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Faust Libraries

NOTE: this documentation was automatically generated using the script

`generateDoc`. This script depends on pandoc and html-xml-utils.

This page provides information on how to use the Faust libraries.

The /libraries folder contains the different Faust libraries. If you wish to add
your own functions to this library collection, you can refer to the “Contributing”
section providing a set of coding conventions.

WARNING: These libraries replace the “old” Faust libraries. They are still being
beta tested so you might encounter bugs while using them. If your codes still
use the “old” Faust libraries, you might want to try to use Bart Brouns’ script
that automatically makes an old Faust code compatible with the new libraries:

https://github.com/magnetophon/faustCompressors/blob/master/newlib.sh. If
you find a bug, please report it at rmichon_at_ccrma_dot_stanford_dot_edu. Thanks ;)

Using the Faust Libraries

The easiest and most standard way to use the Faust libraries is to import stdfaust.lib in your Faust code:

```
import("stdfaust.lib");
```

This will give you access to all the Faust libraries through a series of environments:

- `sf`: all.lib
- `an`: analyzers.lib
- `ba`: basics.lib
- `co`: compressors.lib
- `de`: delays.lib
- `dm`: demos.lib
- `dx`: dx7.lib
- `en`: envelopes.lib
- `fi`: filters.lib
- `ho`: hoa.lib
- `it`: interpolators.lib
- `ma`: maths.lib
- `ef`: misceffects.lib
- `os`: oscillators.lib
- `no`: noises.lib
- `pf`: phaflangers.lib
- `pm`: physmodels.lib
- `rm`: reducemaps.lib
- `re`: reverbs.lib
- `ro`: routes.lib
- `si`: signals.lib
- `so`: soundfiles.lib
- `sp`: spats.lib
- `sy`: synths.lib
- `ve`: vaeffects.lib
- `wa`: webaudio.lib
- `vl`: version.lib

Environments can then be used as follows in your Faust code:

```
import("stdfaust.lib");
process = os.osc(440);
```

In this case, we’re calling the `osc` function from oscillators.lib.

You can also access all the functions of all the libraries directly using the `sf` environment:
import("stdfaust.lib");
process = sf.osc(440);

Alternatively, environments can be created by hand:

os = library("oscillators.lib");
process = os.osc(440);

Finally, libraries can be simply imported in the Faust code (not recommended):

import("oscillators.lib");
process = osc(440);

Contributing

If you wish to add a function to any of these libraries or if you plan to add a new library, make sure that you follow the following conventions:

New Functions

- All functions must be preceded by a markdown documentation header respecting the following format (open the source code of any of the libraries for an example):

  ```
  //-----------------functionName-------------------
  // Description
  //
  // ### Usage
  //
  // ```
  // Usage Example
  //
  // Where:
  //
  // * argument1: argument 1 description
  //-------------------------------

- Every time a new function is added, the documentation should be updated simply by running `make doclib`.
- The environment system (e.g. `os.osc`) should be used when calling a function declared in another library (see the section on Using the Faust Libraries).
- Try to reuse existing functions as much as possible.
- If you have any question, send an e-mail to rmichon_at_ccrma_dot_stanford_dot_edu.
New Libraries

- Any new “standard” library should be declared in `stdfaust.lib` with its own environment (2 letters - see `stdfaust.lib`).
- Any new “standard” library must be added to `generateDoc`.
- Functions must be organized by sections.
- Any new library should at least declare a name and a version.
- The comment based markdown documentation of each library must respect the following format (open the source code of any of the libraries for an example):

```markdown
//############### libraryName ##################
// Description
//
// * Section Name 1
// * Section Name 2
// * ...
//
// It should be used using the `[...,]` environment:
//
// ```
// [...]
// library("libraryName");
// process = [...].functionCall;
// ```
//
// Another option is to import `stdfaust.lib` which already contains the `[...,]` environment:
//
// ```
// ```
// import("stdfaust.lib");
// process = [...].functionCall;
// ```
//```

//##############################################
//================= Section Name ===============
// Description
//==============================================
```

- If you have any question, send an e-mail to rmichon_at_ccrma_dot_stanford_dot_edu.

General Organization

Only the libraries that are considered to be “standard” are documented:

- analyzers.lib
- basics.lib
- compressors.lib
- delays.lib
• demos.lib
• dx7.lib
• envelopes.lib
• filters.lib
• hoa.lib
• interpolators.lib
• maths.lib
• misceffects.lib
• oscillators.lib
• noises.lib
• phaflangers.lib
• physmodels.lib
• reducemaps.lib
• reverbs.lib
• routes.lib
• signals.lib
• soundfiles.lib
• spats.lib
• synths.lib
• tonestacks.lib (not documented but example in /examples/misc)
• tubes.lib (not documented but example in /examples/misc)
• vaeffects.lib
• webaudio.lib
• version.lib

Other deprecated libraries such as music.lib, etc. are present but are not documented to not confuse new users.

The documentation of each library can be found in /documentation/library.html or in /documentation/library.pdf.

A global version number for the standard libraries is defined in version.lib. It follows the semantic versioning structure: MAJOR, MINOR, PATCH. The MAJOR number is increased when we make incompatible changes. The MINOR number is increased when we add functionality in a backwards compatible manner, and the PATCH number when we make backwards compatible bug fixes. By looking at the generated code or the diagram of process = vl.version; one can see the current version of the libraries.

The /examples directory contains all the examples from the /examples folder of the Faust distribution as well as new ones. Most of them were updated to reflect the coding conventions described in the next section. Examples are organized by types in different folders. The /old folder contains examples that are fully deprecated, probably because they were integrated to the libraries and fully rewritten (see freverb.dsp for example). Examples using deprecated libraries were integrated to the general tree but a warning comment was added at their beginning to point readers to the right library and function.
Coding Conventions

In order to have a uniformized library system, we established the following conventions (that hopefully will be followed by others when making modifications to them :-) ).

Documentation

- All the functions that we want to be “public” are documented.
- We used the `faust2md` “standards” for each library: `//###` for main title (library name - equivalent to # in markdown), `//###` for section declarations (equivalent to ## in markdown) and `//---` for function declarations (equivalent to #### in markdown - see `basics.lib` for an example).
- Sections in function documentation should be declared as #### markdown title.
- Each function documentation provides a “Usage” section (see `basics.lib`).

Library Import

To prevent cross-references between libraries we generalized the use of the `library("")` system for function calls in all the libraries. This means that everytime a function declared in another library is called, the environment corresponding to this library needs to be called too. To make things easier, a `stdfaust.lib` library was created and is imported by all the libraries:

```plaintext
an = library("analyzers.lib");
ba = library("basics.lib");
co = library("compressors.lib");
de = library("delays.lib");
dm = library("demos.lib");
dx = library("dx7.lib");
en = library("envelopes.lib");
fi = library("filters.lib");
ho = library("hoa.lib");
it = library("interpolators.lib");
ma = library("maths.lib");
ef = library("misceffects.lib");
os = library("oscillators.lib");
no = library("noises.lib");
pf = library("phaflangers.lib");
pm = library("physmodels.lib");
rm = library("reducemaps.lib");
re = library("reverbs.lib");
ro = library("routes.lib");
sp = library("spats.lib");
si = library("signals.lib");
so = library("soundfiles.lib");
```
sy = library("synths.lib");
ve = library("vaeffects.lib");
wa = library("webaudio.lib");
vl = library("version.lib");

For example, if we wanted to use the smooth function which is now declared in signals.lib, we would do the following:

import("stdfaust.lib");

process = si.smooth(0.999);

This standard is only used within the libraries: nothing prevents coders to still import signals.lib directly and call smooth without ro., etc. It means symbols and function names defined within a library have to be unique to not collide with symbols of any other libraries.

“Demo” Functions

“Demo” functions are placed in demos.lib and have a built-in user interface (UI). Their name ends with the _demo suffix. Each of these function have a .dsp file associated to them in the /examples folder.

Any function containing UI elements should be placed in this library and respect these standards.

“Standard” Functions

“Standard” functions are here to simplify the life of new (or not so new) Faust coders. They are declared in libraries/doc/standardFunctions.md and allow to point programmers to preferred functions to carry out a specific task. For example, there are many different types of lowpass filters declared in filters.lib and only one of them is considered to be standard, etc.

Copyright / License

Now that Faust libraries are less author specific, each function will normally have its own copyright-and-license line in the library source (the .lib file, such as analyzers.lib). If not, see if the function is defined within a section of the .lib file stating the license in source-code comments. If not, then the copyright and license given at the beginning of the .lib file may be assumed, when present. If not, run git blame on the .lib file and ask the person who last edited the function!

Note that it is presently possible for a library function released under one license to utilize another library function having some different license. There is presently no indication of this situation in the Faust compiler output, but such notice is planned. For now, library contributors should strive to use only library
functions having compatible licenses, and concerned end-users must manually
determine the union of licenses applicable to the library functions they are using.

**Standard Functions**

Dozens of functions are implemented in the Faust libraries and many of them are
very specialized and not useful to beginners or to people who only need to use
Faust for basic applications. This section offers an index organized by categories
of the “standard Faust functions” (basic filters, effects, synthesizers, etc.). This
index only contains functions without a user interface (UI). Faust functions with
a built-in UI can be found in demos.lib.

**Analysis Tools**

<table>
<thead>
<tr>
<th>Function Type</th>
<th>Function Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Amplitude Follower</td>
<td>an.amp_follower</td>
<td>Classic analog audio envelope follower</td>
</tr>
<tr>
<td>Octave Analyzers</td>
<td>an.mth_octave_analyzer[N]</td>
<td>Octave analyzers</td>
</tr>
</tbody>
</table>

**Basic Elements**

<table>
<thead>
<tr>
<th>Function Type</th>
<th>Function Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Beats</td>
<td>ba.beat</td>
<td>Pulses at a specific tempo</td>
</tr>
<tr>
<td>Block</td>
<td>si.block</td>
<td>Terminate n signals</td>
</tr>
<tr>
<td>Break Point Function</td>
<td>ba.bpf</td>
<td>Break Point Function (BPF)</td>
</tr>
<tr>
<td>Bus</td>
<td>si.bus</td>
<td>Bus of n signals</td>
</tr>
<tr>
<td>Bypass (Mono)</td>
<td>ba.bypass1</td>
<td>Mono bypass</td>
</tr>
<tr>
<td>Bypass (Stereo)</td>
<td>ba.bypass2</td>
<td>Stereo bypass</td>
</tr>
<tr>
<td>Count Elements</td>
<td>ba.count</td>
<td>Count elements in a list</td>
</tr>
<tr>
<td>Count Down</td>
<td>ba.countdown</td>
<td>Samples count down</td>
</tr>
<tr>
<td>Count Up</td>
<td>ba.countup</td>
<td>Samples count up</td>
</tr>
<tr>
<td>Delay (Integer)</td>
<td>de.delay</td>
<td>Integer delay</td>
</tr>
<tr>
<td>Delay (Float)</td>
<td>de.fdelay</td>
<td>Fractional delay</td>
</tr>
<tr>
<td>Down Sample</td>
<td>ba.downSample</td>
<td>Down sample a signal</td>
</tr>
<tr>
<td>Impulsify</td>
<td>ba.impulsify</td>
<td>Turns a signal into an impulse</td>
</tr>
<tr>
<td>Sample and Hold</td>
<td>ba.sAndH</td>
<td>Sample and hold</td>
</tr>
<tr>
<td>Signal Crossing</td>
<td>ro.cross</td>
<td>Cross n signals</td>
</tr>
<tr>
<td>Smoother (Default)</td>
<td>si.smoo</td>
<td>Exponential smoothing</td>
</tr>
<tr>
<td>Smoother</td>
<td>si.smooth</td>
<td>Exponential smoothing with controllable pole</td>
</tr>
<tr>
<td>Take Element</td>
<td>ba.take</td>
<td>Take en element from a list</td>
</tr>
<tr>
<td>Time</td>
<td>ba.time</td>
<td>A simple timer</td>
</tr>
</tbody>
</table>
## Conversion

<table>
<thead>
<tr>
<th>Function Type</th>
<th>Function Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dB to Linear</td>
<td>ba.db2linear</td>
<td>Converts dB to linear values</td>
</tr>
<tr>
<td>Linear to dB</td>
<td>ba.linear2db</td>
<td>Converts linear values to dB</td>
</tr>
<tr>
<td>MIDI Key to Hz</td>
<td>ba.midikey2hz</td>
<td>Converts a MIDI key number into a frequency</td>
</tr>
<tr>
<td>Hz to MIDI Key</td>
<td>ba.hz2midikey</td>
<td>Converts a frequency into MIDI key number</td>
</tr>
<tr>
<td>Pole to T60</td>
<td>ba.pole2tau</td>
<td>Converts a pole into a time constant (t60)</td>
</tr>
<tr>
<td>Samples to Seconds</td>
<td>ba.samp2sec</td>
<td>Converts samples to seconds</td>
</tr>
<tr>
<td>Seconds to Samples</td>
<td>ba.sec2samp</td>
<td>Converts seconds to samples</td>
</tr>
<tr>
<td>T60 to Pole</td>
<td>ba.tau2pole</td>
<td>Converts a time constant (t60) into a pole</td>
</tr>
</tbody>
</table>

## Effects

<table>
<thead>
<tr>
<th>Function Type</th>
<th>Function Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Wah</td>
<td>ve.autowah</td>
<td>Auto-Wah effect</td>
</tr>
<tr>
<td>Compressor</td>
<td>co.compressor_mono</td>
<td>Dynamic range compressor</td>
</tr>
<tr>
<td>Distortion</td>
<td>ef.cubicnl</td>
<td>Cubic nonlinearity distortion</td>
</tr>
<tr>
<td>Crybaby</td>
<td>ve.crybaby</td>
<td>Crybaby wah pedal</td>
</tr>
<tr>
<td>Echo</td>
<td>ef.echo</td>
<td>Simple echo</td>
</tr>
<tr>
<td>Flanger</td>
<td>pf.flanger_stereo</td>
<td>Flanging effect</td>
</tr>
<tr>
<td>Gate</td>
<td>ef.gate_mono</td>
<td>Mono signal gate</td>
</tr>
<tr>
<td>Limiter</td>
<td>co.limiter_1176_R4_mono</td>
<td>Limiter</td>
</tr>
<tr>
<td>Phaser</td>
<td>pf.phaser2_stereo</td>
<td>Phaser effect</td>
</tr>
<tr>
<td>Reverb (FDN)</td>
<td>re.fdnrev0</td>
<td>Feedback delay network reverberator</td>
</tr>
<tr>
<td>Reverb (Freeverb)</td>
<td>re.mono_freeverb</td>
<td>Most “famous” Schroeder reverberator</td>
</tr>
<tr>
<td>Reverb (Simple)</td>
<td>re.jcrev</td>
<td>Simple Schroeder reverberator</td>
</tr>
<tr>
<td>Reverb (Zita)</td>
<td>re.zita_revl_stereo</td>
<td>High quality FDN reverberator</td>
</tr>
<tr>
<td>Panner</td>
<td>sp.panner</td>
<td>Linear stereo panner</td>
</tr>
<tr>
<td>Pitch Shift</td>
<td>ef.transpose</td>
<td>Simple pitch shifter</td>
</tr>
<tr>
<td>Panner</td>
<td>sp.spat</td>
<td>N outputs spatializer</td>
</tr>
<tr>
<td>Speaker Simulator</td>
<td>ef.speakerbp</td>
<td>Simple speaker simulator</td>
</tr>
<tr>
<td>Stereo Width</td>
<td>ef.stereo_width</td>
<td>Stereo width effect</td>
</tr>
<tr>
<td>Vocoder</td>
<td>ve.vocoder</td>
<td>Simple vocoder</td>
</tr>
<tr>
<td>Wah</td>
<td>ve.wah4</td>
<td>Wah effect</td>
</tr>
</tbody>
</table>

## Envelope Generators

<table>
<thead>
<tr>
<th>Function Type</th>
<th>Function Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ADSR</td>
<td>en.adsr</td>
<td>Attack/Decay/Sustain/Release envelope generator</td>
</tr>
<tr>
<td>AR</td>
<td>en.ar</td>
<td>Attack/Release envelope generator</td>
</tr>
<tr>
<td>ASR</td>
<td>en.asr</td>
<td>Attack/Sustain/Release envelope generator</td>
</tr>
</tbody>
</table>
### Filters

<table>
<thead>
<tr>
<th>Function Type</th>
<th>Function Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Exponential</td>
<td>en.smoothEnvelope</td>
<td>Exponential envelope generator</td>
</tr>
<tr>
<td>Bandpass (Butterworth)</td>
<td>fi.bandpass</td>
<td>Generic butterworth bandpass</td>
</tr>
<tr>
<td>Bandpass (Resonant)</td>
<td>fi.resonbp</td>
<td>Virtual analog resonant bandpass</td>
</tr>
<tr>
<td>Bandstop (Butterworth)</td>
<td>fi.bandstop</td>
<td>Generic butterworth bandstop</td>
</tr>
<tr>
<td>Biquad</td>
<td>fi.tf2</td>
<td>“Standard” biquad filter</td>
</tr>
<tr>
<td>Comb (Allpass)</td>
<td>fi.allpass_fcomb</td>
<td>Schroeder allpass comb filter</td>
</tr>
<tr>
<td>Comb (Feedback)</td>
<td>fi.fb_fcomb</td>
<td>Feedback comb filter</td>
</tr>
<tr>
<td>Comb (Feedforward)</td>
<td>fi.ff_fcomb</td>
<td>Feed-forward comb filter.</td>
</tr>
<tr>
<td>DC Blocker</td>
<td>fi.dbcblocker</td>
<td>Default dc blocker</td>
</tr>
<tr>
<td>Filterbank</td>
<td>fi.filterbank</td>
<td>Generic filter bank</td>
</tr>
<tr>
<td>FIR (Arbitrary Order)</td>
<td>fi.fir</td>
<td>Nth-order FIR filter</td>
</tr>
<tr>
<td>High Shelf</td>
<td>fi.highshelf</td>
<td>High shelf</td>
</tr>
<tr>
<td>Highpass (Butterworth)</td>
<td>fi.highpass</td>
<td>Nth-order Butterworth highpass</td>
</tr>
<tr>
<td>Highpass (Resonant)</td>
<td>fi.resonhp</td>
<td>Virtual analog resonant highpass</td>
</tr>
<tr>
<td>IIR (Arbitrary Order)</td>
<td>fi.iir</td>
<td>Nth-order IIR filter</td>
</tr>
<tr>
<td>Level Filter</td>
<td>fi.levelfilter</td>
<td>Dynamic level lowpass</td>
</tr>
<tr>
<td>Low Shelf</td>
<td>fi.lowshelf</td>
<td>Low shelf</td>
</tr>
<tr>
<td>Lowpass (Butterworth)</td>
<td>fi.lowpass</td>
<td>Nth-order Butterworth lowpass</td>
</tr>
<tr>
<td>Lowpass (Resonant)</td>
<td>fi.resonlp</td>
<td>Virtual analog resonant lowpass</td>
</tr>
<tr>
<td>Notch Filter</td>
<td>fi.notch</td>
<td>Simple notch filter</td>
</tr>
<tr>
<td>Peak Equalizer</td>
<td>fi.peak_eq</td>
<td>Peaking equalizer section</td>
</tr>
</tbody>
</table>

### Oscillators/Sound Generators

<table>
<thead>
<tr>
<th>Function Type</th>
<th>Function Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Impulse</td>
<td>os.impulse</td>
<td>Generate an impulse on start-up</td>
</tr>
<tr>
<td>Impulse Train</td>
<td>os.imptrain</td>
<td>Band-limited impulse train</td>
</tr>
<tr>
<td>Phasor</td>
<td>os.phasor</td>
<td>Simple phasor</td>
</tr>
<tr>
<td>Pink Noise</td>
<td>no.pink_noise</td>
<td>Pink noise generator</td>
</tr>
<tr>
<td>Pulse Train</td>
<td>os.pulsetrain</td>
<td>Band-limited pulse train</td>
</tr>
<tr>
<td>Pulse Train (Low Frequency)</td>
<td>os.lf_imptrain</td>
<td>Low-frequency pulse train</td>
</tr>
<tr>
<td>Sawtooth</td>
<td>os.sawtooth</td>
<td>Band-limited sawtooth wave</td>
</tr>
<tr>
<td>Sawtooth (Low Frequency)</td>
<td>os.lf_saw</td>
<td>Low-frequency sawtooth wave</td>
</tr>
<tr>
<td>Sine (Filter-Based)</td>
<td>os.oscs</td>
<td>Sine oscillator (filter-based)</td>
</tr>
<tr>
<td>Sine (Table-Based)</td>
<td>os.osc</td>
<td>Sine oscillator (table-based)</td>
</tr>
<tr>
<td>Square</td>
<td>os.square</td>
<td>Band-limited square wave</td>
</tr>
<tr>
<td>Square (Low Frequency)</td>
<td>os.lf_squarewave</td>
<td>Low-frequency square wave</td>
</tr>
<tr>
<td>Function Type</td>
<td>Function Name</td>
<td>Description</td>
</tr>
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**Primitives**

**User Interface Primitives**

**button**

Creates a button in the user interface. The `button` is a primitive circuit with one output and no input. The signal produced by the `button` is 0 when not pressed and 1 while pressed.

**Usage**

