwise NON EQUIVALENCE)

$a || b$ (logical OR; not audio-rate)

$a \wedge b$ (b not audio-rate)

$a \# b$ (no rate restriction)

Mathematical Operations:Comparators and Accumulators.

`clear avar1 [, avar2] [, avar3] [...]`

`vincr accum, aincr`

Mathematical Operations:Mathematical Functions.

`abs(x)` (no rate restriction)

`ceil(x)` (init-, control-, or audio-rate arg allowed)

`exp(x)` (no rate restriction)

`floor(x)` (init-, control-, or audio-rate arg allowed)

`frac(x)` (init-rate or control-rate args; also works at audio rate in Csound5)

`int(x)` (init-rate or control-rate; also works at audio rate in Csound5)

`log(x)` (no rate restriction)

`log10(x)` (no rate restriction)

`logbtwo(x)` (init-rate or control-rate args only)

`powoftwo(x)` (init-rate or control-rate args only)

`round(x)` (init-, control-, or audio-rate arg allowed)

`sqrt(x)` (no rate restriction)

Mathematical Operations:Trigonometric Functions.

`cos(x)` (no rate restriction)

`cosh(x)` (no rate restriction)

`cosinv(x)` (no rate restriction)

`sin(x)` (no rate restriction)

`sinh(x)` (no rate restriction)

`sininv(x)` (no rate restriction)

`tan(x)` (no rate restriction)

`tanh(x)` (no rate restriction)

`taninv(x)` (no rate restriction)

Mathematical Operations:Amplitude Functions.

`ampdb(x)` (no rate restriction)

`ampdbfs(x)` (no rate restriction)

`db(x)`

`dbamp(x)` (init-rate or control-rate args only)

`dbfsamp(x)` (init-rate or control-rate args only)

Mathematical Operations:Random Functions.

`birnd(x)` (init- or control-rate only)

`rnd(x)` (init- or control-rate only)

Mathematical Operations:Opcode Equivalents of Functions.

`ares divz xa, xb, ksubst`

`ires divz ia, ib, isubst`

`kres divz ka, kb, ksubst`

`ares mac asig1, ksig1 [, asig2] [, ksig2] [, asig3] [, ksig3] [...]`

ares maca asig1 , asig2 [, asig3] [, asig4] [, asig5] [...]

aout polynomial ain, k0 [, k1 [, k2 [...]]]

ares pow aarg, kpow [, inorm]

ires pow iarg, ipow [, inorm]

kres pow karg, kpow [, inorm]

ares product asig1, asig2 [, asig3] [...]

ares sum asig1 [, asig2] [, asig3] [...]

ares taninv2 ay, ax

ires taninv2 iy, ix

kres taninv2 ky, kx

Pitch Converters:Functions.

cent(x)

cpsmidinn (MidiNoteNumber) (init- or control-rate args only)

cpsoct (oct) (no rate restriction)

cpspch (pch) (init- or control-rate args only)

octave(x)

octcps (cps) (init- or control-rate args only)

octmidinn (MidiNoteNumber) (init- or control-rate args only)

octpch (pch) (init- or control-rate args only)

pchmidinn (MidiNoteNumber) (init- or control-rate args only)

pchoct (oct) (init- or control-rate args only)

semitone(x)

Pitch Converters:Tuning Opcodes.

icps cps2pch ipch, iequal

kcps cpstun ktrig, kindex, kfn

icps cpstuni index, ifn

icps cpsxpch ipch, iequal, irepeat, ibase

Real-time MIDI:Input.

kaft aftouch [imin] [, imax]

ival chanctrl ichnl, ictlno [, ilow] [, ihigh]

kval chanctrl ichnl, ictlno [, ilow] [, ihigh]

idest ctrl14 ichan, ictlno1, ictlno2, imin, imax [, ifn]

kdest ctrl14 ichan, ictlno1, ictlno2, kmin, kmax [, ifn]

idest ctrl21 ichan, ictlno1, ictlno2, ictlno3, imin, imax [, ifn]

kdest ctrl21 ichan, ictlno1, ictlno2, ictlno3, kmin, kmax [, ifn]

idest ctrl7 ichan, ictlno, imin, imax [, ifn]

kdest ctrl7 ichan, ictlno, kmin, kmax [, ifn]

adest ctrl7 ichan, ictlno, kmin, kmax [, ifn] [, icutoff]

ctrlinit ichnl, ictlno1, ival1 [, ictlno2] [, ival2] [, ictlno3] \
[, ival3] [...ival32]

initc14 ichan, ictlno1, ictlno2, ivalue

initc21 ichan, ictlno1, ictlno2, ictlno3, ivalue

initc7 ichan, ictlno, ivalue

```
massign ichnl, insnum[, ireset]

massign ichnl, "insname"[, ireset]


idest midic14 ictlno1, ictlno2, imin, imax [, ifn]

kdest midic14 ictlno1, ictlno2, kmin, kmax [, ifn]


idest midic21 ictlno1, ictlno2, ictlno3, imin, imax [, ifn]

kdest midic21 ictlno1, ictlno2, ictlno3, kmin, kmax [, ifn]


idest midic7 ictlno, imin, imax [, ifn]

kdest midic7 ictlno, kmin, kmax [, ifn]


ival midictrl inum [, imin] [, imax]

kval midictrl inum [, imin] [, imax]


ival notnum


ibend pchbend [imin] [, imax]

kbend pchbend [imin] [, imax]


pgmassign ipgm, inst[, ichnl]

pgmassign ipgm, "insname"[, ichnl]


ires polyaft inote [, ilow] [, ihigh]

kres polyaft inote [, ilow] [, ihigh]


ival veloc [ilow] [, ihigh]
```

Real-time MIDI:Output.

```
nrpn kchan, kparmnum, kparmvalue

outiat ichn, ivalue, imin, imax

outic ichn, inum, ivalue, imin, imax
```

outic14 ichn, imsb, ilsb, ivalue, imin, imax

outipat ichn, inotenum, ivalue, imin, imax

outipb ichn, ivalue, imin, imax

outipc ichn, iprog, imin, imax

outkat kchn, kvalue, kmin, kmax

outkc kchn, knum, kvalue, kmin, kmax

outkc14 kchn, kmsb, klsb, kvalue, kmin, kmax

outkpat kchn, knotenum, kvalue, kmin, kmax

outkpb kchn, kvalue, kmin, kmax

outkpc kchn, kprog, kmin, kmax

Real-time MIDI:Converters.

iamp ampmidi iscal [, ifn]

icps cpsmidi

icps cpsmidib [irange]

kcps cpsmidib [irange]

icps cpstmid ifn

ioct octmidi

ioct octmidib [irange]

koct octmidib [irange]

ipch pchmidi

ipch pchmidib [irange]

kpch pchmidib [irange]

Real-time MIDI:Generic I/O.

kstatus, kchan, kdata1, kdata2 midiin

midout kstatus, kchan, kdata1, kdata2

Real-time MIDI:Event Extenders.

kflag release

xtratim iextradur

Real-time MIDI>Note Output.

midion kchn, knum, kvel

midion2 kchn, knum, kvel, ktrig

moscil kchn, knum, kvel, kdur, kpause

noteoff ichn, inum, ivel

noteon ichn, inum, ivel

noteondur ichn, inum, ivel, idur

noteondur2 ichn, inum, ivel, idur

Real-time MIDI:MIDI/Score Interoperability.

midichannelaftertouch xchannelaftertouch [, ilow] [, ihigh]

ichn midichn

midicontrolchange xcontroller, xcontrollervalue [, ilow] [, ihigh]

mididefault xdefault, xvalue

midinoteoff xkey, xvelocity

midinoteoncps xcps, xvelocity

midinoteonkey xkey, xvelocity

midinoteonoct xoct, xvelocity

midinoteonpch xpch, xvelocity

midipitchbend xpitchbend [, ilow] [, ihigh]

midipolyaftertouch xpolyaftertouch, xcontrolvalue [, ilow] [, ihigh]

midiprogramchange xprogram

Real-time MIDI: System Realtime.

mclock ifreq

mrtmsg imsgtype

Real-time MIDI: Slider Banks.

i1,...,i16 s16b14 ichan, ictlno_msb1, ictlno_lsb1, imin1, imax1, \
initvalue1, ifn1,..., ictlno_msb16, ictlno_lsb16, imin16, imax16, initvalue16, ifn16

k1,...,k16 s16b14 ichan, ictlno_msb1, ictlno_lsb1, imin1, imax1, \
initvalue1, ifn1,..., ictlno_msb16, ictlno_lsb16, imin16, imax16, initvalue16, ifn16

i1,...,i32 s32b14 ichan, ictlno_msb1, ictlno_lsb1, imin1, imax1, \
initvalue1, ifn1,..., ictlno_msb32, ictlno_lsb32, imin32, imax32, initvalue32, ifn32

k1,...,k32 s32b14 ichan, ictlno_msb1, ictlno_lsb1, imin1, imax1, \
initvalue1, ifn1,..., ictlno_msb32, ictlno_lsb32, imin32, imax32, initvalue32, ifn32

i1,...,i16 slider16 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \
ictlnum16, imin16, imax16, init16, ifn16

k1,...,k16 slider16 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \
ictlnum16, imin16, imax16, init16, ifn16

```

k1,...,k16 slider16f ichan, ictlnum1, imin1, imax1, init1, ifn1, \
    icutoff1,..., ictlnum16, imin16, imax16, init16, ifn16, icutoff16

kflag slider16table ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \
    init1, ifn1, .... , ictlnum16, imin16, imax16, init16, ifn16

kflag slider16tablef ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \
    init1, ifn1, icutoff1, .... , ictlnum16, imin16, imax16, init16, ifn16, icutoff16

i1,...,i32 slider32 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \
    ictlnum32, imin32, imax32, init32, ifn32

k1,...,k32 slider32 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \
    ictlnum32, imin32, imax32, init32, ifn32

k1,...,k32 slider32f ichan, ictlnum1, imin1, imax1, init1, ifn1, icutoff1, \
    ..., ictlnum32, imin32, imax32, init32, ifn32, icutoff32

kflag slider32table ichan, ioutTable, ioffset, ictlnum1, imin1, \
    imax1, init1, ifn1, .... , ictlnum32, imin32, imax32, init32, ifn32

kflag slider32tablef ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \
    init1, ifn1, icutoff1, .... , ictlnum32, imin32, imax32, init32, ifn32, icutoff32

i1,...,i64 slider64 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \
    ictlnum64, imin64, imax64, init64, ifn64

k1,...,k64 slider64 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \
    ictlnum64, imin64, imax64, init64, ifn64

k1,...,k64 slider64f ichan, ictlnum1, imin1, imax1, init1, ifn1, \
    icutoff1,..., ictlnum64, imin64, imax64, init64, ifn64, icutoff64

kflag slider64table ichan, ioutTable, ioffset, ictlnum1, imin1, \
    imax1, init1, ifn1, .... , ictlnum64, imin64, imax64, init64, ifn64

kflag slider64tablef ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \
    init1, ifn1, icutoff1, .... , ictlnum64, imin64, imax64, init64, ifn64, icutoff64

i1,...,i8 slider8 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \
    ictlnum8, imin8, imax8, init8, ifn8

k1,...,k8 slider8 ichan, ictlnum1, imin1, imax1, init1, ifn1,..., \
    ictlnum8, imin8, imax8, init8, ifn8

k1,...,k8 slider8f ichan, ictlnum1, imin1, imax1, init1, ifn1, icutoff1, \
    ..., ictlnum8, imin8, imax8, init8, ifn8, icutoff8

kflag slider8table ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \
    init1, ifn1,..., ictlnum8, imin8, imax8, init8, ifn8

kflag slider8tablef ichan, ioutTable, ioffset, ictlnum1, imin1, imax1, \

```



```
init1, ifn1, icutoff1, .... , ictlnum8, imin8, imax8, init8, ifn8, icutoff8
```

```
k1, k2, ...., k16 sliderKawai imin1, imax1, init1, ifn1, \  
    imin2, imax2, init2, ifn2, ..., imin16, imax16, init16, ifn16
```

Spectral Processing:STFT.

```
htableseg ifn1, idur1, ifn2 [, idur2] [, ifn3] [...]
```

```
ares pvadd ktimpnt, kfmmod, ifilcod, ifn, ibins [, ibinoffset] \  
    [, ibinincr] [, iextractmode] [, ifreqlim] [, igatefn]
```

```
pvbufread ktimpnt, ifile
```

```
ares pvcross ktimpnt, kfmmod, ifile, kampscale1, kampscale2 [, ispecwp]
```

```
ares pvinterp ktimpnt, kfmmod, ifile, kfreqscale1, kfreqscale2, \  
    kampscale1, kampscale2, kfreqinterp, kampinterp
```

```
ares pvoc ktimpnt, kfmmod, ifilcod [, ispecwp] [, iextractmode] \  
    [, ifreqlim] [, igatefn]
```

```
kfreq, kamp pvread ktimpnt, ifile, ibin
```

```
htableseg ifn1, idur1, ifn2 [, idur2] [, ifn3] [...]
```

```
tablexseg ifn1, idur1, ifn2 [, idur2] [, ifn3] [...]
```

```
ares vpvoc ktimpnt, kfmmod, ifile [, ispecwp] [, ifn]
```

Spectral Processing:LPC.

```
ares lpfreson asig, kfrqratio
```

```
lpinterp islot1, islot2, kmix
```

```
krmsr, krms0, kerr, kcps lpread ktimpnt, ifilcod [, inpoles] [, ifrmrate]
```

```
ares lpreson asig
```

```
lpslot islot
```

Spectral Processing:Non-Standard.

```
wsig specaddm wsig1, wsig2 [, imul2]

wsig specdiff wsignin

specdisp wsig, iprd [, iwtflg]

wsig specfilt wsignin, ifhtim

wsig spechist wsignin

koct, kamp specptrk wsig, kvar, ilo, ihi, istr, idbthresh, inptls, \
    irolloff [, iodd] [, iconfs] [, interp] [, ifprd] [, iwtflg]

wsig specscal wsignin, ifscale, ifthresh

ksum specsum wsig [, interp]

wsig spectrum xsig, iprd, iocts, ifrqa [, iq] [, ihann] [, idbout] \
    [, idsprd] [, idsinrs]
```

Spectral Processing:Streaming.

```
fsig binit fin, isize

ftrks partials ffr, fphs, kthresh, kminpts, kmaxgap, imaxtracks

ares pvsadsyn fsrc, inoscs, kfmod [, ibinoffset] [, ibinincr] [, iinit]

fsig pvsanal ain, ifftsize, ioverlap, iwinsize, iwintype [, iformat] [, iinit]

fsig pvsarp fsigin, kbin, kdepth, kgain

fsig pvsbandp fsigin, xlowcut,
    xlowfull, xhighfull, xhighcut[, ktype]

fsig pvsbandr fsigin, xlowcut,
    xlowfull, xhighfull, xhighcut[, ktype]

kamp, kfr pvsbin fsig, kbin
```

fsig pvsblur fsigin, kblurtime, imaxdel

ihandle, ktime pvsbuffer fsig, ilen

fsig pvdbufread ktime, khandle[, ilo, ihi]

fsig pvscale fsigin, kscal[, ikeepform, igain]]

kcent pvscent fsig

fsig pvscross fsrc, fdest, kamp1, kamp2

fsig pvsdemix fleft, fright, kpos, kwidth, ipoints

fsig pvdiskinSFname, ktscal, kgain[, ioffset, ichan]

pvsdisp fsig[, ibins, iwtflg]

fsig pvfilter fsigin, fsigfil, kdepth[, igain]

fsig pvsfread ktimpt, ifn [, ichan]

fsig pvsfreeze fsigin, kfreeza, kfreezf

pvsftr fsrc, ifna [, ifnf]

kflag pvsftw fsrc, ifna [, ifnf]

pvsfwrite fsig, ifile

fsig pvshift fsigin, kshift, klowest[, ikeepform, igain]

ffr, fphs pvsifd ain, ifftsize, ihopsize, iwintype[, iscal]

fsig pvsin kchan[, isize, iolap, iwinsize, iwintype, iformat]

ioverlap, inumbins, iwinsize, iformat pvsinfo fsrc

fsig pvsinit isize[, iolap, iwinsize, iwintype, iformat]

fsig pvsmaska fsrc, ifn, kdepth

`fsig pvsmix fsigin1, fsigin2`

`fsig pvsmooth fsigin, kacf, kfcf`

`fsig pvsmorph fsigl, fsig2, kampint, kfrqint`

`fsig pvsosc kamp, kfreq, ktype, isize [,ioverlap] [, iwinsize] [, iwintype] [, iformat]`