```plaintext
button("play") : _;
```

Where "play" is the name of the `button` in the interface.

**checkbox**

Creates a checkbox in the user interface. The `checkbox` is a primitive circuit with one output and no input. The signal produced by the checkbox is 0 when not checked and 1 when checked.

**Usage**

```plaintext
checkbox("play") : _;
```

Where "play" is the name of the `checkbox` in the interface.
hslider
Creates a horizontal slider in the user interface. The hslider is a primitive circuit with one output and no input. hslider produces a signal between a minimum and a maximum value based on the position of the slider cursor.

Usage
hslider("volume",-10,-70,12,0.1) : _;
Where volume is the name of the slider in the interface, -10 the default value of the slider when the program starts, -70 the minimum value, 12 the maximum value, and 0.1 the step the determines the precision of the control.

nentry
Creates a numerical entry in the user interface. The nentry is a primitive circuit with one output and no input. nentry produces a signal between a minimum and a maximum value based on the user input.

Usage
nentry("volume",-10,-70,12,0.1) : _;
Where volume is the name of the numerical entry in the interface, -10 the default value of the entry when the program starts, -70 the minimum value, 12 the maximum value, and 0.1 the step the determines the precision of the control.

vslider
Creates a vertical slider in the user interface. The vslider is a primitive circuit with one output and no input. vslider produces a signal between a minimum and a maximum value based on the position of the slider cursor.

Usage
vslider("volume",-10,-70,12,0.1) : _;
Where volume is the name of the slider in the interface, -10 the default value of the slider when the program starts, -70 the minimum value, 12 the maximum value, and 0.1 the step the determines the precision of the control.

analyzers.lib
Faust Analyzers library. Its official prefix is an.
Amplitude Tracking

(an.)amp_follower

Classic analog audio envelope follower with infinitely fast rise and exponential decay. The amplitude envelope instantaneously follows the absolute value going up, but then floats down exponentially. amp_follower is a standard Faust function.

Usage

_ : amp_follower(rel) : _

Where:

- rel: release time = amplitude-envelope time-constant (sec) going down

Reference


(an.)amp_follower_ud

Envelope follower with different up and down time-constants (also called a “peak detector”).

Usage

_ : amp_follower_ud(att,rel) : _

Where:

- att: attack time = amplitude-envelope time constant (sec) going up
- rel: release time = amplitude-envelope time constant (sec) going down

Note

We assume rel >> att. Otherwise, consider rel ~ max(rel,att). For audio, att is normally faster (smaller) than rel (e.g., 0.001 and 0.01). Use amp_follower_ar below to remove this restriction.

Reference

(an.) amp_follower_ar

Envelope follower with independent attack and release times. The release can be shorter than the attack (unlike in amp_follower_ud above).

Usage

_ : amp_follower_ar(att,rel) : _;

- Author Jonatan Liljedahl, revised by RM

__________________________

Spectrum-Analyzers

Spectrum-analyzers split the input signal into a bank of parallel signals, one for each spectral band. They are related to the Mth-Octave Filter-Banks in filters.lib. The documentation of this library contains more details about the implementation. The parameters are:

- M: number of band-slices per octave (>1)
- N: total number of bands (>2)
- ftop = upper bandlimit of the Mth-octave bands (<SR/2)

In addition to the Mth-octave output signals, there is a highpass signal containing frequencies from ftop to SR/2, and a “dc band” lowpass signal containing frequencies from 0 (dc) up to the start of the Mth-octave bands. Thus, the N output signals are

highpass(ftop), MthOctaveBands(M,N-2,ftop), dcBand(ftop*2^(-M*(N-1)))

A Spectrum-Analyzer is defined here as any band-split whose bands span the relevant spectrum, but whose band-signals do not necessarily sum to the original signal, either exactly or to within an allpass filtering. Spectrum analyzer outputs are normally at least nearly “power complementary”, i.e., the power spectra of the individual bands sum to the original power spectrum (to within some negligible tolerance).

Increasing Channel Isolation

Go to higher filter orders - see Regalia et al. or Vaidyanathan (cited below) regarding the construction of more aggressive recursive filter-banks using elliptic or Chebyshev prototype filters.

References

- “Multirate Systems and Filter Banks”, P. Vaidyanathan, Prentice-Hall, 1993
• Elementary filter theory: https://ccrma.stanford.edu/~jos/filters/

(an.) \texttt{mth\_octave\_analyzer}

Octave analyzer. \texttt{mth\_octave\_analyzer}[N] are standard Faust functions.

Usage

\_ : \texttt{mth\_octave\_analyzer}(O,M,ftop,N) : \texttt{par}(i,N,\_); // Oth-order Butterworth
\_ : \texttt{mth\_octave\_analyzer6e}(M,ftop,N) : \texttt{par}(i,N,\_); // 6th-order elliptic

Also for convenience:

\_ : \texttt{mth\_octave\_analyzer3}(M,ftop,N) : \texttt{par}(i,N,\_); // 3d-order Butterworth
\_ : \texttt{mth\_octave\_analyzer5}(M,ftop,N) : \texttt{par}(i,N,\_); // 5th-roder Butterworth
\texttt{mth\_octave\_analyzer\_default} = \texttt{mth\_octave\_analyzer6e};

Where:

- \texttt{O}: order of filter used to split each frequency band into two
- \texttt{M}: number of band-slices per octave
- \texttt{ftop}: highest band-split crossover frequency (e.g., 20 kHz)
- \texttt{N}: total number of bands (including dc and Nyquist)

\underline{Mth-Octave Spectral Level}

Spectral Level: Display (in bar graphs) the average signal level in each spectral band.

(an.) \texttt{mth\_octave\_spectral\_level6e}

Spectral level display.

Usage:

\_ : \texttt{mth\_octave\_spectral\_level6e}(M,ftop,NBands,tau,dB\_offset) : \_;

Where:

- \texttt{M}: bands per octave
- \texttt{ftop}: lower edge frequency of top band
- \texttt{NBands}: number of passbands (including highpass and dc bands),
- \texttt{tau}: spectral display averaging-time (time constant) in seconds,
- \texttt{dB\_offset}: constant dB offset in all band level meters.

Also for convenience:

\texttt{mth\_octave\_spectral\_level\_default} = \texttt{mth\_octave\_spectral\_level6e};
\texttt{spectral\_level} = \texttt{mth\_octave\_spectral\_level}(2,10000,20);
A bunch of special cases based on the different analyzer functions described above:

third_octave_analyzer(N) = mth_octave_analyzer_default(3,10000,N);
third_octave_filterbank(N) = mth_octave_filterbank_default(3,10000,N);
half_octave_analyzer(N) = mth_octave_analyzer_default(2,10000,N);
half_octave_filterbank(N) = mth_octave_filterbank_default(2,10000,N);
octave_filterbank(N) = mth_octave_filterbank_default(1,10000,N);
octave_analyzer(N) = mth_octave_analyzer_default(1,10000,N);

Usage
See mth_octave_spectral_level_demo in demos.lib.

Arbitary-Crossover Filter-Banks and Spectrum Analyzers

These are similar to the Mth-octave analyzers above, except that the band-split frequencies are passed explicitly as arguments.

(an.)analyzer
Analyzer.

Usage
_ : analyzer(O,freqs) : par(i,N,_); // No delay equalizer

Where:
• O: band-split filter order (ODD integer required for filterbank[i])
• freqs: (fc1,fc2,…,fcNs) [in numerically ascending order], where Ns=N-1 is the number of octave band-splits (total number of bands N=Ns+1).

If frequencies are listed explicitly as arguments, enclose them in parens:
_ : analyzer(3,(fc1,fc2)) : _,_,_
(an.)goertzelOpt
Optimized Goertzel filter.

Usage
_ : goertzelOpt(freq,N) : _;
Where:

- freq: frequency to be analyzed
- N: the Goertzel block size

Reference


(an.)goertzelComp
Complex Goertzel filter.

Usage
_ : goertzelComp(freq,N) : _;
Where:

- freq: frequency to be analyzed
- N: the Goertzel block size

Reference


(an.)goertzel
Same as goertzelOpt.

Usage
_ : goertzel(freq,N) : _;
Where:

- freq: frequency to be analyzed
- N: the Goertzel block size
Reference


---

(an.)fft

Fast Fourier Transform (FFT)

Usage

```plaintext
si.cbus(N) : fft(N) : si.cbus(N);
```

Where:

- `si.cbus(N)` is a bus of `N` complex signals, each specified by real and imaginary parts: `(r0,i0), (r1,i1), (r2,i2), ...`
- `N` is the FFT size (must be a power of 2: 2, 4, 8, 16, ...)
- `fft(N)` performs a length `N` FFT for complex signals (radix 2)
- The output is a bank of `N` complex signals containing the complex spectrum over time: `(R0, I0), (R1, I1), ...`
  - The dc component is `(R0, I0)`, where `I0=0` for real input signals.

FFTs of Real Signals:

- To perform a sliding FFT over a real input signal, you can say