`pvsout fsig, kchan`

`kfr, kamp pvspitch fsig, kthresh`

`fsig pvstencil fsigin, kgain, klevel, iftable`

`fsig pvsvoc famp, fexc, kdepth, kgain`

`ares pvsynth fsrc, [iinit]`

`asig resyn fin, kscal, kpitch, kmaxtracks, ifn`

`asig sinsyn fin, kscal, kmaxtracks, ifn`

`asig tradsyn fin, kscal, kpitch, kmaxtracks, ifn`

`fsig trcross fin1, fin2, ksearch,kdepth[,kmode]`

`fsig trfilter fin, kamnt, ifn`

`fsig, kfr,kamp trhighest fin1, kscal`

`fsig, kfr,kamp trlowest fin1, kscal`

`fsig trmix fin1, fin2`

`fsig trscale fin, kpitch[, kgain]`

`fsig trshift fin, kpshift[, kgain]`

`fsiglow, fsighi trsplit fin, ksplit[, kgainlow, kgainhigh]`

Spectral Processing:ATS.

```
ar ATSadd ktimepnt, kfmod, iatsfile, ifn, ipartials[, ipartialoffset, \
    ipartialincr, igatefn]

ar ATSaddnz ktimepnt, iatsfile, ibands[, ibandoffset, ibandincr]

ATSbufread ktimepnt, kfmod, iatsfile, ipartials[, ipartialoffset, \
    ipartialincr]

ar ATScross ktimepnt, kfmod, iatsfile, ifn, kmylev, kbuflev, ipartials \
    [, ipartialoffset, ipartialincr]

idata ATSinfo iatsfile, ilocation

kamp ATSinterpread kfreq

kfrq, kamp ATSpartialtap ipartialnum

kfreq, kamp ATSread ktimepnt, iatsfile, ipartial

kenergy ATSreadnz ktimepnt, iatsfile, iband

ar ATSSinnoi ktimepnt, ksinlev, knzlev, kfmod, iatsfile, ipartials \
    [, ipartialoffset, ipartialincr]
```

Spectral Processing:Loris.

```
lorismorph isrcidx, itgtidx, istoreidx, kfreqmorphenv, kampmorphenv, kbwmorphenv

ar lorisplay ireadidx, kfreqenv, kampenv, kbwenv

lorisread ktimpnt, ifilcod, istoreidx, kfreqenv, kampenv, kbwenv[, ifadetime]
```

Strings:Definition.

```
Sdst strget indx

strset iarg, istring
```

Strings:Manipulation.

puts Sstr, ktrig[, inonl]

Sdst sprintf Sfmt, xarg1[, xarg2[, ...]]

Sdst sprintfk Sfmt, xarg1[, xarg2[, ...]]

Sdst sprintfk Sfmt, xarg1[, xarg2[, ...]]

Sdst strcat Ssrc1, Ssrc2

Sdst strcatk Ssrc1, Ssrc2

ires strcmp S1, S2

kres strcmpk S1, S2

Sdst strcpy Ssrc

Sdst = Ssrc

Sdst strcpyk Ssrc

ipos strindex S1, S2

kpos strindexk S1, S2

ilen strlen Sstr

klen strlenk Sstr

ipos strrindex S1, S2

kpos strrindexk S1, S2

Sdst strsub Ssrc[, istart[, iend]]

Sdst strsubk Ssrc, kstart, kend

Strings:Conversion.

ichr strchr Sstr[, ipos]

kchr strchark Sstr[, kpos]

Sdst strlower Ssrc

Sdst strlowerk Ssrc

ir strtod Sstr

ir strtod indx

kr strtodk Sstr

kr strtodk kndx

ir strtol Sstr

ir strtol indx

kr strtolk Sstr

kr strtolk kndx

Sdst strupper Ssrc

Sdst strupperk Ssrc

Vectorial:Tables.

vtaba andx, ifn, aout1 [, aout2, aout3, , aoutN]

vtabi indx, ifn, iout1 [, iout2, iout3, , ioutN]

vtabk kndx, ifn, kout1 [, kout2, kout3, , koutN]

vtablelk kfn,kout1 [, kout2, kout3, , koutN]

vtablea andx, kfn, kinterp, ixmode, aout1 [, aout2, aout3, , aoutN]

vtablei indx, ifn, interp, ixmode, iout1 [, iout2, iout3, , ioutN]

```

vtablek  kndx, kfn, kinterp, ixmode, kout1 [, kout2, kout3, .... , koutN ]

vtablewa andx, kfn, ixmode, ainarg1 [, ainarg2, ainarg3 , .... , ainargN ]

vtablewi  indx, ifn, ixmode, inarg1 [, inarg2, inarg3 , .... , inargN ]

vtablewk  kndx, kfn, ixmode, kinarg1 [, kinarg2, kinarg3 , .... , kinargN ]

vtabwa  andx, ifn, ainarg1 [, ainarg2, ainarg3 , .... , ainargN ]

vtabwi  indx, ifn, inarg1 [, inarg2, inarg3 , .... , inargN ]

vtabwk  kndx, ifn, kinarg1 [, kinarg2, kinarg3 , .... , kinargN ]

```

Vectorial:Scalar operations.

```

vadd  ifn, kval, kelements [, kdstoffset] [, kverbose]

vadd_i  ifn, ival, ielements [, idstoffset]

vexp  ifn, kval, kelements [, kdstoffset] [, kverbose]

vexp_i  ifn, ival, ielements[, idstoffset]

vmult  ifn, kval, kelements [, kdstoffset] [, kverbose]

vmult_i  ifn, ival, ielements [, idstoffset]

vpow  ifn, kval, kelements [, kdstoffset] [, kverbose]

vpow_i  ifn, ival, ielements [, idstoffset]

```

Vectorial:Vectorial operations.

```

vaddv  ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]

vaddv_i  ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]

vcopy  ifn, ifn2, kelements [, kdstoffset] [, ksrcoffset] [, kverbose]

```


`vcopy_i ifn, ifn2, ielements [,idstoffset, isrcoffset]`

`vdivv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]`

`vdivv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]`

`vexpv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]`

`vexpv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]`

`vmap ifn1, ifn2, ielements [,idstoffset, isrcoffset]`

`vmultv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]`

`vmultv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]`

`vpowv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]`

`vpowv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]`

`vsubv ifn1, ifn2, kelements [, kdstoffset] [, ksrcoffset] [,kverbose]`

`vsubv_i ifn1, ifn2, ielements [, idstoffset] [, isrcoffset]`

Vectorial:Envelopes.

`vexpseg ifnout, ielements, ifn1, idurl, ifn2 [, idur2, ifn3 [...]]`

`vlinseg ifnout, ielements, ifn1, idurl, ifn2 [, idur2, ifn3 [...]]`

Vectorial:Limiting and Wrapping.

`vlimit ifn, kmin, kmax, ielements`

`vmirror ifn, kmin, kmax, ielements`

`vwrap ifn, kmin, kmax, ielements`

Vectorial:Delay Paths.

```
kout vdelayk  iksig, kdel, imaxdel [, iskip, imode]
```

```
vecdelay  ifn, ifnIn, ifnDel, ielements, imaxdel [, iskip]
```

```
vport ifn, khtime, ielements [, ifnInit]
```

Vectorial:Random.

```
vrandh  ifn, krange, kcps, ielements [, idstoffset] [, iseed]  
        [, isize] [, ioffset]
```

```
vrandi  ifn, krange, kcps, ielements [, idstoffset] [, iseed]  
        [, isize] [, ioffset]
```

Vectorial:Cellular Automata.

```
vcella ktrig, kreinit, ioutFunc, initStateFunc, \  
        iRuleFunc, ielements, irulelen [, iradius]
```

Zak Patch System.

```
zacl kfirst, klast
```

```
zakinit isizea, isizek
```

```
ares zamod asig, kzamod
```

```
ares zar kndx
```

```
ares zarg kndx, kgain
```

```
zaw asig, kndx
```

```
zawm asig, kndx [, imix]
```

```
ir zir indx
```

ziw isig, indx

ziwm isig, indx [, imix]

zkcl kfirst, klast

kres zkmod ksig, kzkmod

kres zkr kndx

zkw ksig, kndx

zkwm ksig, kndx [, imix]

Plugin Hosting:DSSI and LADSPA.

dssiactivate ihandle, ktoggle

aout1 [, aout2, aout3, aout4] dssiaudio ihandle, ain1 [,ain2, ain3, ain4]

dssictls ihandle, iport, kvalue, ktrigger

ihandle dssiinit ilibraryname, ipluginindex [, iverbose]

dssilist

Plugin Hosting:VST.

aout1,aout2 vstaudio instance, [ain1, ain2]

aout1,aout2 vstaudiog instance, [ain1, ain2]

vstbankload instance, ipath

vstedit instance

vstinfo instance

instance vstinit ilibrarypath [,iverbose]

vstmidiout instance, kstatus, kchan, kdata1, kdata2

vstnote instance, kchan, knote, kveloc, kdur

vstparamset instance, kparam, kvalue

kvalue vstparamget instance, kparam

vstprogset instance, kprogram

OSC.

ihandle OSCinit iport

kans OSClisten ihandle, idest, itype [, xdata1, xdata2, ...]

OSCsend kwhen, ihost, iport, idestination, itype [, kdata1, kdata2, ...]

Network.

remoteport iportnum

asig sockrecv iport, ilength

asigl, asigr sockrecvs iport, ilength

asig strecv Sipaddr, iport

socksend asig, Sipaddr, iport, ilength

socksends asigl, asigr, Sipaddr, iport,
 ilength

stsend asig, Sipaddr, iport

Remote Opcodes.

insglobalisource, instrnum [,instrnum...]

insremotidestination, isource, instrnum [,instrnum...]

midglobalisource, instrnum [,instrnum...]

midremotidestination, isource, instrnum [,instrnum...]

Mixer Opcodes.

MixerClear

kgain MixerGetLevel isend, ibuss

asignal MixerReceive ibuss, ichannel

MixerSend asignal, isend, ibuss, ichannel

MixerSetLevel isend, ibuss, kgain

Python Opcodes.

pyassign "variable", kvalue

pyassigni "variable", ivalue

pylassign "variable", kvalue

pylassigni "variable", ivalue

pyassignt ktrigger, "variable", kvalue

pylassignt ktrigger, "variable", kvalue

kresult pyeval "expression"

iresult pyevali "expression"

kresult pyleval "expression"

iresult pylevali "expression"

kresult pyevalt ktrigger, "expression"

kresult pylevalt ktrigger, "expression"

pyexec "filename"

pyexeci "filename"

`pylexec "filename"`

`pylexeci "filename"`

`pyexec ktrigger, "filename"`

`plyexec ktrigger, "filename"`

`pyinit`

`pyrun "statement"`

`pyruni "statement"`

`pylrun "statement"`

`pylruni "statement"`

`pyrunt ktrigger, "statement"`

`pylrun ktrigger, "statement"`

Image Processing Opcodes.

`iimagenum imagecreate iwidth, iheight`

`imagefree iimagenum`

`ared agreeen ablue imagegetpixel iimagenum, ax, ay`

`kred kgreen kblue imagegetpixel iimagenum, kx, ky`

`iimagenum imageload filename`

`imagesave iimagenum, filename`

`imagegetpixel iimagenum, ax, ay, ared, agreeen, ablue`

`imagegetpixel iimagenum, kx, ky, kred, kgreen, kblue`

`iwidth iheight imagesize iimagenum`

Miscellaneous.

`ires system_i itrigger, Scmd, [inowait]`

kres system ktrig, Scmd, [knowait]

Utilities.

csound -U atsa [flags] infilename outfilename

cs [-OPTIONS] <name> [CSOUND OPTIONS ...]

csb64enc [OPTIONS ...] infile1 [infile2 [...]]

csound -U cvanal [flags] infilename outfilename

cvanal [flags] infilename outfilename

dnoise [flags] -i noise_ref_file -o output_soundfile input_soundfile

envext [-flags] soundfile

csound -U envext [-flags] soundfile

extractor [OPTIONS ...] infile

het_export het_file ctext_file

csound -U het_export het_file ctext_file

het_import ctext_file het_file

csound -U het_import ctext_file het_file

csound -U hetro [flags] infilename outfilename

hetro [flags] infilename outfilename

csound -U lpanal [flags] infilename outfilename

lpanal [flags] infilename outfilename

makecsd [OPTIONS ...] infile1 [infile2 [...]]

mixer [OPTIONS ...] infile [[OPTIONS...] infile] ...

pv_export pv_file ctext_file

`csound -U pv_export pv_file ctext_file`

`pv_import ctext_file pv_file`

`csound -U pv_import ctext_file pv_file`

`csound -U pvanal [flags] infilename outfilename`

`pvanal [flags] infilename outfilename`

`csound -U pvlook [flags] infilename`

`pvlook [flags] infilename`

`scale [OPTIONS ...] infile`

`csound -U sdif2ad [flags] infilename outfilename`

`csound -U sndinfo [options] soundfilenames ...`

`sndinfo [options] soundfilenames ...`

`srconv [flags] infile`

Appendix A. Pitch Conversion

Table A.1. Pitch Conversion

Note	Hz	cpspch	MIDI
C-1	8.176	3.00	0
C#-1	8.662	3.01	1
D-1	9.177	3.02	2
D#-1	9.723	3.03	3
E-1	10.301	3.04	4
F-1	10.913	3.05	5
F#-1	11.562	3.06	6
G-1	12.250	3.07	7
G#-1	12.978	3.08	8
A-1	13.750	3.09	9
A#-1	14.568	3.10	10
B-1	15.434	3.11	11
C0	16.352	4.00	12
C#0	17.324	4.01	13
D0	18.354	4.02	14
D#0	19.445	4.03	15
E0	20.602	4.04	16
F0	21.827	4.05	17
F#0	23.125	4.06	18
G0	24.500	4.07	19
G#0	25.957	4.08	20
A0	27.500	4.09	21
A#0	29.135	4.10	22
B0	30.868	4.11	23
C1	32.703	5.00	24
C#1	34.648	5.01	25
D1	36.708	5.02	26
D#1	38.891	5.03	27
E1	41.203	5.04	28
F1	43.654	5.05	29
F#1	46.249	5.06	30
G1	48.999	5.07	31
G#1	51.913	5.08	32
A1	55.000	5.09	33
A#1	58.270	5.10	34
B1	61.735	5.11	35
C2	65.406	6.00	36
C#2	69.296	6.01	37
D2	73.416	6.02	38

Pitch Conversion

Note	Hz	cpspch	MIDI
D#2	77.782	6.03	39
E2	82.407	6.04	40
F2	87.307	6.05	41
F#2	92.499	6.06	42
G2	97.999	6.07	43
G#2	103.826	6.08	44
A2	110.000	6.09	45
A#2	116.541	6.10	46
B2	123.471	6.11	47
C3	130.813	7.00	48
C#3	138.591	7.01	49
D3	146.832	7.02	50
D#3	155.563	7.03	51
E3	164.814	7.04	52
F3	174.614	7.05	53
F#3	184.997	7.06	54
G3	195.998	7.07	55
G#3	207.652	7.08	56
A3	220.000	7.09	57
A#3	233.082	7.10	58
B3	246.942	7.11	59
C4	261.626	8.00	60
C#4	277.183	8.01	61
D4	293.665	8.02	62
D#4	311.127	8.03	63
E4	329.628	8.04	64
F4	349.228	8.05	65
F#4	369.994	8.06	66
G4	391.995	8.07	67
G#4	415.305	8.08	68
A4	440.000	8.09	69
A#4	466.164	8.10	70
B4	493.883	8.11	71
C5	523.251	9.00	72
C#5	554.365	9.01	73
D5	587.330	9.02	74
D#5	622.254	9.03	75
E5	659.255	9.04	76
F5	698.456	9.05	77
F#5	739.989	9.06	78
G5	783.991	9.07	79
G#5	830.609	9.08	80
A5	880.000	9.09	81
A#5	932.328	9.10	82