```plaintext
process = signal : an.rtocv(N) : an.fft(N);
```

where `an.rtocv` converts a real (scalar) signal to a complex vector signal having a zero imaginary part.

- See `an.rfft_analyzer_c` (in `analyzers.lib`) and related functions for more detailed usage examples.

- Use `an.rfft_spectral_level(N, tau, dB_offset)` to display the power spectrum of a real signal.

- See `dm.fft_spectral_level_demo(N)` in `demos.lib` for an example GUI driving `an.rfft_spectral_level()`.

Reference

- Decimation-in-time (DIT) Radix-2 FFT

---

(an.)ifft

Inverse Fast Fourier Transform (IFFT).
Usage

\texttt{si.cbus(N) : ifft(N) : si.cbus(N)};

Where:

- \texttt{N} is the IFFT size (power of 2)
- Input is a complex spectrum represented as interleaved real and imaginary parts: (R0, I0), (R1,I1), (R2,I2), \ldots
- Output is a bank of \texttt{N} complex signals giving the complex signal in the time domain: (r0, i0), (r1,i1), (r2,i2), \ldots

\texttt{basics.lib}

A library of basic elements. Its official prefix is \texttt{ba}.

\textbf{Conversion Tools}

\texttt{(ba.)samp2sec}

Converts a number of samples to a duration in seconds. \texttt{samp2sec} is a standard Faust function.

Usage

\texttt{samp2sec(n) : _}

Where:

- \texttt{n}: number of samples

\texttt{(ba.)sec2samp}

Converts a duration in seconds to a number of samples. \texttt{samp2sec} is a standard Faust function.

Usage

\texttt{sec2samp(d) : _}

Where:

- \texttt{d}: duration in seconds
(ba.)db2linear
Converts a loudness in dB to a linear gain (0-1). db2linear is a standard Faust function.

Usage
db2linear(l) : _
Where:
• l: loudness in dB

(ba.)linear2db
Converts a linear gain (0-1) to a loudness in dB. linear2db is a standard Faust function.

Usage
linear2db(g) : _
Where:
• g: a linear gain

(ba.)lin2LogGain
Converts a linear gain (0-1) to a log gain (0-1).

Usage
lin2LogGain(n) : _

(ba.)log2LinGain
Converts a log gain (0-1) to a linear gain (0-1).

Usage
log2LinGain(n) : _
(ba.)\texttt{tau2pole}

Returns a real pole giving exponential decay. Note that t60 (time to decay 60 dB) is ~6.91 time constants. \texttt{tau2pole} is a standard Faust function.

Usage

_ : smooth(tau2pole(tau)) : _

Where:

\begin{itemize}
  \item \texttt{tau}: time-constant in seconds
\end{itemize}

------------------------

(\textit{ba.})\texttt{pole2tau}

Returns the time-constant, in seconds, corresponding to the given real, positive pole in (0,1). \texttt{pole2tau} is a standard Faust function.

Usage

\texttt{pole2tau}(pole) : _

Where:

\begin{itemize}
  \item \texttt{pole}: the pole
\end{itemize}

------------------------

(\textit{ba.})\texttt{midikey2hz}

Converts a MIDI key number to a frequency in Hz (MIDI key 69 = A440). \texttt{midikey2hz} is a standard Faust function.

Usage

\texttt{midikey2hz}(mk) : _

Where:

\begin{itemize}
  \item \texttt{mk}: the MIDI key number
\end{itemize}

------------------------

(\textit{ba.})\texttt{hz2midikey}

Converts a frequency in Hz to a MIDI key number (MIDI key 69 = A440). \texttt{hz2midikey} is a standard Faust function.
Usage

hz2midkey(f) : _
Where:
  • f: frequency in Hz

(\textit{ba.}) \texttt{semi2ratio}

Converts semitones in a frequency multiplicative ratio. \texttt{semi2ratio} is a standard Faust function.

Usage

\texttt{semi2ratio(semi)} : _
Where:
  • semi: number of semitone

(\textit{ba.}) \texttt{ratio2semi}

Converts a frequency multiplicative ratio in semitones. \texttt{ratio2semi} is a standard Faust function.

Usage

\texttt{ratio2semi(ratio)} : _
Where:
  • ratio: frequency multiplicative ratio

(\textit{ba.}) \texttt{pianokey2hz}

Converts a piano key number to a frequency in Hz (piano key 49 = A440).

Usage

\texttt{pianokey2hz(pk)} : _
Where:
  • pk: the piano key number
(ba.)hz2pianokey
Converts a frequency in Hz to a piano key number (piano key 49 = A440).

Usage
hz2pianokey(f) : _
Where:
• f: frequency in Hz

Counters and Time/Tempo Tools
(ba.)countdown
Starts counting down from n included to 0. While trig is 1 the output is n. The
countdown starts with the transition of trig from 1 to 0. At the end of the
countdown the output value will remain at 0 until the next trig. countdown is a
standard Faust function.

Usage
countdown(n, trig) : _
Where:
• n: the starting point of the countdown
• trig: the trigger signal (1: start at n; 0: decrease until 0)

(ba.)countup
Starts counting up from 0 to n included. While trig is 1 the output is 0. The
countup starts with the transition of trig from 1 to 0. At the end of the countup
the output value will remain at n until the next trig. countup is a standard
Faust function.

Usage
countup(n, trig) : _
Where:
• n: the maximum count value
• trig: the trigger signal (1: start at 0; 0: increase until n)
(ba.)sweep
Counts from 0 to period-1 repeatedly, generating a sawtooth waveform, like os.lf_rawsaw, starting at 1 when run transitions from 0 to 1. Outputs zero while run is 0.

Usage
sweep(period,run) : _

(ba.)time
A simple timer that counts every samples from the beginning of the process. time is a standard Faust function.

Usage
time : _

(ba.)ramp
An linear ramp of ‘n’ samples to reach the next value

Usage
_ : ramp(n) : _
Where:
• n: number of samples to reach the next value

(ba.)tempo
Converts a tempo in BPM into a number of samples.

Usage
tempo(t) : _
Where:
• t: tempo in BPM
(ba.)period
Basic sawtooth wave of period p.

Usage
period(p) : _
Where:
  • p: period as a number of samples

(ba.)pulse
Pulses (10000) generated at period p.

Usage
pulse(p) : _
Where:
  • p: period as a number of samples

(ba.)pulsen
Pulses (11110000) of length n generated at period p.

Usage
pulsen(n,p) : _
Where:
  • n: pulse length as a number of samples
  • p: period as a number of samples

(ba.)cycle
Split nonzero input values into n cycles.

Usage
_ : cycle(n) <:
Where:
  • n: the number of cycles/output signals
(ba.)beat
Pulses at tempo t. beat is a standard Faust function.

Usage
beat(t) : _
Where:
  * t: tempo in BPM

(ba.)pulse_countup
Starts counting up pulses. While trig is 1 the output is counting up, while trig is 0 the counter is reset to 0.

Usage
_ : pulse_countup(trig) : _
Where:
  * trig: the trigger signal (1: start at next pulse; 0: reset to 0)

(ba.)pulse_countdown
Starts counting down pulses. While trig is 1 the output is counting down, while trig is 0 the counter is reset to 0.

Usage
_ : pulse_countdown(trig) : _
Where:
  * trig: the trigger signal (1: start at next pulse; 0: reset to 0)

(ba.)pulse_countup_loop
Starts counting up pulses from 0 to n included. While trig is 1 the output is counting up, while trig is 0 the counter is reset to 0. At the end of the countup (n) the output value will be reset to 0.
Usage
_ : pulse_countup_loop(n,trig) : _
Where:
- n: the highest number of the countup (included) before reset to 0.
- trig: the trigger signal (1: start at next pulse; 0: reset to 0)

(ba.)resetCtr
Function that lets through the mth impulse out of each consecutive group of n impulses.

Usage
_ : resetCtr(n,m) : _
Where:
- n: the total number of impulses being split
- m: index of impulse to allow to be output

(ba.)pulse_countdown_loop
Starts counting down pulses from 0 to n included. While trig is 1 the output is counting down, while trig is 0 the counter is reset to 0. At the end of the countdown (n) the output value will be reset to 0.

Usage
_ : pulse_countdown_loop(n,trig) : _
Where:
- n: the highest number of the countup (included) before reset to 0.
- trig: the trigger signal (1: start at next pulse; 0: reset to 0)

Array Processing/Pattern Matching
(ba.)count
Count the number of elements of list l. count is a standard Faust function.
Usage

count(l)
count((10,20,30,40)) -> 4

Where:
  - l: list of elements

(ba.)take

Take an element from a list. take is a standard Faust function.

Usage

take(P, l)
take(3, (10, 20, 30, 40)) -> 30

Where:
  - P: position (int, known at compile time, P > 0)
  - l: list of elements

(ba.)subseq

Extract a part of a list.

Usage

subseq(l, p, n)
subseq((10, 20, 30, 40, 50, 60), 1, 3) -> (20, 30, 40)
subseq((10, 20, 30, 40, 50, 60), 4, 1) -> 50

Where:
  - l: list
  - p: start point (0: begin of list)
  - n: number of elements

Note:

Faust doesn’t have proper lists. Lists are simulated with parallel compositions and there is no empty list.
Selectors (Conditions)

(ba.)if

if-then-else implemented with a select2. WARNING: since select2 is strict (always evaluating both branches), the resulting if does not have the usual “lazy” semantic of the C if form, and thus cannot be used to protect against forbidden computations like division-by-zero for instance.

Usage

- if(cond, then, else) : _

Where:
- cond: condition
- cond: signal selected while c is true
- else: signal selected while c is false

(ba.)selector

Selects the ith input among n at compile time.

Usage

selector(I,N)

- , , : selector(2,4) : _ // selects the 3rd input among 4

Where:
- I: input to select (int, numbered from 0, known at compile time)
- N: number of inputs (int, known at compile time, N > I)

There is also cselector for selecting among complex input signals of the form (real, imag).

(ba.)select2stereo

Select between 2 stereo signals.

Usage

- , , , : select2stereo(bpc) : ,

Where:
- bpc: the selector switch (0/1)
(ba.)selectn
Selects the ith input among N at run time.

Usage

```
selectn(N,i)
```

```
n_,n_,n_,n_ : selectn(4,2) : _ // selects the 3rd input among 4
```

Where:
- N: number of inputs (int, known at compile time, N > 0)
- i: input to select (int, numbered from 0)

Example test program

```
N = 64;
process = par(n, N, (par(i,N,i) : selectn(N,n)));
```

(ba.)selectmulti
Selects the ith circuit among N at run time (all should have the same number of inputs and outputs) with a crossfade.

Usage

```
selectmulti(n,lgen,id)
```

Where:
- n: crossfade in samples
- lgen: list of circuits
- id: circuit to select (int, numbered from 0)

Example test program

```
process = selectmulti(ma.SR/10, ((3,9),(2,8),(5,7)), nentry("choice", 0, 0, 2, 1));
process = selectmulti(ma.SR/10, ((*_3,_*9),(_*2,_*8),(_*5,_*7)), nentry("choice", 0, 0, 2, 1));
```

Other

(ba.)latch
Latch input on positive-going transition of “clock” (“sample-and-hold”).
**Usage**

_ : latch(clocksig) : _

Where:
- clocksig: hold trigger (0 for hold, 1 for bypass)

---

**(ba.)sAndH**

Sample And Hold. sAndH is a standard Faust function.

**Usage**

_ : sAndH(t) : _

Where:
- t: hold trigger (0 for hold, 1 for bypass)

---

**(ba.)downSample**

Down sample a signal. **WARNING**: this function doesn’t change the rate of a signal, it just holds samples... **downSample** is a standard Faust function.

**Usage**

_ : downSample(freq) : _

Where:
- freq: new rate in Hz

---

**(ba.)peakhold**

Outputs current max value above zero.

**Usage**

_ : peakhold(mode) : _;

Where:
- mode means: 0 - Pass through. A single sample 0 trigger will work as a reset. 1 - Track and hold max value.
(ba.) peakholder
Tracks abs peak and holds peak for ‘n’ samples.

Usage
_ : peakholder(n) : _;

Where:
• n: number of samples

(ba.) impulsify
Turns the signal from a button into an impulse (1,0,0,... when button turns on). impulsify is a standard Faust function.

Usage
button("gate") : impulsify;

(ba.) automat
Record and replay to the values the input signal in a loop.

Usage
hslider(...) : automat(bps, size, init) : _

(ba.) bpf
bpf is an environment (a group of related definitions) that can be used to create break-point functions. It contains three functions:
• start(x,y) to start a break-point function
• end(x,y) to end a break-point function
• point(x,y) to add intermediate points to a break-point function

A minimal break-point function must contain at least a start and an end point:
f = bpf.start(x0,y0) : bpf.end(x1,y1);

A more involved break-point function can contains any number of intermediate points:
f = bpf.start(x0,y0) : bpf.point(x1,y1) : bpf.point(x2,y2) : bpf.end(x3,y3);
In any case the \( x_{\{i\}} \) must be in increasing order (for all \( i, x_{\{i\}} < x_{\{i+1\}} \)). For example the following definition:

\[
f = \text{bpf.start}(x0,y0) : \ldots : \text{bpf.point}(xi,yi) : \ldots : \text{bpf.end}(xn,yn);\]

implements a break-point function \( f \) such that:

- \( f(x) = y_{0} \) when \( x < x_{0} \)
- \( f(x) = y_{n} \) when \( x > x_{n} \)
- \( f(x) = y_{i} + (y_{i+1}-y_{i})*(x-x_{i})/(x_{i+1}-x_{i}) \) when \( x_{i} \leqslant x < x_{i+1} \)

\( \text{bpf} \) is a standard Faust function.

\[\text{(ba.)listInterp}\]
Linearly interpolates between the elements of a list.

**Usage**

\[
\text{index} = 1.69; // \text{range is 0-4}
\text{process} = \text{listInterp}((800,400,350,450,325),\text{index});
\]

Where:

- \( \text{index} \): the index (float) to interpolate between the different values. The range of \( \text{index} \) depends on the size of the list.

\[\text{(ba.)bypass1}\]
Takes a mono input signal, route it to \( e \) and bypass it if \( \text{bpc} = 1 \). \( \text{bypass1} \) is a standard Faust function.

**Usage**

\[
_ : \text{bypass1}(\text{bpc},e) : _
\]

Where:

- \( \text{bpc} \): bypass switch (0/1)
- \( e \): a mono effect

\[\text{(ba.)bypass2}\]
Takes a stereo input signal, route it to \( e \) and bypass it if \( \text{bpc} = 1 \). \( \text{bypass2} \) is a standard Faust function.
Usage
\[ \ldots, \text{bypass2}(\text{bpc}, e) : \ldots, \]
Where:
- \text{bpc}: bypass switch (0/1)
- \text{e}: a stereo effect

\(\text{(ba.)bypass1to2}\)
Bypass switch for effect \(e\) having mono input signal and stereo output. Effect \(e\) is bypassed if \(\text{bpc} = 1\). \(\text{bypass1to2}\) is a standard Faust function.

Usage
\[ \ldots, \text{bypass1}(\text{bpc}, e) : \ldots, \]
Where:
- \text{bpc}: bypass switch (0/1)
- \text{e}: a mono-to-stereo effect

\(\text{(ba.)bypass\_fade}\)
Bypass an arbitrary \((N \times N)\) circuit with ‘\(n\)’ samples crossfade. Once bypassed the effect is replaced by \(\text{par}(i,N,\ldots)\). Bypassed circuits can be chained.

Usage
\[ \ldots, \text{bypass\_fade}(n, b, e) : \ldots \]
or
\[ \ldots, \text{bypass\_fade}(n, b, e) : \ldots \]
- \(n\): number of samples for the crossfade
- \(b\): bypass switch (0/1)
- \(e\): \(N \times N\) circuit

Examples
\[ \text{process} = \text{bypass\_fade}(\text{ma.SR}/10, \text{checkbox("bypass echo")}, \text{echo}); \]
\[ \text{process} = \text{bypass\_fade}(\text{ma.SR}/10, \text{checkbox("bypass reverb")}, \text{freeverb}); \]
(ba.) **toggle**
Triggered by the change of 0 to 1, it toggles the output value between 0 and 1.

**Usage**

_ : toggle : _

**Examples**

button("toggle") : toggle : vbargraph("output", 0, 1)
(an.amp_follower(0.1) > 0.01) : toggle : vbargraph("output", 0, 1) // takes audio input

---

(ba.) **on_and_off**
The first channel set the output to 1, the second channel to 0.

**Usage**

_ , _ : on_and_off : _

**Example**

button("on"), button("off") : on_and_off : vbargraph("output", 0, 1)

---

(ba.) **selectoutn**
Route input to the output among N at run time.

**Usage**

_ : selectoutn(N, i) : _, _, ..., N

Where:

- N: number of outputs (int, known at compile time, N > 0)
- i: output number to route to (int, numbered from 0) (i.e. slider)

**Example**

process = 1 : selectoutn(3, sel) : par(i, 3, vbargraph("v.bargraph %i", 0, 1));
sel = hslider("volume", 0, 0, 2, 1) : int;
Sliding Reduce

Provides various operations on the last N samples using a high order ‘slidingReduce(op,N,maxN,disabledVal,x)” fold-like function:

- **slidingSum(n):** the sliding sum of the last n input samples, CPU-light
- **slidingSump(n,maxn):** the sliding sum of the last n input samples, numerically stable “forever”
- **slidingMax(n,maxn):** the sliding max of the last n input samples
- **slidingMin(n,maxn):** the sliding min of the last n input samples
- **slidingMean(n):** the sliding mean of the last n input samples, CPU-light
- **slidingMeanp(n,maxn):** the sliding mean of the last n input samples, numerically stable “forever”
- **slidingRMS(n):** the sliding RMS of the last n input samples, CPU-light
- **slidingRMSp(n,maxn):** the sliding RMS of the last n input samples, numerically stable “forever”

Working Principle

If we want the maximum of the last 8 values, we can do that as:

```plaintext
simpleMax(x) =
( ( max(x@0,x@1),
    max(x@2,x@3)
  ) :max
 ),
( ( max(x@4,x@5),
    max(x@6,x@7)
  ) :max
 ) :max;
```

`max(x@2,x@3)` is the same as `max(x@0,x@1)@2` but the latter re-uses a value we already computed, so is more efficient. Using the same trick for values 4 trough 7, we can write:

```plaintext
efficientMax(x) =
( ( max(x@0,x@1),
    max(x@0,x@1)@2
  ) :max
 ),
( ( max(x@0,x@1),
    max(x@0,x@1)@2
  ) :max
 )
```
We can rewrite it recursively, so it becomes possible to get the maximum at have any number of values, as long as it’s a power of 2.

```plaintext
recursiveMax =
case {
  (1,x) => x;
  (N,x) => max(recursiveMax(N/2,x), recursiveMax(N/2,x)@2(N/2));
};
```

What if we want to look at a number of values that’s not a power of 2? For each value, we will have to decide whether to use it or not. If N is bigger than the index of the value, we use it, otherwise we replace it with (0-(ma.INFINITY)):

```plaintext
variableMax(N,x) =
max(
  max(
    (x@0 : useVal(0)),
    (x@1 : useVal(1))
  ):max,
  (x@2 : useVal(2)),
  (x@3 : useVal(3))
):max
),
max(
  (x@4 : useVal(4)),
  (x@5 : useVal(5))
):max,
  (x@6 : useVal(6)),
  (x@7 : useVal(7))
):max
)
with {
  useVal(i) = select2((N>=i) , (0-(ma.INFINITY)), _);
};
```

Now it becomes impossible to re-use any values. To fix that let’s first look at
how we’d implement it using recursiveMax, but with a fixed N that is not a
power of 2. For example, this is how you’d do it with N=3:

\[ \text{binaryMaxThree}(x) = \]
\[ ( \text{recursiveMax}(1,x)@0, \text{recursiveMax}(2,x)@1 ) : \text{max}; \]

N=6

\[ \text{binaryMaxSix}(x) = \]
\[ ( \text{recursiveMax}(2,x)@0, \text{recursiveMax}(4,x)@2 ) : \text{max}; \]

Note that \text{recursiveMax}(2,x) is used at a different delay than in
\text{binaryMaxThree}, since it represents 1 and 2, not 2 and 3. Each block
is delayed the combined size of the previous blocks.

N=7

\[ \text{binaryMaxSeven}(x) = \]
\[ ( ( \text{recursiveMax}(1,x)@0, \text{recursiveMax}(2,x)@1 ) : \text{max}, \\
( \text{recursiveMax}(4,x)@3 ) : \text{max}; \]

To make a variable version, we need to know which powers of two are used, and
at which delay time.

Then it becomes a matter of:

- lining up all the different block sizes in parallel: the first \text{par()} statement
- delaying each the appropriate amount: \text{sumOfPrevBlockSizes()}
- turning it on or off: \text{useVal()}
- getting the maximum of all of them: \text{combine()}

In Faust, we can only do that for a fixed maximum number of values: \text{maxN}

\[ \text{variableBinaryMaxN}(N, \text{maxN}, x) = \]
\[ \text{par}(i, \text{maxN} \text{NrBits}, \text{recursiveMax}(\text{pow2}(i), x)@\text{sumOfPrevBlockSizes}(N, \text{maxN}, i) : \text{useVal}(i) : \text{combine}(\text{maxN}) \text{ with } { } \]
\[ \text{sumOfPrevBlockSizes}(N, \text{maxN}, 0) = 0; \]
\[ \text{sumOfPrevBlockSizes}(N, \text{maxN}, i) = (\text{subseq}((\text{allBlockSizes}(N, \text{maxN})), 0, i):\_); \]
allBlockSizes(N,maxN) = par(i, maxNrBits, pow2(i) * isUsed(i) );
maxNrBits = int2nrOfBits(maxN);
// get the maximum of all blocks
combine(2) = max;
combine(N) = max(combine(N-1),_);
// Decide whether or not to use a certain value, based on N
useVal(i) = select2( isUsed(i), (0-(ma.INFINITY)),_);
isUsed(i) = take(i+1,(int2bin(N,maxN)))
};

(ba.)slidingReduce
Fold-like high order function. Apply a commutative binary operation <op> to the last <n> consecutive samples of a signal <x>. For example: slidingReduce(max,128,128,−(ma.INFINITY)) will compute the maximum of the last 128 samples. The output is updated each sample, unlike reduce, where the output is constant for the duration of a block.

Usage
_ : slidingReduce(op,N,maxN,disabledVal) : _

Where:
• N: the number of values to process
• maxN: the maximum number of values to process, needs to be a power of 2
• op: the operator. Needs to be a commutative one.
• disabledVal: the value to use when we want to ignore a value.

In other words, op(x,disabledVal) should equal to x. For example, +(x,0) equals x and min(x,ma.INFINITY) equals x. So if we want to calculate the sum, we need to give 0 as disabledVal, and if we want the minimum, we need to give ma.INFINITY as disabledVal.

(ba.)slidingSum
The sliding sum of the last n input samples.
It will eventually run into numerical trouble when there is a persistent dc component. If that matters in your application, use the more CPU-intensive (ba.)slidingSump.

Usage
_ : slidingSum(N) : _

Where:
• N: the number of values to process

(ba.)slidingSump
The sliding sum of the last n input samples.
It uses a lot more CPU then (ba.)slidingSum(n,maxn), but is numerically stable “forever” in return.

Usage
_ : slidingSump(N,maxN) : _

Where:
• N: the number of values to process
• maxN: the maximum number of values to process, needs to be a power of 2

(ba.)slidingMax
The sliding maximum of the last n input samples.

Usage
_ : slidingMax(N,maxN) : _

Where:
• N: the number of values to process
• maxN: the maximum number of values to process, needs to be a power of 2

(ba.)slidingMin
The sliding minimum of the last n input samples.

Usage
_ : slidingMin(N,maxN) : _

Where:
• N: the number of values to process
• maxN: the maximum number of values to process, needs to be a power of 2
(ba.)slidingMean
The sliding mean of the last n input samples.
It will eventually run into numerical trouble when there is a persistent dc component. If that matters in your application, use the more CPU-intensive (ba.)slidingRMSp.

Usage
_ : slidingMean(N,maxN) : _
Where:
• N: the number of values to process

_______________________________

(ba.)slidingMeanp
The sliding mean of the last n input samples.
It uses a lot more CPU then (ba.)slidingMean(n,maxn), but is numerically stable “forever” in return.

Usage
_ : slidingMeanp(N,maxN) : _
Where:
• N: the number of values to process
• maxN: the maximum number of values to process, needs to be a power of 2

_______________________________

(ba.)slidingRMS
The root mean square of the last n input samples.
It will eventually run into numerical trouble when there is a persistent dc component. If that matters in your application, use the more CPU-intensive (ba.)slidingRMSp.

(ba.)slidingRMSp
The root mean square of the last n input samples.
It uses a lot more CPU then (ba.)slidingRMS(n,maxn), but is numerically stable “forever” in return.
Usage

_ : slidingRMSp(N,maxN) : _

Where:

- $N$: the number of values to process
- $\text{maxN}$: the maximum number of values to process, needs to be a power of 2

compressors.lib

A library of compressor effects. Its official prefix is `co`.

Functions Reference

(co.)peak_compression_gain_mono

Mono dynamic range compressor gain computer. `peak_compression_gain_mono` is a standard Faust function

Usage

_ : peak_compression_gain_mono(strength,thresh,att,rel,knee,prePost) : _

Where:

- `strength`: strength of the compression ($0 = \text{no compression}, 1 \text{ means hard limiting}, >1 \text{ means over-compression}$)
- `thresh`: dB level threshold above which compression kicks in
- `att`: attack time = time constant (sec) when level & compression going up
- `rel`: release time = time constant (sec) coming out of compression
- `knee`: a gradual increase in gain reduction around the threshold: Below `thresh-(knee/2)` there is no gain reduction, above `thresh+(knee/2)` there is the same gain reduction as without a knee, and in between there is a gradual increase in gain reduction.
- `prePost`: places the level detector either at the input or after the gain computer; this turns it from a linear return-to-zero detector into a log domain return-to-threshold detector

(co.)peak_compression_gain_N_chan

N channel dynamic range compressor gain computer. `peak_compression_gain_N_chan` is a standard Faust function

Usage

si.bus(N) : peak_compression_gain_N_chan(strength,thresh,att,rel,knee,prePost,link,N) : si.bus
Where:

- **strength**: strength of the compression (0 = no compression, 1 means hard limiting, >1 means over-compression)
- **thresh**: dB level threshold above which compression kicks in
- **att**: attack time = time constant (sec) when level & compression going up
- **rel**: release time = time constant (sec) coming out of compression
- **knee**: a gradual increase in gain reduction around the threshold: Below thresh-(knee/2) there is no gain reduction, above thresh+(knee/2) there is the same gain reduction as without a knee, and in between there is a gradual increase in gain reduction.
- **prePost**: places the level detector either at the input or after the gain computer; this turns it from a linear return-to-zero detector into a log domain return-to-threshold detector
- **link**: the amount of linkage between the channels. 0 = each channel is independent, 1 = all channels have the same amount of gain reduction
- **N**: the number of channels of the compressor

(co.)FFcompressor_N_chan

feed forward N channel dynamic range compressor. **FFcompressor_N_chan** is a standard Faust function

**Usage**

\[
\text{si.bus}(N) : \text{FFcompressor}_{N\_chan}(\text{strength, thresh, att, rel, knee, prePost, link, meter}, N) : \text{si.bus}(N)
\]

Where:

- **strength**: strength of the compression (0 = no compression, 1 means hard limiting, >1 means over-compression)
- **thresh**: dB level threshold above which compression kicks in
- **att**: attack time = time constant (sec) when level & compression going up
- **rel**: release time = time constant (sec) coming out of compression
- **knee**: a gradual increase in gain reduction around the threshold: Below thresh-(knee/2) there is no gain reduction, above thresh+(knee/2) there is the same gain reduction as without a knee, and in between there is a gradual increase in gain reduction.
- **prePost**: places the level detector either at the input or after the gain computer; this turns it from a linear return-to-zero detector into a log domain return-to-threshold detector
- **link**: the amount of linkage between the channels. 0 = each channel is independent, 1 = all channels have the same amount of gain reduction
- **meter**: a gain reduction meter. It can be implemented like so: meter = <::(ba.linear2db:max(maxGR):meter_group((hbargraph("[1]\[unit:dB]\[tooltip: gain reduction in dB]"", maxGR, 0))):attach;
- **N**: the number of channels of the compressor
(co.)FBcompressor_N_chan

feed back N channel dynamic range compressor. FBcompressor_N_chan is a standard Faust function

Usage

si.bus(N) : FBcompressor_N_chan(strength,thresh,att,rel,knee,prePost,link,meter,N) : si.bus(N)

Where:

• strength: strength of the compression (0 = no compression, 1 means hard limiting, >1 means over-compression)
• thresh: dB level threshold above which compression kicks in
• att: attack time = time constant (sec) when level & compression going up
• rel: release time = time constant (sec) coming out of compression
• knee: a gradual increase in gain reduction around the threshold: Below thresh-(knee/2) there is no gain reduction, above thresh+(knee/2) there is the same gain reduction as without a knee, and in between there is a gradual increase in gain reduction.
• prePost: places the level detector either at the input or after the gain computer; this turns it from a linear return-to-zero detector into a log domain return-to-threshold detector
• link: the amount of linkage between the channels. 0 = each channel is independent, 1 = all channels have the same amount of gain reduction
• meter: a gain reduction meter. It can be implemented like so: meter = <::(ba.linear2db:max(maxGR):meter_group((hbargraph("\[1\]\[unit:dB\]\[tooltip:gain reduction in dB\], maxGR, 0))))):attach;
• N: the number of channels of the compressor

(co.)FFFBcompressor_N_chan

feed forward / feed back N channel dynamic range compressor. the feedback part has a much higher strength, so they end up sounding similar

FFFBcompressor_N_chan is a standard Faust function

Usage

si.bus(N) : FFFBcompressor_N_chan(strength,thresh,att,rel,knee,prePost,link,meter,FBFF,N) : si.bus(N)

Where:

• strength: strength of the compression (0 = no compression, 1 means hard limiting, >1 means over-compression)
• thresh: dB level threshold above which compression kicks in
• att: attack time = time constant (sec) when level & compression going up
• rel: release time = time constant (sec) coming out of compression
• knee: a gradual increase in gain reduction around the threshold: Below thresh-(knee/2) there is no gain reduction, above thresh+(knee/2) there
is the same gain reduction as without a knee, and in between there is a gradual increase in gain reduction.

- **prePost**: places the level detector either at the input or after the gain computer; this turns it from a linear return-to-zero detector into a log domain return-to-threshold detector.
- **link**: the amount of linkage between the channels. 0 = each channel is independent, 1 = all channels have the same amount of gain reduction.
- **FFFB**: fade between feed forward (0) and feed back (1) compression.
- **meter**: a gain reduction meter. It can be implemented like so:
  ```
  meter = <:(, (ba.linear2db:max(maxGR):meter_group((hbargraph("[1][unit:dB][tooltip:
  gain reduction in dB]", maxGR, 0))))):attach;
  ```
- **N**: the number of channels of the compressor.

(co.) **RMS_compression_gain_mono**

Mono RMS dynamic range compressor gain computer. **RMS_compression_gain_mono** is a standard Faust function.

**Usage**

```
_ : RMS_compression_gain_mono(strength,thresh,att,rel,knee,prePost) : _
```

Where:

- **strength**: strength of the compression (0 = no compression, 1 means hard limiting, >1 means over-compression)
- **thresh**: dB level threshold above which compression kicks in
- **att**: attack time = time constant (sec) when level & compression going up
- **rel**: release time = time constant (sec) coming out of compression
- **knee**: a gradual increase in gain reduction around the threshold: Below thresh-(knee/2) there is no gain reduction, above thresh+(knee/2) there is the same gain reduction as without a knee, and in between there is a gradual increase in gain reduction.
- **prePost**: places the level detector either at the input or after the gain computer; this turns it from a linear return-to-zero detector into a log domain return-to-threshold detector.

(co.) **RMS_compression_gain_N_chan**

RMS N channel dynamic range compressor gain computer. **RMS_compression_gain_N_chan** is a standard Faust function.

**Usage**

```
si.bus(N) : RMS_compression_gain_N_chan(strength,thresh,att,rel,knee,prePost,link,N) : si.bus(N)
```

Where:
- **strength**: strength of the compression (0 = no compression, 1 means hard limiting, >1 means over-compression)
- **thresh**: dB level threshold above which compression kicks in
- **att**: attack time = time constant (sec) when level & compression going up
- **rel**: release time = time constant (sec) coming out of compression
- **knee**: a gradual increase in gain reduction around the threshold: Below thresh-(knee/2) there is no gain reduction, above thresh+(knee/2) there is the same gain reduction as without a knee, and in between there is a gradual increase in gain reduction.
- **prePost**: places the level detector either at the input or after the gain computer; this turns it from a linear return-to-zero detector into a log domain return-to-threshold detector
- **link**: the amount of linkage between the channels. 0 = each channel is independent, 1 = all channels have the same amount of gain reduction
- **N**: the number of channels of the compressor

(RMS_FFFBcompressor_NChan)

RMS feed forward / feed back N channel dynamic range compressor. The feedback part has a much higher strength, so they end up sounding similar. RMS_FFFBcompressor_NChan is a standard Faust function

**Usage**

```
si.bus(N) : RMS_FFFBcompressor_NChan(strength,thresh,att,rel,knee,prePost,link,FBFF,meter,N)
```

Where:

- **strength**: strength of the compression (0 = no compression, 1 means hard limiting, >1 means over-compression)
- **thresh**: dB level threshold above which compression kicks in
- **att**: attack time = time constant (sec) when level & compression going up
- **rel**: release time = time constant (sec) coming out of compression
- **knee**: a gradual increase in gain reduction around the threshold: Below thresh-(knee/2) there is no gain reduction, above thresh+(knee/2) there is the same gain reduction as without a knee, and in between there is a gradual increase in gain reduction.
- **prePost**: places the level detector either at the input or after the gain computer; this turns it from a linear return-to-zero detector into a log domain return-to-threshold detector
- **link**: the amount of linkage between the channels. 0 = each channel is independent, 1 = all channels have the same amount of gain reduction
- **FFFB**: fade between feed forward (0) and feed back (1) compression.
- **meter**: a gain reduction meter. It can be implemented like so: meter = <:(ba.linear2db:max(maxGR):meter_group((hbargraph("[1][unit:dB][tooltip: gain reduction in dB]", maxGR, 0))):attach;
- **N**: the number of channels of the compressor

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**(co.)RMS_FBcompressor_peak_limiter_N_chan**

N channel RMS feed back compressor into peak limiter feeding back into the FB compressor. By combining them this way, they complement each other optimally: The RMS compressor doesn’t have to deal with the peaks, and the peak limiter get’s spared from the steady state signal. the feedback part has a much higher strength, so they end up sounding similar  

**RMS_FBcompressor_peak_limiter_N_chan** is a standard Faust function

**Usage**