Pitch Conversion

Note	Hz	cpSPch	MIDI
B5	987.767	9.11	83
C6	1046.502	10.00	84
C#6	1108.731	10.01	85
D6	1174.659	10.02	86
D#6	1244.508	10.03	87
E6	1318.510	10.04	88
F6	1396.913	10.05	89
F#6	1479.978	10.06	90
G6	1567.982	10.07	91
G#6	1661.219	10.08	92
A6	1760.000	10.09	93
A#6	1864.655	10.10	94
B6	1975.533	10.11	95
C7	2093.005	11.00	96
C#7	2217.461	11.01	97
D7	2349.318	11.02	98
D#7	2489.016	11.03	99
E7	2637.020	11.04	100
F7	2793.826	11.05	101
F#7	2959.955	11.06	102
G7	3135.963	11.07	103
G#7	3322.438	11.08	104
A7	3520.000	11.09	105
A#7	3729.310	11.10	106
B7	3951.066	11.11	107
C8	4186.009	12.00	108
C#8	4434.922	12.01	109
D8	4698.636	12.02	110
D#8	4978.032	12.03	111
E8	5274.041	12.04	112
F8	5587.652	12.05	113
F#8	5919.911	12.06	114
G8	6271.927	12.07	115
G#8	6644.875	12.08	116
A8	7040.000	12.09	117
A#8	7458.620	12.10	118
B8	7902.133	12.11	119
C9	8372.018	13.00	120
C#9	8869.844	13.01	121
D9	9397.273	13.02	122
D#9	9956.063	13.03	123
E9	10548.08	13.04	124
F9	11175.30	13.05	125
F#9	11839.82	13.06	126

Pitch Conversion

Note	Hz	cpspch	MIDI
G9	12543.85	13.07	127

Appendix B. Sound Intensity Values

Table B.1. Sound Intensity Values (for a 1000 Hz tone)

Dynamics	Intensity (W/m ²)	Level (dB)
pain	1	120
fff	10 ⁻²	100
f	10 ⁻⁴	80
p	10 ⁻⁶	60
ppp	10 ⁻⁸	40
threshold	10 ⁻¹²	0

Appendix C. Formant Values

Table C.1. alto “a”

Values	f1	f2	f3	f4	f5
freq (Hz)	800	1150	2800	3500	4950
amp (dB)	0	-4	-20	-36	-60
bw (Hz)	80	90	120	130	140

Table C.2. alto “e”

Values	f1	f2	f3	f4	f5
freq (Hz)	400	1600	2700	3300	4950
amp (dB)	0	-24	-30	-35	-60
bw (Hz)	60	80	120	150	200

Table C.3. alto “i”

Values	f1	f2	f3	f4	f5
freq (Hz)	350	1700	2700	3700	4950
amp (dB)	0	-20	-30	-36	-60
bw (Hz)	50	100	120	150	200

Table C.4. alto “o”

Values	f1	f2	f3	f4	f5
freq (Hz)	450	800	2830	3500	4950
amp (dB)	0	-9	-16	-28	-55
bw (Hz)	70	80	100	130	135

Table C.5. alto “u”

Values	f1	f2	f3	f4	f5
freq (Hz)	325	700	2530	3500	4950
amp (dB)	0	-12	-30	-40	-64
bw (Hz)	50	60	170	180	200

Table C.6. bass “a”

Values	f1	f2	f3	f4	f5
freq (Hz)	600	1040	2250	2450	2750
amp (dB)	0	-7	-9	-9	-20

Values	f1	f2	f3	f4	f5
bw (Hz)	60	70	110	120	130

Table C.7. bass “e”

Values	f1	f2	f3	f4	f5
freq (Hz)	400	1620	2400	2800	3100
amp (dB)	0	-12	-9	-12	-18
bw (Hz)	40	80	100	120	120

Table C.8. bass “i”

Values	f1	f2	f3	f4	f5
freq (Hz)	250	1750	2600	3050	3340
amp (dB)	0	-30	-16	-22	-28
bw (Hz)	60	90	100	120	120

Table C.9. bass “o”

Values	f1	f2	f3	f4	f5
freq (Hz)	400	750	2400	2600	2900
amp (dB)	0	-11	-21	-20	-40
bw (Hz)	40	80	100	120	120

Table C.10. bass “u”

Values	f1	f2	f3	f4	f5
freq (Hz)	350	600	2400	2675	2950
amp (dB)	0	-20	-32	-28	-36
bw (Hz)	40	80	100	120	120

Table C.11. countertenor “a”

Values	f1	f2	f3	f4	f5
freq (Hz)	660	1120	2750	3000	3350
amp (dB)	0	-6	-23	-24	-38
bw (Hz)	80	90	120	130	140

Table C.12. countertenor “e”

Values	f1	f2	f3	f4	f5
freq (Hz)	440	1800	2700	3000	3300
amp (dB)	0	-14	-18	-20	-20

Values	f1	f2	f3	f4	f5
bw (Hz)	70	80	100	120	120

Table C.13. countertenor “i”

Values	f1	f2	f3	f4	f5
freq (Hz)	270	1850	2900	3350	3590
amp (dB)	0	-24	-24	-36	-36
bw (Hz)	40	90	100	120	120

Table C.14. countertenor “o”

Values	f1	f2	f3	f4	f5
freq (Hz)	430	820	2700	3000	3300
amp (dB)	0	-10	-26	-22	-34
bw (Hz)	40	80	100	120	120

Table C.15. countertenor “u”

Values	f1	f2	f3	f4	f5
freq (Hz)	370	630	2750	3000	3400
amp (dB)	0	-20	-23	-30	-34
bw (Hz)	40	60	100	120	120

Table C.16. soprano “a”

Values	f1	f2	f3	f4	f5
freq (Hz)	800	1150	2900	3900	4950
amp (dB)	0	-6	-32	-20	-50
bw (Hz)	80	90	120	130	140

Table C.17. soprano “e”

Values	f1	f2	f3	f4	f5
freq (Hz)	350	2000	2800	3600	4950
amp (dB)	0	-20	-15	-40	-56
bw (Hz)	60	100	120	150	200

Table C.18. soprano “i”

Values	f1	f2	f3	f4	f5
freq (Hz)	270	2140	2950	3900	4950
amp (dB)	0	-12	-26	-26	-44

Values	f1	f2	f3	f4	f5
bw (Hz)	60	90	100	120	120

Table C.19. soprano “o”

Values	f1	f2	f3	f4	f5
freq (Hz)	450	800	2830	3800	4950
amp (dB)	0	-11	-22	-22	-50
bw (Hz)	40	80	100	120	120

Table C.20. soprano “u”

Values	f1	f2	f3	f4	f5
freq (Hz)	325	700	2700	3800	4950
amp (dB)	0	-16	-35	-40	-60
bw (Hz)	50	60	170	180	200

Table C.21. tenor “a”

Values	f1	f2	f3	f4	f5
freq (Hz)	650	1080	2650	2900	3250
amp (dB)	0	-6	-7	-8	-22
bw (Hz)	80	90	120	130	140

Table C.22. tenor “e”

Values	f1	f2	f3	f4	f5
freq (Hz)	400	1700	2600	3200	3580
amp (dB)	0	-14	-12	-14	-20
bw (Hz)	70	80	100	120	120

Table C.23. tenor “i”

Values	f1	f2	f3	f4	f5
freq (Hz)	290	1870	2800	3250	3540
amp (dB)	0	-15	-18	-20	-30
bw (Hz)	40	90	100	120	120

Table C.24. tenor “o”

Values	f1	f2	f3	f4	f5
freq (Hz)	400	800	2600	2800	3000
amp (dB)	0	-10	-12	-12	-26

Values	f1	f2	f3	f4	f5
bw (Hz)	70	80	100	130	135

Table C.25. tenor “u”

Values	f1	f2	f3	f4	f5
freq (Hz)	350	600	2700	2900	3300
amp (dB)	0	-20	-17	-14	-26
bw (Hz)	40	60	100	120	120

Appendix D. Modal Frequency Ratios

Contributed by Scott Lindroth

John Bower, a student of Scott Lindroth, compiled this list of modal frequencies for various objects and materials. Some modes work better than others, and most need to be in a particular frequency range to sound plausible. Caveat emptor.

In general, wooden objects will not sound "wooden" unless a stochastic component is present in the sound (try banded waveguides). Nonetheless, some of the wooden objects make wonderful metallic instruments as well.

This ratios can be useful together with opcodes like *mode* or *streson*.