```plaintext
si.bus(N) : RMS_FBcompressor_peak_limiter_N_chan(strength,thresh,threshLim,att,rel,knee,link,meter,N) : si.bus(N)
```

Where:

- **strength**: strength of the compression (0 = no compression, 1 means hard limiting, >1 means over-compression)
- **thresh**: dB level threshold above which compression kicks in
- **threshLim**: dB level threshold above which the brick wall limiter kicks in
- **att**: attack time = time constant (sec) when level & compression going up this is also used as the release time of the limiter
- **rel**: release time = time constant (sec) coming out of compression
- **knee**: a gradual increase in gain reduction around the threshold: Below thresh-(knee/2) there is no gain reduction, above thresh+(knee/2) there is the same gain reduction as without a knee, and in between there is a gradual increase in gain reduction. the limiter uses a knee half this size
- **link**: the amount of linkage between the channels. 0 = each channel is independent, 1 = all channels have the same amount of gain reduction
- **meter**: a gain reduction meter. It can be implemented like so: meter = <:<(ba.linear2db:max(maxGR):meter_group((hbargraph(“[1][unit:dB][tooltip:gain reduction in dB]”, maxGR, 0))))):attach;
- **N**: the number of channels of the compressor

**Backward compatibility section**

**Functions Reference**

**(co.)compressor_mono**

Mono dynamic range compressors. **compressor_mono** is a standard Faust function.

**Usage**

```plaintext
_ : compressor_mono(ratio,thresh,att,rel) : _
```

Where:

- **ratio**: compression ratio (1 = no compression, >1 means compression)
• **thresh**: dB level threshold above which compression kicks in (0 dB = max level)
• **att**: attack time = time constant (sec) when level & compression going up
• **rel**: release time = time constant (sec) coming out of compression

**References**
- Albert Graef’s “faust2pd”/examples/synth/compressor_.dsp
- More features: [https://github.com/magnetophon/faustCompressors](https://github.com/magnetophon/faustCompressors)

**(co.)compressor_stereo**
Stereo dynamic range compressors.

**Usage**

```
_,_ : compressor_stereo(ratio, thresh, att, rel) : _,_
```

Where:
- **ratio**: compression ratio (1 = no compression, >1 means compression)
- **thresh**: dB level threshold above which compression kicks in (0 dB = max level)
- **att**: attack time = time constant (sec) when level & compression going up
- **rel**: release time = time constant (sec) coming out of compression

**References**
- Albert Graef’s “faust2pd”/examples/synth/compressor_.dsp
- More features: [https://github.com/magnetophon/faustCompressors](https://github.com/magnetophon/faustCompressors)

**(co.)limiter_1176_R4_mono**
A limiter guards against hard-clipping. It can be implemented as a compressor having a high threshold (near the clipping level), fast attack and release, and high ratio. Since the ratio is so high, some knee smoothing is desirable (“soft limiting”). This example is intended to get you started using compressor_* as a limiter, so all parameters are hardwired to nominal values here. Ratios: 4 (moderate compression), 8 (severe compression), 12 (mild limiting), or 20
to 1 (hard limiting) Att: 20-800 MICROseconds (Note: scaled by ratio in the 1176) Rel: 50-1100 ms (Note: scaled by ratio in the 1176) Mike Shipley likes 4:1 (Grammy-winning mixer for Queen, Tom Petty, etc.) Faster attack gives “more bite” (e.g. on vocals) He hears a bright, clear eq effect as well (not implemented here). limiter_1176_R4_mono is a standard Faust function.

Usage

_{ } : limiter_1176_R4_mono : _;

Reference:

(co.)limiter_1176_R4_stereo

A limiter guards against hard-clipping. It can be implemented as a compressor having a high threshold (near the clipping level), fast attack and release, and high ratio. Since the ratio is so high, some knee smoothing is desirable (“soft limiting”). This example is intended to get you started using compressor_* as a limiter, so all parameters are hardwired to nominal values here. Ratios: 4 (moderate compression), 8 (severe compression), 12 (mild limiting), or 20 to 1 (hard limiting) Att: 20-800 MICROseconds (Note: scaled by ratio in the 1176) Rel: 50-1100 ms (Note: scaled by ratio in the 1176) Mike Shipley likes 4:1 (Grammy-winning mixer for Queen, Tom Petty, etc.) Faster attack gives “more bite” (e.g. on vocals) He hears a bright, clear eq effect as well (not implemented here)

Usage

_{ },_ : limiter_1176_R4_stereo : _,_,;

Reference:

delays.lib

This library contains a collection of delay functions. Its official prefix is de.
Basic Delay Functions

(de.)delay

Simple d samples delay where n is the maximum delay length as a number of samples. Unlike the `@` delay operator, here the delay signal d is explicitly bounded to the interval [0..n]. The consequence is that delay will compile even if the interval of d can’t be computed by the compiler. delay is a standard Faust function.

Usage

`_ : delay(n,d) : _`

Where:

- n: the max delay length (in samples)
- d: the delay length as a number of samples (integer)

(de.)fdelay

Simple d samples fractional delay based on 2 interpolated delay lines where n is the maximum delay length as a number of samples.

(de.)sdelay

s(mooth)delay: a mono delay that doesn’t click and doesn’t transpose when the delay time is changed.

Usage

`_ : sdelay(N,it,dt) : _`

Where:

- N: maximal delay in samples
- it: interpolation time (in samples) for example 1024
- dt: delay time (in samples)

Lagrange Interpolation

(de.)fdelaylti and (de.)fdelayltv

Fractional delay line using Lagrange interpolation.
Usage

_ : fdelaylt[i|v](order, maxdelay, delay, inputsignal) : _

Where order=1,2,3,... is the order of the Lagrange interpolation polynomial.

fdelaylti is most efficient, but designed for constant/slowly-varying delay.
fdelayltv is more expensive and more robust when the delay varies rapidly.

NOTE: The requested delay should not be less than (N-1)/2.

References

- https://ccrma.stanford.edu/~jos/pasp/Lagrange_Interpolation.html
  - (variable-delay case) https://ccrma.stanford.edu/~jos/Interpolation/Time_Varying_Lagrange_Interpolation.html

(de.)fdelay[n]

For convenience, fdelay1, fdelay2, fdelay3, fdelay4, fdelay5 are also available where n is the order of the interpolation.

Thiran Allpass Interpolation

Thiran Allpass Interpolation

Reference

https://ccrma.stanford.edu/~jos/pasp/Thiran_Allpass_Interpolators.html

(de.)fdelay[n]a

Delay lines interpolated using Thiran allpass interpolation.

Usage

_ : fdelay[N]a(maxdelay, delay, inputsignal) : _

(exactly like fdelay)

Where:
- $N=1, 2, 3, \text{ or } 4$ is the order of the Thiran interpolation filter, and the delay argument is at least $N - 1/2$.

**Note**

The interpolated delay should not be less than $N - 1/2$. (The allpass delay ranges from $N - 1/2$ to $N + 1/2$.) This constraint can be alleviated by altering the code, but be aware that allpass filters approach zero delay by means of pole-zero cancellations. The delay range $[N-1/2, N+1/2]$ is not optimal. What is?

Delay arguments too small will produce an UNSTABLE allpass!

Because allpass interpolation is recursive, it is not as robust as Lagrange interpolation under time-varying conditions. (You may hear clicks when changing the delay rapidly.)

First-order allpass interpolation, delay $d$ in $[0.5, 1.5]$.

---

demos.lib

This library contains a set of demo functions based on examples located in the `/examples` folder. Its official prefix is `dm`.

**Analyzers**

(dm.) mth_octave_spectral_level_demo

Demonstrate `mth_octave_spectral_level` in a standalone GUI.

**Usage**

```matlab
_ : mth_octave_spectral_level_demo(BandsPerOctave);
_ : spectral_level_demo : _; // 2/3 octave
```

---

**Filters**

(dm.) parametric_eq_demo

A parametric equalizer application.

**Usage:**

```matlab
_ : parametric_eq_demo : _;
```
(dm.)spectral_tilt_demo
A spectral tilt application.

Usage
_ : spectral_tilt_demo(N) : _ ;
Where:
  • N: filter order (integer)
All other parameters interactive

(dm.)mth_octave_filterbank_demo and (dm.)filterbank_demo
Graphic Equalizer: Each filter-bank output signal routes through a fader.

Usage
_ : mth_octave_filterbank_demo(M) : _
_ : filterbank_demo : _
Where:
  • N: number of bands per octave

Effects
(dm.)cubicnl_demo
Distortion demo application.

Usage:
_ : cubicnl_demo : _;

(dm.)gate_demo
Gate demo application.

Usage
_,_ : gate_demo : _,_ ;
(dm.)compressor_demo
Compressor demo application.

Usage
_ , _ : compressor_demo : _ , _ ;
________________________________________

(dm.)moog_vcf_demo
Illustrate and compare all three Moog VCF implementations above.

Usage
_ : moog_vcf_demo : _ ;
________________________________________

(dm.)wah4_demo
Wah pedal application.

Usage
_ : wah4_demo : _ ;
________________________________________

(dm.)crybaby_demo
Crybaby effect application.

Usage
_ : crybaby_demo : _ ;
________________________________________

(dm.)flanger_demo
Flanger effect application.

Usage
_ , _ : flanger_demo : _ , _ ;
________________________________________
(dm.)phasis2_demo
Phaser effect demo application.

Usage
_:_ : phaser2_demo : _:_;

(dm.)freeverb_demo
Freeverb demo application.

Usage
_:_ : freeverb_demo : _:_;

(dm.)stereo_reverb_tester
Handy test inputs for reverberator demos below.

Usage
_ : stereo_reverb_tester : _

(dm.)fdnrev0_demo
A reverb application using fdnrev0.

Usage
_:_ : fdnrev0_demo(N,NB,BBSO) : _:_

Where:

- n: Feedback Delay Network (FDN) order / number of delay lines used = order of feedback matrix / 2, 4, 8, or 16 [extend primes array below for 32, 64, ...]
- nb: Number of frequency bands / Number of (nearly) independent T60 controls / Integer 3 or greater
- bbso = Butterworth band-split order / order of lowpass/highpass bandsplit used at each crossover freq / odd positive integer

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(dm.)zita_rev_fdn_demo
Reverb demo application based on zita_rev_fdn.

Usage
si.bus(8) : zita_rev_fdn_demo : si.bus(8)

(dm.)zita_light
Light version of dm.zita_rev1 with only 2 UI elements.

Usage
_,_ : zita_light : _,_

(dm.)zita_rev1
Example GUI for zita_rev1_stereo (mostly following the Linux zita-rev1 GUI).

Only the dry/wet and output level parameters are “dezippered” here. If parameters are to be varied in real time, use smooth(0.999) or the like in the same way.

Usage
_,_ : zita_rev1 : _,_

Reference
http://www.kokkinizita.net/linuxaudio/zita-rev1-doc/quickguide.html

Generators
(dm.)sawtooth_demo
An application demonstrating the different sawtooth oscillators of Faust.

Usage
sawtooth_demo : _
(dm.)virtual_analog_oscillator_demo
Virtual analog oscillator demo application.

Usage
virtual_analog_oscillator_demo : _

______________________________

(dm.)oscrs_demo
Simple application demoing filter based oscillators.

Usage
oscrs_demo : _

______________________________

(dm.)velvet_noise_demo
Listen to velvet_noise!

Usage
velvet_noise_demo : _

______________________________

(dm.)latch_demo
Illustrate latch operation

Usage
echo 'import("stdfaust.lib");' > latch_demo.dsp
echo 'process = dm.latch_demo;' >> latch_demo.dsp
faust2octave latch_demo.dsp
Octave:1> plot(faustout);

______________________________

(dm.)envelopes_demo
Illustrate various envelopes overlaid, including their gate * 1.1.
Usage

echo 'import("stdfaust.lib");' > envelopes_demo.dsp
echo 'process = dm.envelopes_demo;' >> envelopes_demo.dsp
faust2octave envelopes_demo.dsp
Octave:1> plot(faustout);

(dm.)fft_spectral_level_demo

Make a real-time spectrum analyzer using FFT from analyzers.lib

Usage

echo 'import("stdfaust.lib");' > fft_spectral_level_demo.dsp
echo 'process = dm.fft_spectral_level_demo;' >> fft_spectral_level_demo.dsp
Mac:
  faust2caqt fft_spectral_level_demo.dsp
  open fft_spectral_level_demo.app
Linux GTK:
  faust2jack fft_spectral_level_demo.dsp
  ./fft_spectral_level_demo
Linux QT:
  faust2jaqt fft_spectral_level_demo.dsp
  ./fft_spectral_level_demo

(dm.)reverse_echo_demo(nChans)

Multichannel echo effect with reverse delays

Usage

echo 'import("stdfaust.lib");' > reverse_echo_demo.dsp
echo 'nChans = 3; // Any integer > 1 should work here' >> reverse_echo_demo.dsp
echo 'process = dm.reverse_echo_demo(nChans);' >> reverse_echo_demo.dsp
Mac:
  faust2caqt reverse_echo_demo.dsp
  open reverse_echo_demo.app
Linux GTK:
  faust2jack reverse_echo_demo.dsp
  ./reverse_echo_demo
Linux QT:
  faust2jaqt reverse_echo_demo.dsp
  ./reverse_echo_demo
Etc.
Use Positive-Pass Filter pospass() to frequency-shift a sine tone. First, a real sinusoid is converted to its analytic-signal form using pospass() to filter out its negative frequency component. Next, it is multiplied by a modulating complex sinusoid at the shifting frequency to create the frequency-shifted result. The real and imaginary parts are output to channels 1 & 2. For a more interesting frequency-shifting example, check the “Use Mic” checkbox to replace the input sinusoid by mic input. Note that frequency shifting is not the same as frequency scaling. A frequency-shifted harmonic signal is usually not harmonic. Very small frequency shifts give interesting chirp effects when there is feedback around the frequency shifter.

Usage

```bash
echo 'import("stdfaust.lib");' > pospass_demo.dsp
echo 'process = dm.pospass_demo;' >> pospass_demo.dsp
Mac:
  faust2caqt pospass_demo.dsp
  open pospass_demo.app
Linux GTK:
  faust2jack pospass_demo.dsp
  ./pospass_demo
Linux QT:
  faust2jaqt pospass_demo.dsp
  ./pospass_demo
Etc.
```

Psychoacoustic harmonic exciter, with GUI.

Usage

```bash
_ : exciter : _
```

References

- https://secure.aes.org/forum/pubs/ebriefs/?elib=16939
- https://www.researchgate.net/publication/25833577_Modeling_the_Harmonic_Exciter
Use example of the vocoder function where an impulse train is used as excitation.

Usage
_ : vocoder_demo : _;

-------------------------------

dx7.lib

Yamaha DX7 emulation library. Its official prefix is dx.

(dx.) dx7_ampf

DX7 amplitude conversion function. 3 versions of this function are available:

• dx7_amp_bpf: BPF version (same as in the CSOUND toolkit)
• dx7_amp_func: estimated mathematical equivalent of dx7_amp_bpf
• dx7_ampf: default (sugar for dx7_amp_func)

Usage:
dx7AmpPreset : dx7_ampf_bpf : _

Where:
• dx7AmpPreset: DX7 amplitude value (0-99)

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(dx.) dx7_egraterisef

DX7 envelope generator rise conversion function. 3 versions of this function are available:

• dx7_egraterise_bpf: BPF version (same as in the CSOUND toolkit)
• dx7_egraterise_func: estimated mathematical equivalent of dx7_egraterise_bpf
• dx7_egraterisef: default (sugar for dx7_egraterise_func)

Usage:
dx7envelopeRise : dx7_egraterisef : _

Where:
• dx7envelopeRise: DX7 envelope rise value (0-99)
(dx.)dx7_egraterisepercf

DX7 envelope generator percussive rise conversion function. 3 versions of this function are available:

- dx7_egrateriseperc_bpf: BPF version (same as in the CSOUND toolkit)
- dx7_egrateriseperc_func: estimated mathematical equivalent of dx7_egrateriseperc_bpf
- dx7_egraterisepercf: default (sugar for dx7_egrateriseperc_func)

Usage:

dx7envelopePercRise : dx7_egraterisepercf : _

Where:

- dx7envelopePercRise: DX7 envelope percussive rise value (0-99)

(dx.)dx7_egratedecayf

DX7 envelope generator decay conversion function. 3 versions of this function are available:

- dx7_egratedecay_bpf: BPF version (same as in the CSOUND toolkit)
- dx7_egratedecay_func: estimated mathematical equivalent of dx7_egratedecay_bpf
- dx7_egratedecayf: default (sugar for dx7_egratedecay_func)

Usage:

dx7envelopeDecay : dx7_egratedecayf : _

Where:

- dx7envelopeDecay: DX7 envelope decay value (0-99)

(dx.)dx7_egratedecaypercf

DX7 envelope generator percussive decay conversion function. 3 versions of this function are available:

- dx7_egratedecayperc_bpf: BPF version (same as in the CSOUND toolkit)
- dx7_egratedecayperc_func: estimated mathematical equivalent of dx7_egratedecayperc_bpf
- dx7_egratedecaypercf: default (sugar for dx7_egratedecayperc_func)
Usage:
dx7envelopePercDecay : dx7_egratedecaypercf : 
Where:
  • dx7envelopePercDecay: DX7 envelope decay value (0-99)

(dx.)dx7_eglv2peakf
DX7 envelope level to peak conversion function. 3 versions of this function are available:
  • dx7_eglv2peak_bpf: BPF version (same as in the CSOUND toolkit)
  • dx7_eglv2peak_func: estimated mathematical equivalent of dx7_eglv2peak_bpf
  • dx7_eglv2peakf: default (sugar for dx7_eglv2peak_func)

Usage:
dx7Level : dx7_eglv2peakf : 
Where:
  • dx7Level: DX7 level value (0-99)

(dx.)dx7_velsensf
DX7 velocity sensitivity conversion function.

Usage:
dx7Velocity : dx7_velsensf : 
Where:
  • dx7Velocity: DX7 level value (0-8)

(dx.)dx7_fdbkscalef
DX7 feedback scaling conversion function.

Usage:
dx7Feedback : dx7_fdbkscalef : 
Where:
  • dx7Feedback: DX7 feedback value
(dx.) dx7_op

DX7 Operator. Implements a phase-modulable sine wave oscillator connected to a DX7 envelope generator.

Usage:

\[ \text{dx7\_op}(\text{freq}, \text{phaseMod}, \text{outLev}, \text{R1}, \text{R2}, \text{R3}, \text{R4}, \text{L1}, \text{L2}, \text{L3}, \text{L4}, \text{keyVel}, \text{rateScale}, \text{type}, \text{gain}, \text{gate}) : \]

Where:

- **freq**: frequency of the oscillator
- **phaseMod**: phase deviation (-1 - 1)
- **outLev**: preset output level (0-99)
- **R1**: preset envelope rate 1 (0-99)
- **R2**: preset envelope rate 2 (0-99)
- **R3**: preset envelope rate 3 (0-99)
- **R4**: preset envelope rate 4 (0-99)
- **L1**: preset envelope level 1 (0-99)
- **L2**: preset envelope level 2 (0-99)
- **L3**: preset envelope level 3 (0-99)
- **L4**: preset envelope level 4 (0-99)
- **keyVel**: preset key velocity sensitivity (0-99)
- **rateScale**: preset envelope rate scale
- **type**: preset operator type
- **gain**: general gain
- **gate**: trigger signal

(dx.) dx7_algo

DX7 algorithms. Implements the 32 DX7 algorithms (a quick Google search should give you more details on this). Each algorithm uses 6 operators.

Usage:

\[ \text{dx7\_algo}(\text{algN}, \text{egR1}, \text{egR2}, \text{egR3}, \text{egR4}, \text{egL1}, \text{egL2}, \text{egL3}, \text{egL4}, \text{outLevel}, \text{keyVelSens}, \text{ampModSens}, \text{opMode}, \text{opFreq}, \text{opDetune}, \text{opRateScale}, \text{feedback}, \text{lfoDelay}, \text{lfoDepth}, \text{lfoSpeed}, \text{freq}, \text{gain}, \text{gate}) : \]

Where:

- **algN**: algorithm number (0-31, should be an int...)
- **egR1**: preset envelope rates 1 (a list of 6 values between 0-99)
- **egR2**: preset envelope rates 2 (a list of 6 values between 0-99)
- **egR3**: preset envelope rates 3 (a list of 6 values between 0-99)
- **egR4**: preset envelope rates 4 (a list of 6 values between 0-99)
- **egL1**: preset envelope levels 1 (a list of 6 values between 0-99)
- `egl2`: preset envelope levels 2 (a list of 6 values between 0-99)
- `egl3`: preset envelope levels 3 (a list of 6 values between 0-99)
- `egl4`: preset envelope levels 4 (a list of 6 values between 0-99)
- `outlev`: preset output levels (a list of 6 values between 0-99)
- `keyvel`: preset key velocity sensitivities (a list of 6 values between 0-99)
- `ampmodsen`: preset amplitude sensitivities (a list of 6 values between 0-99)
- `opmode`: preset operator mode (a list of 6 values between 0-1)
- `opfreq`: preset operator frequencies (a list of 6 values between 0-99)
- `opdetune`: preset operator detuning (a list of 6 values between 0-99)
- `opratescale`: preset operator rate scale (a list of 6 values between 0-99)
- `feedback`: preset operator feedback (a list of 6 values between 0-99)
- `lfoDelay`: preset LFO delay (a list of 6 values between 0-99)
- `lfoDepth`: preset LFO depth (a list of 6 values between 0-99)
- `lfoSpeed`: preset LFO speed (a list of 6 values between 0-99)
- `freq`: fundamental frequency
- `gain`: general gain
- `gate`: trigger signal

(dx.)dx7_ui
Generic DX7 function where all parameters are controllable using UI elements. The `master-with-mute` branch must be used for this function to work. This function is MIDI-compatible.

Usage
dx7_ui : _

---

envelopes.lib
This library contains a collection of envelope generators. Its official prefix is `en`.

Functions Reference
(en.) smoothEnvelope
An envelope with an exponential attack and release. `smoothEnvelope` is a standard Faust function.

Usage
smoothEnvelope(ar,t) : _
- `ar`: attack and release duration (s)
- t: trigger signal (attack is triggered when t>0, release is triggered when t=0)

_____

(en.) ar

AR (Attack, Release) envelope generator (useful to create percussion envelopes). ar is a standard Faust function.

Usage

ar(at, rt, t) : _

Where:
- at: attack (sec)
- rt: release (sec)
- t: trigger signal (attack is triggered when t>0, release is triggered when t=0)

_____

(en.) arfe


Usage

arfe(a, r, f, t) : _

Where:
- a, r: attack (sec), release (sec)
- f: final value to approach upon release (such as 0)
- t: trigger signal (attack is triggered when t>0, release is triggered when t=0)

_____

(en.) are


Usage

are(a, r, t) : _

Where:
• a: attack (sec)
• r: release (sec)
• t: trigger signal (attack is triggered when \( t>0 \), release is triggered when \( t=0 \))

(\textit{en.})asr

ASR (Attack, Sustain, Release) envelope generator. \texttt{asr} is a standard Faust function.

Usage

\texttt{asr(at,sl,rt,t)} : 

Where:

• \texttt{at}: attack (sec)
• \texttt{sl}: sustain level (between 0..1)
• \texttt{r}: release (sec)
• \texttt{t}: trigger signal (attack is triggered when \( t>0 \), release is triggered when \( t=0 \))

(\textit{en.})adsr

ADSR (Attack, Decay, Sustain, Release) envelope generator. \texttt{adsr} is a standard Faust function.

Usage

\texttt{adsr(at,dt,sl,rt,gate)} : 

Where:

• \texttt{at}: attack time (sec)
• \texttt{dt}: decay time (sec)
• \texttt{sl}: sustain level (between 0..1)
• \texttt{rt}: release time (sec)
• \texttt{gate}: trigger signal (attack is triggered when \texttt{gate}>0, release is triggered when \texttt{gate}=0)

(\textit{en.})adsre

ADSRE (Attack, Decay, Sustain, Release) envelope generator with Exponential segments.
Usage

adsre(a,d,s,r,g) : _

Where:
- a: attack (sec)
- d: decay (sec)
- s: sustain (fraction of t: 0-1)
- r: release (sec)
- t: trigger signal (attack is triggered when t>0, release is triggered when t=0)

(en.) asre

ASRE (Attack, Sustain, Release) envelope generator with Exponential segments.

Usage

asre(a,s,r,g) : _

Where:
- a: attack (sec)
- s: sustain (fraction of t: 0-1)
- r: release (sec)
- t: trigger signal (attack is triggered when t>0, release is triggered when t=0)

(en.) dx7envelope

DX7 operator envelope generator with 4 independent rates and levels. It is essentially a 4 points BPF.

Usage

dx7_envelope(R1,R2,R3,R4,L1,L2,L3,L4,t) : _

Where:
- RN: rates in seconds
- LN: levels (0-1)
- t: trigger signal
filters.lib

Faust Filters library; Its official prefix is f1.

The Filters library is organized into 18 sections:

- Basic Filters
- Comb Filters
- Direct-Form Digital Filter Sections
- Direct-Form Second-Order Biquad Sections
- Ladder/Lattice Digital Filters
- Useful Special Cases
- Ladder/Lattice Allpass Filters
- Digital Filter Sections Specified as Analog Filter Sections
- Simple Resonator Filters
- Butterworth Lowpass/Highpass Filters
- Special Filter-Bank Delay-Equalizing Allpass Filters
- Elliptic (Cauer) Lowpass Filters
- Elliptic Highpass Filters
- Butterworth Bandpass/Bandstop Filters
- Elliptic Bandpass Filters
- Parametric Equalizers (Shelf, Peaking)
- Mth-Octave Filter-Banks
- Arbitrary-Crossover Filter-Banks and Spectrum Analyzers

For more information, see ../documentation/library.pdf

Basic Filters

(f1.)zero

One zero filter. Difference equation: \( y(n) = x(n) - zx(n-1) \).

Usage

_ : zero(z) : _

Where:

- \( z \): location of zero along real axis in z-plane

Reference

https://ccrma.stanford.edu/~jos/filters/One_Zero.html

(f1.)pole

One pole filter. Could also be called a “leaky integrator”. Difference equation:

\( y(n) = x(n) + py(n-1) \).
Usage
_ : pole(p) : _

Where:
- \( p \): pole location = feedback coefficient

Reference
https://ccrma.stanford.edu/~jos/filters/One_Pole.html

____________________________

(fi.) integrator

Same as pole(1) [implemented separately for block-diagram clarity].

____________________________

(fi.) dcblockerat

DC blocker with configurable break frequency. The amplitude response is substantially flat above \( fb \), and sloped at about +6 dB/octave below \( fb \). Derived from the analog transfer function \( H(s) = \frac{s}{(s+2\pi//f_b)} \) by the low-frequency-matching bilinear transform method (i.e., the standard frequency-scaling constant \( 2*SR \)).

Usage
_ : dcblockerat(fb) : _

Where:
- \( fb \): “break frequency” in Hz, i.e., -3 dB gain frequency.

Reference

____________________________

(fi.) dcblocker

DC blocker. Default dc blocker has -3dB point near 35 Hz (at 44.1 kHz) and high-frequency gain near 1.0025 (due to no scaling). dcblocker is as standard Faust function.

Usage
_ : dcblocker : _

____________________________
Comb Filters

(ff.) ff_comb

Feed-Forward Comb Filter. Note that *ff_comb* requires integer delays (uses delay internally). *ff_comb* is a standard Faust function.

Usage

_ : ff_comb(maxdel,intdel,b0,bM) : _

Where:

- `maxdel`: maximum delay (a power of 2)
- `intdel`: current (integer) comb-filter delay between 0 and maxdel
- `del`: current (float) comb-filter delay between 0 and maxdel
- `b0`: gain applied to delay-line input
- `bM`: gain applied to delay-line output and then summed with input

Reference

https://ccrma.stanford.edu/~jos/pasp/Feedforward_Comb_Filters.html

(ff.) ff_fcomb

Feed-Forward Comb Filter. Note that *ff_fcomb* takes floating-point delays (uses fdelay internally). *ff_fcomb* is a standard Faust function.

Usage

_ : ff_fcomb(maxdel,del,b0,bM) : _

Where:

- `maxdel`: maximum delay (a power of 2)
- `intdel`: current (integer) comb-filter delay between 0 and maxdel
- `del`: current (float) comb-filter delay between 0 and maxdel
- `b0`: gain applied to delay-line input
- `bM`: gain applied to delay-line output and then summed with input

Reference

https://ccrma.stanford.edu/~jos/pasp/Feedforward_Comb_Filters.html
(fi.) **ffcombfilter**

Typical special case of `ff_comb()` where: $b0 = 1$.

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(ffi.) **fb_comb**

Feed-Back Comb Filter (integer delay).

**Usage**

```python
_: fb_comb(maxdel,intdel,b0,aN) :
```

Where:

- `maxdel`: maximum delay (a power of 2)
- `intdel`: current (integer) comb-filter delay between 0 and `maxdel`
- `del`: current (float) comb-filter delay between 0 and `maxdel`
- `b0`: gain applied to delay-line input and forwarded to output
- `aN`: minus the gain applied to delay-line output before summing with the input and feeding to the delay line

**Reference**

https://ccrma.stanford.edu/~jos/pasp/Feedback_Comb_Filters.html

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(ffi.) **fb_fcomb**

Feed-Back Comb Filter (floating point delay).

**Usage**