Table D.1. Modal Frequency Ratios

Instrument	Modal Frequency Ratios
Dahina tabla	[1, 2.89, 4.95, 6.99, 8.01, 9.02]
Bayan tabla	[1, 2.0, 3.01, 4.01, 4.69, 5.63]
Red Cedar wood plate	[1, 1.47, 2.09, 2.56]
Redwood wood plate	[1, 1.47, 2.11, 2.57]
Douglas Fir wood plate	[1, 1.42, 2.11, 2.47]
uniform wooden bar	[1, 2.572, 4.644, 6.984, 9.723, 12]
uniform aluminum bar	[1, 2.756, 5.423, 8.988, 13.448, 18.680]
Xylophone	[1, 3.932, 9.538, 16.688, 24.566, 31.147]
Vibraphone 1	[1, 3.984, 10.668, 17.979, 23.679, 33.642]
Vibraphone 2	[1, 3.997, 9.469, 15.566, 20.863, 29.440]
Chalandi plates	([62, 107, 360, 460, 863] Hz +-2Hz) [1, 1.72581, 5.80645, 7.41935, 13.91935] ratios
tibetan bowl (180mm)	([221, 614, 1145, 1804, 2577, 3456, 4419] Hz) 934g, 180mm [1, 2.77828, 5.18099, 8.16289, 11.66063, 15.63801, 19.99 ratios
tibetan bowl (152 mm)	([314, 836, 1519, 2360, 3341, 4462, 5696] Hz) 563g, 152mm [1, 2.66242, 4.83757, 7.51592, 10.64012, 14.21019, 18.14027] ratios
tibetan bowl (140 mm)	([528, 1460, 2704, 4122, 5694] Hz) 557g, 140mm [1, 2.76515, 5.12121, 7.80681, 10.78409] ratios
Wine Glass	[1, 2.32, 4.25, 6.63, 9.38]
small handbell	([1312.0, 1314.5, 2353.3, 2362.9, 3306.5, 3309.4, 3923.8, 3928.2, 4966.6, 4993.7, 5994.4, 6003.0, 6598.9, 6619.7, 7971.7, 7753.2, 8413.1, 8453.3, 9292.4, 9305.2, 9602.3, 9912.4] Hz)

Instrument	Modal Frequency Ratios
	[1, 1.0019054878049, 1.7936737804878, 1.8009908536585, 2.5201981707317, 2.5224085365854, 2.9907012195122, 2.9940548780488, 3.7855182926829, 3.8061737804878, 4.5689024390244, 4.5754573170732, 5.0296493902439, 5.0455030487805, 6.0759908536585, 5.9094512195122, 6.4124237804878, 6.4430640243902, 7.0826219512195, 7.0923780487805, 7.3188262195122, 7.5551829268293] ratios
spinel sphere with diameter of 3.6675mm	<p>([977.25, 1003.16, 1390.13, 1414.93, 1432.84, 1465.34, 1748.48, 1834.20, 1919.90, 1933.64, 1987.20, 2096.48, 2107.10, 2202.08, 2238.40, 2280.10, 0 /*2290.53 calculated*/ , 2400.88, 2435.85, 2507.80, 2546.30, 2608.55, 2652.35, 2691.70, 2708.00] Hz)</p> <p>[1, 1.026513174725, 1.4224916858532, 1.4478690202098, 1.4661959580455, 1.499452545408, 1.7891839345101, 1.8768994627782, 1.9645945254541, 1.9786543873113, 2.0334612432847, 2.1452852391916, 2.1561524686621, 2.2533435661294, 2.2905090816065, 2.3331798413917, 0, 2.4567715528268, 2.4925556408289, 2.5661806088514, 2.6055768738808, 2.6692760296751, 2.7140956766436, 2.7543617293425, 2.7710411870043] ratios</p>
pot lid	[1, 3.2, 6.23, 6.27, 9.92, 14.15] ratios

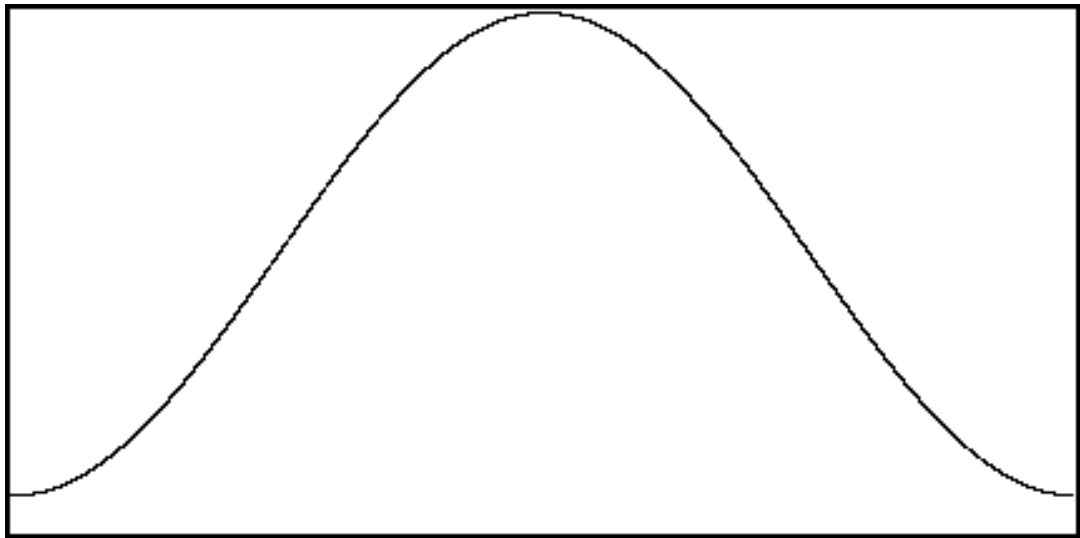
Appendix E. Window Functions

Windowing functions are used for analysis, and as waveform envelopes, particularly in granular synthesis. Window functions are built in to some opcodes, but others require a function table to generate the window. *GEN20* is used for this purpose. The diagram of each window below, is accompanied by the f statement used to generate the it.

Hamming.

Example E.1. Hamming window function statement

```
f81 0 8192 20 1 1
```

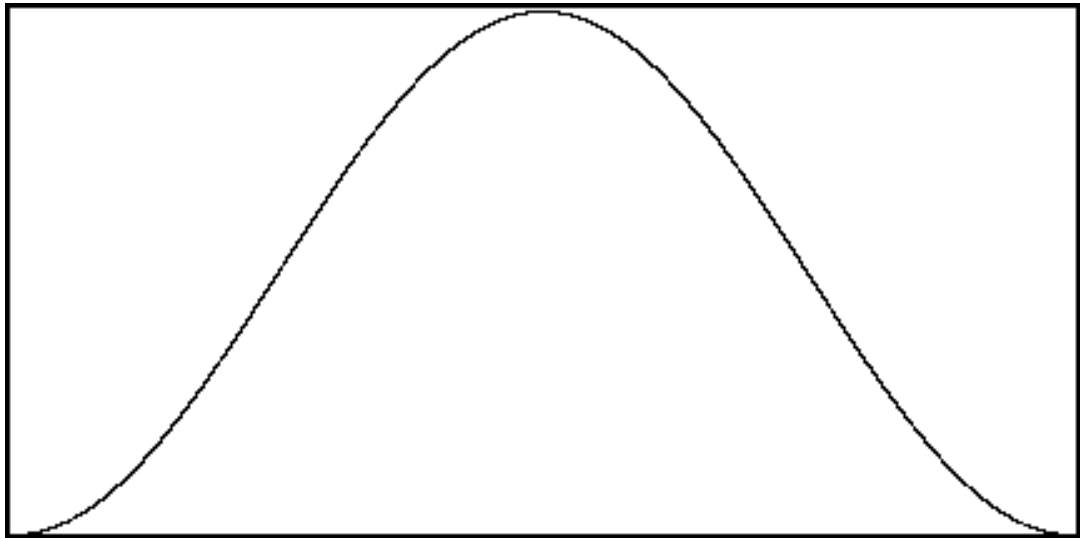


Hamming Window Function.

Hanning.

Example E.2. Hanning window function statement

```
f82 0 8192 20 2 1
```

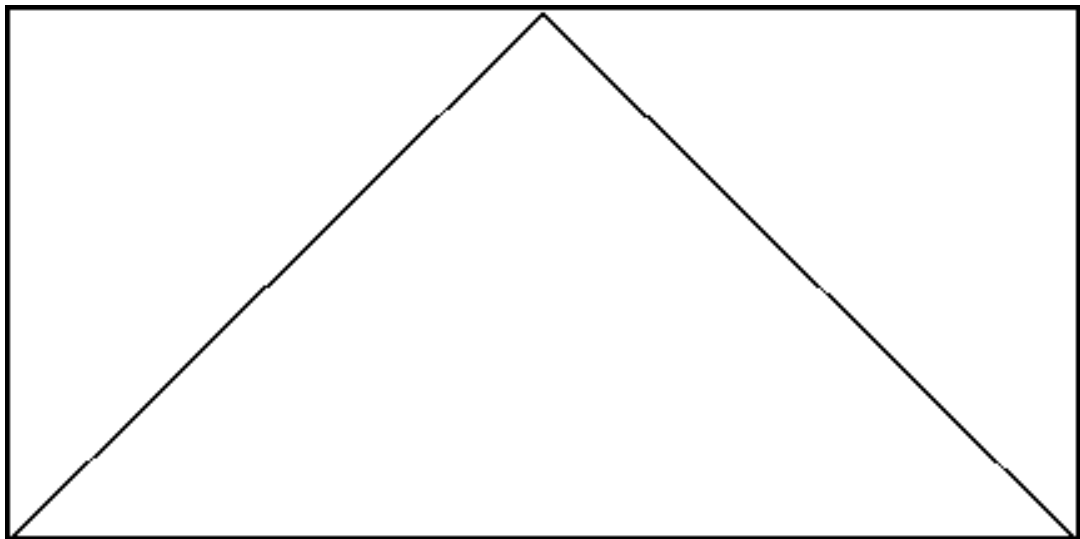


Hanning Window Function

Bartlett.

Example E.3. Bartlett window function statement

```
f83  0  8192  20  3  1
```

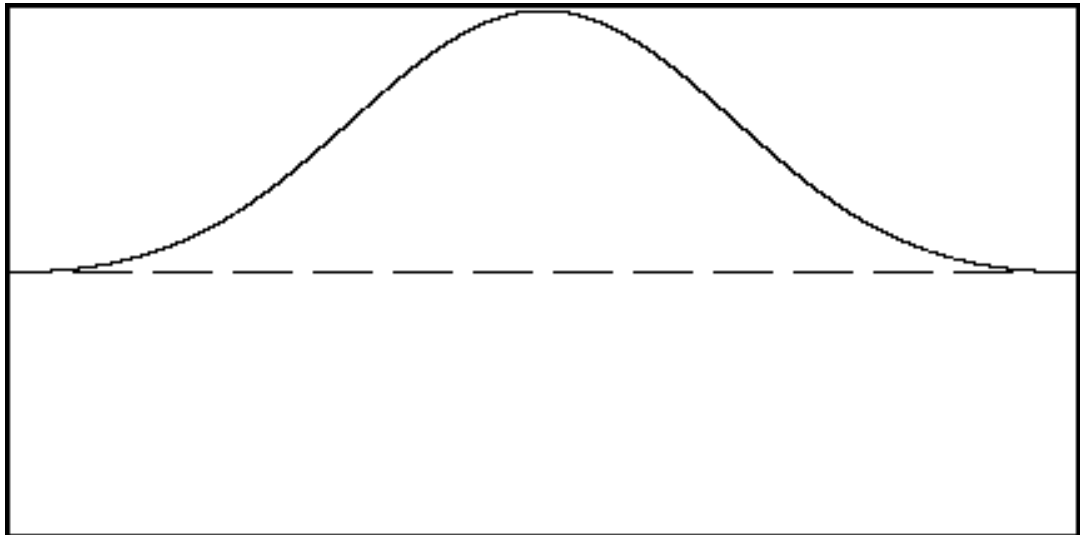


Bartlett Window Function

Blackman.

Example E.4. Blackman window function statement

```
f84  0  8192  20  4  1
```

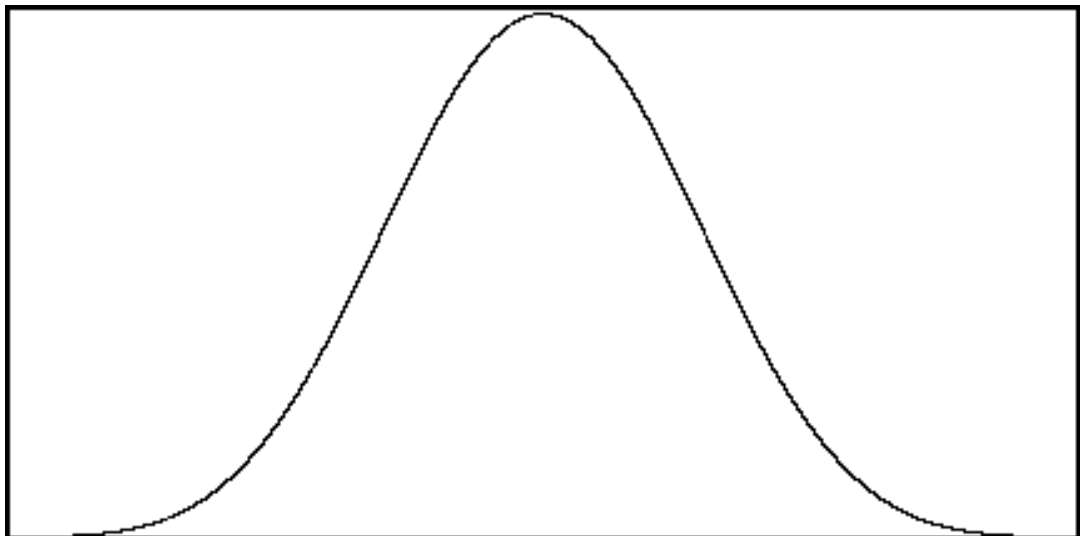


Blackman Window Function

Blackman-Harris.

Example E.5. Blackman-Harris window function statement

```
f85    0    8192    20    5    1
```

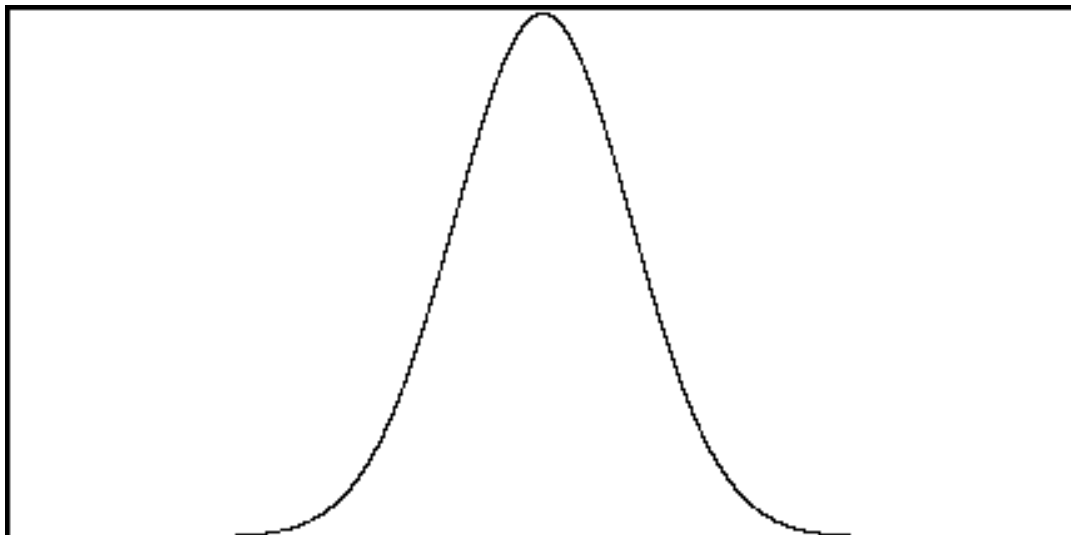


Blackman-Harris Window Function

Gaussian.