```python
_: fb_fcomb(maxdel,del,b0,aN) :
```

Where:

- `maxdel`: maximum delay (a power of 2)
- `intdel`: current (integer) comb-filter delay between 0 and `maxdel`
- `del`: current (float) comb-filter delay between 0 and `maxdel`
- `b0`: gain applied to delay-line input and forwarded to output
- `aN`: minus the gain applied to delay-line output before summing with the input and feeding to the delay line

**Reference**

https://ccrma.stanford.edu/~jos/pasp/Feedback_Comb_Filters.html

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Special case of \texttt{fb\_comb (rev1(maxdel,N,g))}. The “rev1 section” dates back to the 1960s in computer-music reverberation. See the \texttt{jcrev} and \texttt{brassrev} in \texttt{reverbs.lib} for usage examples.

\texttt{(fi.)fbcombfilter and (fi.)ffbcombfilter}

Other special cases of Feed-Back Comb Filter.

Usage \begin{verbatim}
_ : fbcombfilter(maxdel,intdel,g) :
_ : ffbcombfilter(maxdel,del,g) :
\end{verbatim}

Where:
- \texttt{maxdel}: maximum delay (a power of 2)
- \texttt{intdel}: current (integer) comb-filter delay between 0 and \texttt{maxdel}
- \texttt{del}: current (float) comb-filter delay between 0 and \texttt{maxdel}
- \texttt{g}: feedback gain

Reference
https://ccrma.stanford.edu/~jos/pasp/Feedback\_Comb\_Filters.html

\texttt{(fi.)allpass\_comb}

Schroeder Allpass Comb Filter. Note that

\texttt{allpass\_comb(maxlen,len,aN) = ff\_comb(maxlen,len,aN,1) : fb\_comb(maxlen,len-1,1,aN)};

which is a direct-form-1 implementation, requiring two delay lines. The implementation here is direct-form-2 requiring only one delay line.

Usage \begin{verbatim}
_ : allpass\_comb(maxdel,intdel,aN) :
\end{verbatim}

Where:
- \texttt{maxdel}: maximum delay (a power of 2)
- \texttt{intdel}: current (integer) comb-filter delay between 0 and \texttt{maxdel}
- \texttt{del}: current (float) comb-filter delay between 0 and \texttt{maxdel}
- \texttt{aN}: minus the feedback gain

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Schroeder Allpass Comb Filter. Note that

\[
\text{allpass\_comb} = \text{ff\_comb} : \text{fb\_comb};
\]

which is a direct-form-1 implementation, requiring two delay lines. The implementation here is direct-form-2 requiring only one delay line.

\text{allpass\_fcomb} is a standard Faust library.

Usage

\[
\begin{align*}
_ : &\text{allpass\_comb}(	ext{maxdel},\text{intdel},\text{aN}) : _ \\
_ : &\text{allpass\_fcomb}(	ext{maxdel},\text{del},\text{aN}) : _
\end{align*}
\]

Where:

- \text{maxdel}: maximum delay (a power of 2)
- \text{intdel}: current (float) comb-filter delay between 0 and maxdel
- \text{del}: current (float) comb-filter delay between 0 and maxdel
- aN: minus the feedback gain

Special case of \text{allpass\_comb} (rev2(maxlen,len,g)). The “rev2 section” dates back to the 1960s in computer-music reverberation. See the jcrev and brassrev in reverbs.lib for usage examples.

Same as \text{allpass\_fcomb} but use fdelay5 and fdelay1a internally (Interpolation helps - look at an fft of faust2octave on
Direct-Form Digital Filter Sections

(i.) iir

Nth-order Infinite-Impulse-Response (IIR) digital filter, implemented in terms of the Transfer-Function (TF) coefficients. Such filter structures are termed “direct form”.

iir is a standard Faust function.

Usage

_ : iir(bcoeffs,acoeffs) : _

Where:

- **order**: filter order (int) = max(#poles,#zeros)
- **bcoeffs**: (b0,b1,...,b_order) = TF numerator coefficients
- **acoeffs**: (a1,...,a_order) = TF denominator coeffs (a0=1)

Reference

https://ccrma.stanford.edu/~jos/filters/Four_Direct_Forms.html

(fii.) fir

FIR filter (convolution of FIR filter coefficients with a signal)

Usage

_ : fir(bv) : _

fir is standard Faust function.

Where:

- **bv** = b0,b1,..,bn is a parallel bank of coefficient signals.

Note

bv is processed using pattern-matching at compile time, so it must have this normal form (parallel signals).
Example
Smoothing white noise with a five-point moving average:

\[
\text{bv} = .2,.2,.2,.2,.2; \\
\text{process} = \text{noise} : \text{fir(bv)};
\]

Equivalent (note double parens):

\[
\text{process} = \text{noise} : \text{fir}((.2,.2,.2,.2,.2));
\]

---

(fi.)\text{conv} and (fi.)\text{convN}
Convolution of input signal with given coefficients.

Usage

\[
_ : \text{conv}((k1,k2,k3,\ldots,kN)) : _; \quad \text{// Argument = one signal bank} \\
_ : \text{convN}(N,(k1,k2,k3,\ldots)) : _; \quad \text{// Useful when } N < \text{count}((k1,\ldots))
\]

---

(fi.)\text{tf1}, (fi.)\text{tf2} and (fi.)\text{tf3}
\text{tfN} = N\text{'}th-order direct-form digital filter.

Usage

\[
_ : \text{tf1}(b0,b1,a1) : _ \\
_ : \text{tf2}(b0,b1,b2,a1,a2) : _ \\
_ : \text{tf3}(b0,b1,b2,b3,a1,a2,a3) : _
\]

Where:
- \(a\): the poles
- \(b\): the zeros

Reference

https://ccrma.stanford.edu/~jos/fp/Direct_Form_I.html

---

(fi.)\text{notchw}
Simple notch filter based on a biquad (\text{tf2}). \text{notchw} is a standard Faust function.
Usage:
\_ \_ : notchw(width,freq) : \_

Where:
- \textit{width}: “notch width” in Hz (approximate)
- \textit{freq}: “notch frequency” in Hz

Reference
https://ccrma.stanford.edu/~jos/pasp/Phasing_2nd_Order_Allpass_Filters.html

---

**Direct-Form Second-Order Biquad Sections**

Direct-Form Second-Order Biquad Sections

Reference
https://ccrma.stanford.edu/~jos/filters/Four_Direct_Forms.html

\((fi.) tf21, (fi.) tf22, (fi.) tf22t and (fi.) tf21t\)

\(tfN = N^{th}\)-order direct-form digital filter where:
- \textit{tf21} is \textit{tf2}, direct-form 1
- \textit{tf22} is \textit{tf2}, direct-form 2
- \textit{tf22t} is \textit{tf2}, direct-form 2 transposed
- \textit{tf21t} is \textit{tf2}, direct-form 1 transposed

Usage
\_ \_ : tf21(b0,b1,b2,a1,a2) : \_

\_ \_ : tf22(b0,b1,b2,a1,a2) : \_

\_ \_ : tf22t(b0,b1,b2,a1,a2) : \_

\_ \_ : tf21t(b0,b1,b2,a1,a2) : \_

Where:
- \textit{a}: the poles
- \textit{b}: the zeros

Reference
https://ccrma.stanford.edu/~jos/fp/Direct_Form_I.html
Ladder/Lattice Digital Filters

Ladder and lattice digital filters generally have superior numerical properties relative to direct-form digital filters. They can be derived from digital waveguide filters, which gives them a physical interpretation.

(fi.) av2sv

Compute reflection coefficients sv from transfer-function denominator av.

Usage
sv = av2sv(av)

Where:
• av: parallel signal bank a_1,...,a_N
• sv: parallel signal bank s_1,...,s_N

where r_0 = i_th reflection coefficient, and a_i = coefficient of z^(-i) in the filter transfer-function denominator A(z).

Reference
https://ccrma.stanford.edu/~jos/filters/Step_Down_Procedure.html (where reflection coefficients are denoted by k rather than s).

(fi.) bvav2nuv

Compute lattice tap coefficients from transfer-function coefficients.

Usage
nuv = bvav2nuv(bv,av)

Where:
• av: parallel signal bank a_1,...,a_N
• bv: parallel signal bank b_0,b_1,...,a_N
• nuv: parallel signal bank n_1,...,nu_N

where nui is the i_th tap coefficient, b_i is the coefficient of z^(-i) in the filter numerator, a_i is the coefficient of z^(-i) in the filter denominator.

(fi.) iir_lat2

Two-multiply lattice IIR filter of arbitrary order.
Usage

_ : iir_lat2(bv,av) : _

Where:

- bv: zeros as a bank of parallel signals
- av: poles as a bank of parallel signals

\text{(fi.)allpassnt}

Two-multiply lattice allpass (nested order-1 direct-form-ii allpasses).

Usage

_ : allpassnt(n,sv) : _

Where:

- n: the order of the filter
- sv: the reflection coefficients (-1 1)

\text{(fi.)iir_kl}

Kelly-Lochbaum ladder IIR filter of arbitrary order.

Usage

_ : iir_kl(bv,av) : _

Where:

- bv: zeros as a bank of parallel signals
- av: poles as a bank of parallel signals

\text{(fi.)allpassklt}

Kelly-Lochbaum ladder allpass.

Usage:

_ : allpassklt(n,sv) : _

Where:

- n: the order of the filter
- sv: the reflection coefficients (-1 1)
### iir_lat1

One-multiply lattice IIR filter of arbitrary order.

**Usage**

_ : iir_lat1(bv,av) : _

Where:

- bv: zeros as a bank of parallel signals
- av: poles as a bank of parallel signals

### allpassn1mt

One-multiply lattice allpass with tap lines.

**Usage**

_ : allpassn1mt(n,sv) : _

Where:

- n: the order of the filter
- sv: the reflection coefficients (-1 1)

### iir_nl

Normalized ladder filter of arbitrary order.

**Usage**

_ : iir_nl(bv,av) : _

Where:

- bv: zeros as a bank of parallel signals
- av: poles as a bank of parallel signals

**References**

- [https://ccrma.stanford.edu/~jos/pasp/Normalized_Scattering_Junctions.html](https://ccrma.stanford.edu/~jos/pasp/Normalized_Scattering_Junctions.html)
Normalized ladder allpass filter of arbitrary order.

Usage:
_ : allpassnnlt(n,sv) :

Where:
- n: the order of the filter
- sv: the reflection coefficients (-1,1)

References
- https://ccrma.stanford.edu/~jos/pasp/Normalized_Scattering_Junctions.html

Useful Special Cases

Biquad based on a stable second-order Normalized Ladder Filter (more robust to modulation than tf2 and protected against instability).

Usage
_ : tf2np(b0,b1,b2,a1,a2) :

Where:
- a: the poles
- b: the zeros

Second-order transformer-normalized digital waveguide resonator.

Usage
_ : wgr(f,r) :

Where:
- f: resonance frequency (Hz)
• r: loss factor for exponential decay (set to 1 to make a numerically stable oscillator)

References
• https://ccrma.stanford.edu/~jos/pasp/Power_Normalized_Waveguide_Filters.html
• https://ccrma.stanford.edu/~jos/pasp/Digital_Waveguide_Oscillator.html

(fi.)nlf2
Second order normalized digital waveguide resonator.

Usage
_ : nlf2(f,r) : _

Where:
• f: resonance frequency (Hz)
• r: loss factor for exponential decay (set to 1 to make a sinusoidal oscillator)

Reference
https://ccrma.stanford.edu/~jos/pasp/Power_Normalized_Waveguide_Filters.html

(fi.)apnl
Passive Nonlinear Allpass based on Pierce switching springs idea. Switch between allpass coefficient a1 and a2 at signal zero crossings.

Usage
_ : apnl(a1,a2) : _

Where:
• a1 and a2: allpass coefficients

Reference
Ladder/Lattice Allpass Filters

An allpass filter has gain 1 at every frequency, but variable phase. Ladder/lattice allpass filters are specified by reflection coefficients. They are defined here as nested allpass filters, hence the names allpassn*.

References

- https://ccrma.stanford.edu/~jos/pasp/Conventional_Ladder_Filters.html
- https://ccrma.stanford.edu/~jos/pasp/Nested_Allpass_Filters.html
- Linear Prediction of Speech, Markel and Gray, Springer Verlag, 1976

(allpassn)

Two-multiply lattice - each section is two multiply-adds.

Usage:

_ : allpassn(n,sv) : _

Where:

- n: the order of the filter
- sv: the reflection coefficients (-1 1)

References


(normalpassn)

Normalized form - four multiplies and two adds per section, but coefficients can be time varying and nonlinear without “parametric amplification” (modulation of signal energy).

Usage:

_ : allpassnn(n,tv) : _

Where:

- n: the order of the filter
- tv: the reflection coefficients (-PI PI)
((fl.)allpasskl
Kelly-Lochbaum form - four multiplies and two adds per section, but all signals have an immediate physical interpretation as traveling pressure waves, etc.

Usage:
_ : allpasskl(n,sv) : _

Where:
- n: the order of the filter
- sv: the reflection coefficients (-1 1)

((fl.)allpasslm
One-multiply form - one multiply and three adds per section. Normally the most efficient in special-purpose hardware.

Usage:
_ : allpasslm(n,sv) : _

Where:
- n: the order of the filter
- sv: the reflection coefficients (-1 1)

Digital Filter Sections Specified as Analog Filter Sections
((fl.)tf2s and (fl.)tf2snp
Second-order direct-form digital filter, specified by ANALOG transfer-function polynomials B(s)/A(s), and a frequency-scaling parameter. Digitization via the bilinear transform is built in.

Usage
_ : tf2s(b2,b1,b0,a1,a0,w1) : _

Where:
\[
H(s) = \frac{b2 s^2 + b1 s + b0}{s^2 + a1 s + a0}
\]

and \(w_1\) is the desired digital frequency (in radians/second) corresponding to analog frequency 1 rad/sec (i.e., \(s = j\)).
Example

A second-order ANALOG Butterworth lowpass filter, normalized to have cutoff frequency at 1 rad/sec, has transfer function

\[
H(s) = \frac{1}{s^2 + a_1 s + 1}
\]

where \( a_1 = \sqrt{2} \). Therefore, a DIGITAL Butterworth lowpass cutting off at \( SR/4 \) is specified as \( \text{tf2s}(0, 0, 1, \sqrt{2}, 1, \pi \cdot SR/2) \);

Method

Bilinear transform scaled for exact mapping of \( w_1 \).

Reference


\((fi.)\) tf3slf

Analogous to \( \text{tf2s} \) above, but third order, and using the typical low-frequency-matching bilinear-transform constant \( 2/T \) ("lf" series) instead of the specific-frequency-matching value used in \( \text{tf2s} \) and \( \text{tf1s} \). Note the lack of a "\( w_1 \)" argument.

Usage

\[
_ : \text{tf3slf}(b_3, b_2, b_1, b_0, a_3, a_2, a_1, a_0) : _
\]

\((fi.)\) tf1s

First-order direct-form digital filter, specified by ANALOG transfer-function polynomials \( B(s)/A(s) \), and a frequency-scaling parameter.

Usage

\[
\text{tf1s}(b_1, b_0, a_0, w_1)
\]

Where:

\[
\frac{b_1}{s + b_0}
\]

\( H(s) = \frac{s + a_0}{s + a_0} \)

and \( w_1 \) is the desired digital frequency (in radians/second) corresponding to analog frequency 1 rad/sec (i.e., \( s = j \)).
Example

A first-order ANALOG Butterworth lowpass filter, normalized to have cutoff frequency at 1 rad/sec, has transfer function

\[ H(s) = \frac{1}{s + 1} \]

so \( b_0 = a_0 = 1 \) and \( b_1 = 0 \). Therefore, a DIGITAL first-order Butterworth lowpass with gain -3dB at \( SR/4 \) is specified as

\[ \text{tf1s}(0,1,1,\pi*SR/2); // digital half-band order 1 Butterworth \]

Method

Bilinear transform scaled for exact mapping of \( w_1 \).

Reference


(fii.)tf2sb

Bandpass mapping of \( \text{tf2s} \): In addition to a frequency-scaling parameter \( w_1 \) (set to HALF the desired passband width in rad/sec), there is a desired center-frequency parameter \( w_c \) (also in rad/s). Thus, \( \text{tf2sb} \) implements a fourth-order digital bandpass filter section specified by the coefficients of a second-order analog lowpass prototype section. Such sections can be combined in series for higher orders. The order of mappings is (1) frequency scaling (to set lowpass cutoff \( w_1 \)), (2) bandpass mapping to \( w_c \), then (3) the bilinear transform, with the usual scale parameter \( 2*SR \). Algebra carried out in maxima and pasted here.

Usage

_ : \text{tf2sb}(b2,b1,b0,a1,a0,w1,w_c) : _

(fii.)tf1sb

First-to-second-order lowpass-to-bandpass section mapping, analogous to \( \text{tf2sb} \) above.

Usage

_ : \text{tf1sb}(b1,b0,a0,w1,w_c) : _
Simple Resonator Filters

(fi.) *resonlp*

Simple resonant lowpass filter based on *tf2s* (virtual analog). *resonlp* is a standard Faust function.

Usage

```plaintext
_ : resonlp(fc,Q,gain) : _
```

Where:

• `fc`: center frequency (Hz)
• `Q`: q
• `gain`: gain (0-1)

---

(fi.) *resonhp*

Simple resonant highpass filters based on *tf2s* (virtual analog). *resonhp* is a standard Faust function.

Usage

```plaintext
_ : resonlp(fc,Q,gain) : _
_ : resonhp(fc,Q,gain) : _
_ : resonbp(fc,Q,gain) : _
```

Where:

• `fc`: center frequency (Hz)
• `Q`: q
• `gain`: gain (0-1)

---

(fi.) *resonbp*

Simple resonant bandpass filters based on *tf2s* (virtual analog). *resonbp* is a standard Faust function.

Usage