Example E.6. Gaussian window function statement

```
f86    0    8192    20    6    1
```



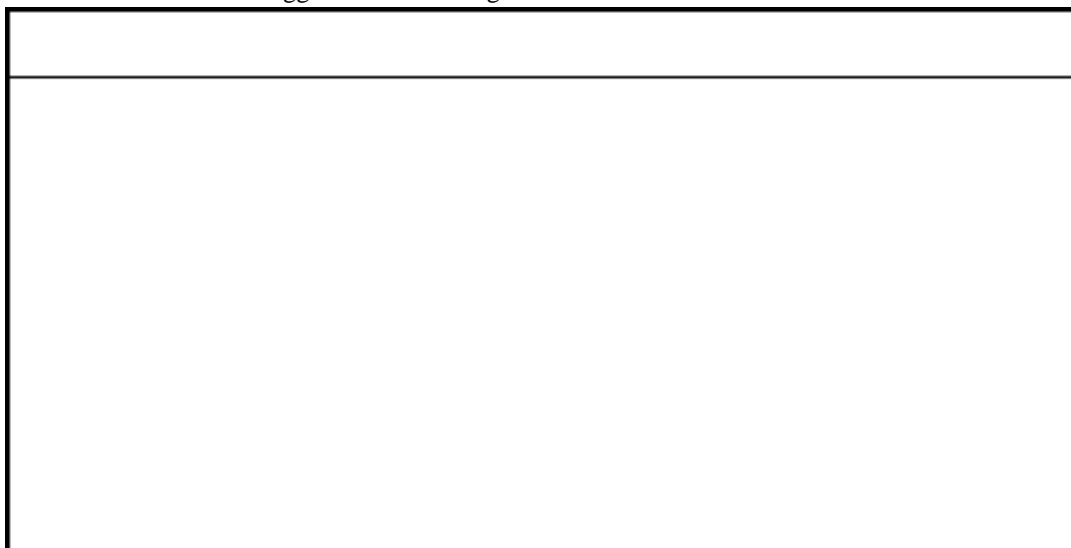
Gaussian Window Function

Rectangle.

Example E.7. Rectangle window function statement

```
f88  0  8192  -20  8  .1
```

Note: Vertical scale is exaggerated in this diagram.

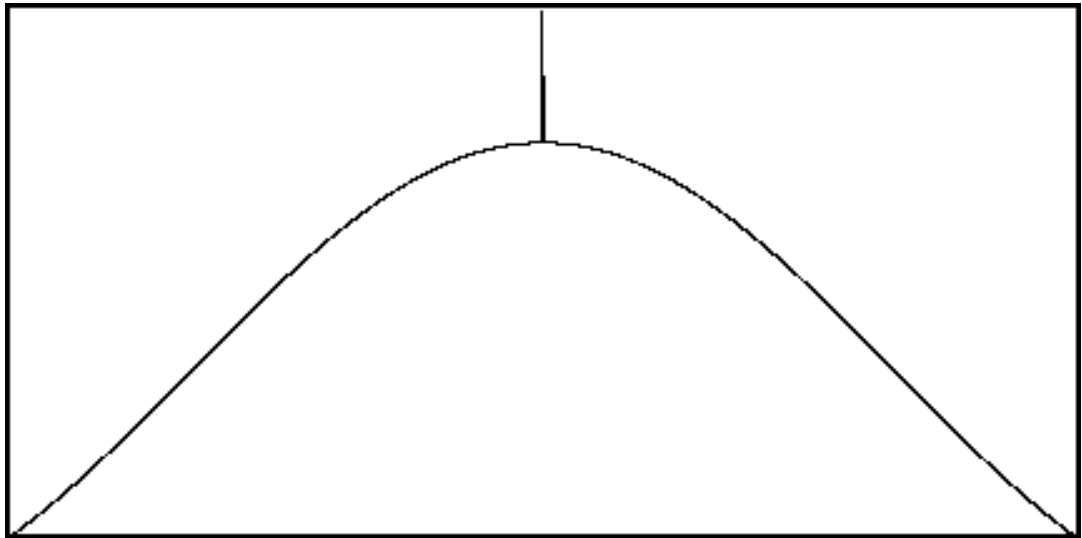


Rectangle Window Function

Sync.

Example E.8. Sync window function statement

```
f89  0  4096  -20  9  .75
```

Sync Window Function

Appendix F. SoundFont2 File Format

Beginning with Csound Version 4.07, *Csound supports the SoundFont2 sample file format*. SoundFont2 (or SF2) is a widespread standard which allows encoding banks of wavetable-based sounds into a binary file. In order to understand the usage of these opcodes, the user must have some knowledge of the SF2 format, so a brief description of this format follows.

The SF2 format is made by generator and modulator objects. All current Csound opcodes regarding SF2 support the generator function only.

There are several levels of generators having a hierarchical structure. The most basic kind of generator object is a sample. Samples may or may not be be looped, and are associated with a MIDI note number, called the base-key. When a sample is associated with a range of MIDI note numbers, a range of velocities, a transposition (coarse and fine tuning), a scale tuning, and a level scaling factor, the sample and its associations make up a “split.” A set of splits, together with a name, make up an “instrument.” When an instrument is associated with a key range, a velocity range, a level scaling factor, and a transposition, the instrument and its associations make up a “layer.” A set of layers, together with a name, makes up a “preset.” Presets are normally the final sound-generating structures ready for the user. They generate sound according to the settings of their lower-level components.

Both sample data and structure data is embedded in the same SF2 binary file. A single SF2 file can contain up to a maximum of 128 banks of 128 preset programs, for a total of 16384 presets in one SF2 file. The maximum number of layers, instruments, splits, and samples is not defined, and probably is only limited by the computer's memory.

Appendix G. Csound Double (64-bit) vs. Float (32-bit)

Csound can be built to use 64-bit DOUBLES internally to do processing versus regular Csound's 32-bit FLOATS. This larger resolution for processing internally yields a much "cleaner" sound but at the expense of extended processing time. Because it does require much longer to process, Csound compiled for doubles is typically used after a work is finished for a final production run. If you are using csound for realtime output, you should use the 32-bit (float) version, which provides faster output. For offline rendering, you can use either, but for the final master, the 64-bit version will produce higher quality output.

The actual resolution should be the same as for the type of the audio sample variable. For "float" Csound, that is a 32-bit, single-precision floating point number. It has 24 bits of precision in the mantissa. For "double" Csound, that is a 64-bit, double-precision floating point number in the mantissa. It has 52 bits of precision. For each decimal digit, there is between 3 and 4 bits. So, there are 7 digits of precision for "float" and 16 digits of precision for "double."

For each multiplication or division, depending on whether the operands are integers, rational numbers, repeating decimals, or irrational decimals, there may or may not be rounding error in the result. If there is rounding error, it is at most 1 bit of lost precision per operation (in addition to any rounding in the binary representation of rational or real numbers).

For float samples, if the signal stays within the mantissa, the signal to noise ratio is 6.02 times 24, or 144 dB. At worst, each operation will create 6.02 dB of additional noise due to rounding error. Our ears have an effective dynamic range of 120 to 130 dB, but we like our music compressed to a dynamic range of AT MOST 60 dB (and usually much less, say 20 dB). That gives us $(144 - 60) / 6.02 = \text{about } 10$ worst-case operations before we could possibly hear any degradation. In practice, we have several times that many operations before we hear any degradation or noise.

For double samples, if the signal stays within the mantissa, the signal to noise ratio is 6.02 times 52, or 313 dB. At worst, each operation will create 6.02 dB of additional noise due to rounding error. That gives us $(313 - 60) / 6.02 = \text{about } 42$, in practice several times that, of operations before there is any audible degradation or noise.

But if you trace the number of arithmetic operations in typical Csound instruments or other software synthesizers, the very complex instruments definitely are pushing into the range of audible degradation on good monitors with float, so it is no surprise that in some cases blindfold, controlled ABX testing confirms *occasional* audible differences between music synthesized with "float" Csound versus the same music synthesized with "double" Csound. Equally, it is no surprise that there are easily audible differences between digital and analog implementations of the same synthesis algorithms.

There is a good deal more "digital signal processing headroom" with double Csound, and therefore it should be used for all music rendered for critical listening. The float version should only be used where its speed advantage of about 15% is critical for real-time performance.

Notes On Using Csound built for double precision

1. hetro, PVOC-EX analysis and pvanal files generated for 32-bit Csound (float) will work with 64-bit Csound (double precision).
2. lpanal and cvanal files generated for Csound will not work with Csound64.

Glossary

G

Guard Point

A guard point is the last position on a function table. If the length is, say 1024, the table will have 1024+1 (1025) points: the extra point is the guard point.

In any case, for a 1024-point table, the first point is index 0 and the last 1023; index 1024 is not really used)

The reason for a guard-point is that some opcodes interpolate to obtain a table value, in which case, when the table index is say, 1023.5, we need the value of the 1024 pos in order to interpolate.

There are two ways of filling this point (writing the value that goes in it):

1. Default way: by copying the value of the 1st point in the table
2. Extended Guard-Point: extending the contour of the table (continuing to calculate the table for one extra point)

In general the first mode is used for wrap-around applications, such as an oscillator (which loops continuously reading the table). The second use is for one-shot readouts, such as envelopes, where the last point needs to be interpolated correctly following the table contour (we are not looping back to the beginning of the table)