```plaintext
_ : resonlp(fc,Q,gain) : _
_ : resonhp(fc,Q,gain) : _
_ : resonbp(fc,Q,gain) : _
```

Where:
• fc: center frequency (Hz)
• Q: q
• gain: gain (0-1)

Butterworth Lowpass/Highpass Filters

(lowpass)
Nth-order Butterworth lowpass filter. lowpass is a standard Faust function.

Usage
_ : lowpass(N,fc) : _

Where:
• N: filter order (number of poles) [nonnegative constant integer]
• fc: desired cut-off frequency (-3dB frequency) in Hz

References
• https://ccrma.stanford.edu/~jos/filters/Butterworth_Lowpass_Design.html
• butter function in Octave ("[z,p,g] = butter(N,1,'s');")

(highpass)
Nth-order Butterworth highpass filters. highpass is a standard Faust function.

Usage
_ : highpass(N,fc) : _

Where:
• N: filter order (number of poles) [nonnegative constant integer]
• fc: desired cut-off frequency (-3dB frequency) in Hz

References
• https://ccrma.stanford.edu/~jos/filters/Butterworth_Lowpass_Design.html
• butter function in Octave ("[z,p,g] = butter(N,1,'s');")
Special Filter-Bank Delay-Equalizing Allpass Filters

These special allpass filters are needed by filterbank et al. below. They are equivalent to \((\text{lowpass}(N,fc) -\text{highpass}(N,fc))/2\), but with canceling pole-zero pairs removed (which occurs for odd \(N\)).

Elliptic (Cauer) Lowpass Filters

Elliptic (Cauer) Lowpass Filters

References

- functions ncauer and ellip in Octave

Third-order Elliptic (Cauer) lowpass filter.

**Usage**

\[ _ : \text{lowpass3e}(fc) : _ \]

Where:

- \(fc\): -3dB frequency in Hz

**Design**

For spectral band-slice level display (see octave_analyzer3e):

\[
[z,p,g] = \text{ncauer}(Rp, Rs, 3); \quad \% \text{analog zeros, poles, and gain, where}
Rp = 60 \% \text{dB ripple in stopband}
Rs = 0.2 \% \text{dB ripple in passband}
\]

Sixth-order Elliptic/Cauer lowpass filter.

**Usage**

\[ _ : \text{lowpass6e}(fc) : _ \]

Where:

- \(fc\): -3dB frequency in Hz
Design
For spectral band-slice level display (see octave_analyzer6e):

\[ \{z, p, g\} = \text{ncauer}(Rp, Rs, 6); \]  % analog zeros, poles, and gain, where
\( Rp = 80 \) % dB ripple in stopband
\( Rs = 0.2 \) % dB ripple in passband

Elliptic Highpass Filters

(fi.)\text{highpass3e}
Third-order Elliptic (Cauer) highpass filter. Inversion of \text{lowpass3e} wrt unit circle in s plane \((s \leftarrow 1/s)\)

Usage
\_ : highpass3e(fc) : \_

Where:
\* fc: -3dB frequency in Hz

(fi.)\text{highpass6e}
Sixth-order Elliptic/Cauer highpass filter. Inversion of \text{lowpass3e} wrt unit circle in s plane \((s \leftarrow 1/s)\)

Usage
\_ : highpass6e(fc) : \_

Where:
\* fc: -3dB frequency in Hz

Butterworth Bandpass/Bandstop Filters

(fi.)\text{bandpass}
Order 2*Nh Butterworth bandpass filter made using the transformation \(s \leftarrow s + wc^2/s\) on \text{lowpass(Nh)}, where \(wc\) is the desired bandpass center frequency. The \text{lowpass(Nh)} cutoff \(wi\) is half the desired bandpass width. \text{bandpass} is a standard Faust function.
Usage

Usage: `bandpass(Nh,f1,fu)`

Where:

- `Nh`: HALF the desired bandpass order (which is therefore even)
- `f1`: lower -3dB frequency in Hz
- `fu`: upper -3dB frequency in Hz Thus, the passband width is `fu-f1`, and its center frequency is `(f1+fu)/2`.

Reference

http://cnx.org/content/m16913/latest/

---

(fi.) `bandstop`  
Order 2*Nh Butterworth bandstop filter made using the transformation `s <- s - wc^2/s` on `highpass(Nh)`, where `wc` is the desired bandpass center frequency. The `highpass(Nh)` cutoff `w1` is half the desired bandpass width. `bandstop` is a standard Faust function.

Usage

Usage: `bandstop(Nh,f1,fu)`

Where:

- `Nh`: HALF the desired bandstop order (which is therefore even)
- `f1`: lower -3dB frequency in Hz
- `fu`: upper -3dB frequency in Hz Thus, the passband (stopband) width is `fu-f1`, and its center frequency is `(f1+fu)/2`.

Reference

http://cnx.org/content/m16913/latest/

---

Elliptic Bandpass Filters

(fi.) `bandpass6e`  
Order 12 elliptic bandpass filter analogous to `bandpass(6)`.
Order 24 elliptic bandpass filter analogous to \texttt{bandpass(6)}. 

\texttt{pospass}

Positive-Pass Filter (single-side-band filter)

\textbf{Usage}

\texttt{\_ : pospass(N,fc) : \_,\_}

where

- \(N\): filter order (Butterworth bandpass for positive frequencies).
- \(fc\): lower bandpass cutoff frequency in Hz.
  - Highpass cutoff frequency at \(\text{ma.SR}/2 - fc\) Hz.

\textbf{Example}

- See \texttt{dm.pospass\_demo}
- Look at frequency response:

\textbf{Method}

A filter passing only positive frequencies can be made from a half-band lowpass by modulating it up to the positive-frequency range. Equivalently, down-modulate the input signal using a complex sinusoid at \(-\text{SR}/4\) Hz, lowpass it with a half-band filter, and modulate back up by \(\text{SR}/4\) Hz. In Faust/math notation: \(\text{pospass}(N) = \ast(e^{-j\pi/2n}) : \text{lowpass}(N,\text{SR}/4) : \ast(e^{j\pi/2n})\)

An approximation to the Hilbert transform is given by the imaginary output signal:

\(\text{hilbert}(N) = \text{pospass}(N) : !,\ast(2)\);

\textbf{References}

- https://ccrma.stanford.edu/~jos/sasp/Hilbert_Transform.html

\textbf{Parametric Equalizers (Shelf, Peaking)}

Parametric Equalizers (Shelf, Peaking).
References

- http://www.musicdsp.org/files/Audio-EQ-Cookbook.txt
- https://ccrma.stanford.edu/~jos/filters/Low_High_Shelving_Filters.html
- https://ccrma.stanford.edu/~jos/filters/Peaking_Equalizers.html
- maxmsp.lib in the Faust distribution
- bandfilter.dsp in the faust2pd distribution

**(fi.)low_shelf**

First-order “low shelf” filter (gain boost|cut between dc and some frequency)

`low_shelf` is a standard Faust function.

Usage

```plaintext
_ : lowshelf(N,L0,fx) :
_ : low_shelf(L0,fx) : _ // default case (order 3)
_ : lowshelf_other_freq(N,L0,fx) :
```

Where:

- **N**: filter order 1, 3, 5, ... (odd only). (default should be 3)
- **L0**: desired level (dB) between dc and fx (boost L0>0 or cut L0<0)
- **fx**: -3dB frequency of lowpass band (L0>0) or upper band (L0<0) (see “SHELF SHAPE” below).

The gain at SR/2 is constrained to be 1. The generalization to arbitrary odd orders is based on the well known fact that odd-order Butterworth band-splits are allpass-complementary (see filterbank documentation below for references).

Shelf Shape

The magnitude frequency response is approximately piecewise-linear on a log-log plot (“BODE PLOT”). The Bode “stick diagram” approximation L(If) is easy to state in dB versus dB-frequency If = dB(f):

- **L0 > 0:**
  - L(If) = L0, f between 0 and fx = 1st corner frequency;
  - L(If) = L0 * N * (If - Ifx), f between fx and f2 = 2nd corner frequency;
  - L(If) = 0, If > f2.
  - f2 = Ifx + L0/N = dB-frequency at which level gets back to 0 dB.
- **L0 < 0:**
  - L(If) = L0, f between 0 and If1 = 1st corner frequency;
  - L(If) = - N * (Ifx - If), f between f1 and Ifx = 2nd corner frequency;
  - L(If) = 0, If > Ifx.
  - If1 = Ifx + L0/N = dB-frequency at which level goes up from L0.

See `lowshelf_other_freq`. 
(fi.)high_shelf

First-order “high shelf” filter (gain boost|cut above some frequency). high_shelf is a standard Faust function.

Usage

  _ : highshelf(N,Lpi,fx) : _
  _ : high_shelf(L0,fx) : _  // default case (order 3)
  _ : highshelf_other_freq(N,Lpi,fx) : _

Where:

  • N: filter order 1, 3, 5, ... (odd only).
  • Lpi: desired level (dB) between fx and SR/2 (boost Lpi>0 or cut Lpi<0)
  • fx: -3dB frequency of highpass band (L0>0) or lower band (L0<0) (Use highshelf_other_freq() below to find the other one.)

The gain at dc is constrained to be 1. See lowshelf documentation above for more details on shelf shape.


(fii.)peak_eq

Second order “peaking equalizer” section (gain boost or cut near some frequency)
Also called a “parametric equalizer” section. peak_eq is a standard Faust function.

Usage

  _ : peak_eq(Lfx,fx,B) : _;

Where:

  • Lfx: level (dB) at fx (boost Lfx>0 or cut Lfx<0)
  • fx: peak frequency (Hz)
  • B: bandwidth (B) of peak in Hz


(fii.)peak_eq_cq

Constant-Q second order peaking equalizer section.

Usage

  _ : peak_eq_cq(Lfx,fx,Q) : _;

Where:

  • Lfx: level (dB) at fx
• fx: boost or cut frequency (Hz)
• Q: “Quality factor” = fx/B where B = bandwidth of peak in Hz

(\text{fi.})\text{peak\_eq\_rm}

Regalia-Mitra second order peaking equalizer section.

Usage
\_ : peak_eq_rm(Lfx,fx,tanPiBT) : \_

Where:
• Lfx: level (dB) at fx
• fx: boost or cut frequency (Hz)
• tanPiBT: \tan(\Pi B / \text{SR}), where B = -3dB bandwidth (Hz) when $10^{(Lfx/20)} = 0$ ~ $\Pi B / \text{SR}$ for narrow bandwidths B

Reference

(\text{fi.})\text{spectral\_tilt}

Spectral tilt filter, providing an arbitrary spectral rolloff factor alpha in (-1,1), where -1 corresponds to one pole (-6 dB per octave), and +1 corresponds to one zero (+6 dB per octave). In other words, alpha is the slope of the ln magnitude versus ln frequency. For a “pinking filter” (e.g., to generate 1/f noise from white noise), set alpha to -1/2.

Usage
\_ : spectral_tilt(N,f0,bw,alpha) : \_

Where:
• N: desired integer filter order (fixed at compile time)
• f0: lower frequency limit for desired roll-off band > 0
• bw: bandwidth of desired roll-off band
• alpha: slope of roll-off desired in nepers per neper, between -1 and 1 (ln mag / ln radian freq)

Examples
See spectral_tilt_demo.
Reference

(fi.)levelfilter
Dynamic level lowpass filter. levelfilter is a standard Faust function.

Usage
_ : levelfilter(L,freq) : _
Where:
• L: desired level (in dB) at Nyquist limit (SR/2), e.g., -60
• freq: corner frequency (-3dB point) usually set to fundamental freq
• N: Number of filters in series where L = L/N

Reference

(fi.)levelfilterN
Dynamic level lowpass filter.

Usage
_ : levelfilterN(N,freq,L) : _
Where:
• L: desired level (in dB) at Nyquist limit (SR/2), e.g., -60
• freq: corner frequency (-3dB point) usually set to fundamental freq
• N: Number of filters in series where L = L/N

Reference
Mth-Octave Filter-Banks

Mth-octave filter-banks split the input signal into a bank of parallel signals, one for each spectral band. They are related to the Mth-Octave Spectrum-Analyzers in analysis.lib. The documentation of this library contains more details about the implementation. The parameters are:

- $M$: number of band-slices per octave (>1)
- $N$: total number of bands (>2)
- $ftop$: upper bandlimit of the Mth-octave bands (<SR/2)

In addition to the Mth-octave output signals, there is a highpass signal containing frequencies from $ftop$ to $SR/2$, and a “d.c band” lowpass signal containing frequencies from 0 (d.c) up to the start of the Mth-octave bands. Thus, the $N$ output signals are

$\text{highpass}(ftop), \text{MthOctaveBands}(M,N-2,ftop), \text{dcBand}(ftop*2^{-M*(N-1)})$

A Filter-Bank is defined here as a signal bandsplitter having the property that summing its output signals gives an allpass-filtered version of the filter-bank input signal. A more conventional term for this is an “allpass-complementary filter bank”. If the allpass filter is a pure delay (and possible scaling), the filter bank is said to be a “perfect-reconstruction filter bank” (see Vaidyanathan-1993 cited below for details). A “graphic equalizer”, in which band signals are scaled by gains and summed, should be based on a filter bank.

The filter-banks below are implemented as Butterworth or Elliptic spectrum-analyzers followed by delay equalizers that make them allpass-complementary.

Increasing Channel Isolation

Go to higher filter orders - see Regalia et al. or Vaidyanathan (cited below) regarding the construction of more aggressive recursive filter-banks using elliptic or Chebysheev prototype filters.

References

- “Multirate Systems and Filter Banks”, P. Vaidyanathan, Prentice-Hall, 1993
- Elementary filter theory: https://ccrma.stanford.edu/~jos/filters/

(fi.)mth_octave_filterbank[n]

Allpass-complementary filter banks based on Butterworth band-splitting. For Butterworth band-splits, the needed delay equalizer is easily found.
Usage

_ : mth_octave_filterbank(O,M,ftop,N) : par(i,N,_) ; // 0th-order
_ : mth_octave_filterbank_alt(O,M,ftop,N) : par(i,N,_) ; // dc-inverted version

Also for convenience:

_ : mth_octave_filterbank3(M,ftop,N) : par(i,N,_) ; // 3rd-order Butterworth
_ : mth_octave_filterbank5(M,ftop,N) : par(i,N,_) ; // 5th-order Butterworth

mth_octave_filterbank_default = mth_octave_filterbank5;

Where:

• O: order of filter used to split each frequency band into two
• M: number of band-slices per octave
• ftop: highest band-split crossover frequency (e.g., 20 kHz)
• N: total number of bands (including dc and Nyquist)

Arbitrary-Crossover Filter-Banks and Spectrum Analyzers

These are similar to the Mth-octave analyzers above, except that the band-split frequencies are passed explicitly as arguments.

(fi.)filterbank

Filter bank. filterbank is a standard Faust function.

Usage

_ : filterbank (O,freqs) : par(i,N,_) ; // Butterworth band-splits

Where:

• O: band-split filter order (ODD integer required for filterbank[i])
• freqs: (fc1,fc2,...,fcNs) [in numerically ascending order], where Ns=N-1 is the number of octave band-splits (total number of bands N=Ns+1).

If frequencies are listed explicitly as arguments, enclose them in parens:

_ : filterbank(3,(fc1,fc2)) : _,_,_

(fi.)filterbanki

Inverted-dc filter bank.
Usage

_ : filterbanki(0,freqs) : par(i,N,_); // Inverted-dc version

Where:

- **O**: band-split filter order (ODD integer required for \texttt{filterbank}[i])
- **freqs**: \((fc1,fc2,...,fcNs)\) [in numerically ascending order], where \(Ns=N-1\) is the number of octave band-splits (total number of bands \(N=Ns+1\)).

If frequencies are listed explicitly as arguments, enclose them in parens:

_ : filterbanki(3,(fc1,fc2)) : _,_,_ 

\[\]

\textbf{hoa.lib}

Faust library for high order ambisonic. Its official prefix is \texttt{ho}.

\textbf{(ho.)encoder}

Ambisonic encoder. Encodes a signal in the circular harmonics domain depending on an order of decomposition and an angle.

Usage

encoder(n, x, a) : _

Where:

- **n**: the order
- **x**: the signal
- **a**: the angle

\[\]

\textbf{(ho.)decoder}

Decodes an ambisonics sound field for a circular array of loudspeakers.

Usage

_ : decoder(n, p) : _

Where:

- **n**: the order
- **p**: the number of speakers

\[\]
Note

Number of loudspeakers must be greater or equal to $2n+1$. It’s preferable to use $2n+2$ loudspeakers.

\[ \text{(ho.)} \text{decoderStereo} \]

Decodes an ambisonic sound field for stereophonic configuration. An “home made” ambisonic decoder for stereophonic restitution $(30^\circ - 330^\circ)$: Sound field lose energy around $180^\circ$. You should use \text{inPhase} optimization with punctual sources. 

\[ \text{### Usage} \]

_ : decoderStereo(n) : _

Where:

- n: the order

\[ \text{Optimization Functions} \]

Functions to weight the circular harmonics signals depending to the ambisonics optimization. It can be \text{basic} for no optimization, \text{maxRe} or \text{inPhase}.

\[ \text{(ho.)} \text{optimBasic} \]

The basic optimization has no effect and should be used for a perfect circle of loudspeakers with one listener at the perfect center loudspeakers array.

\[ \text{Usage} \]

_ : optimBasic(n) : _

Where:

- n: the order

\[ \text{(ho.)} \text{optimMaxRe} \]

The maxRe optimization optimize energy vector. It should be used for an auditory confined in the center of the loudspeakers array.

\[ \text{Usage} \]

_ : optimMaxRe(n) : _

Where:
• n: the order

(ho.)optimInPhase
The inPhase Optimization optimize energy vector and put all loudspeakers signals n phase. It should be used for an auditory.

Usage
```
" optimInPhase(n) : _ "
```
here:
n: the order

(ho.)wider
Can be used to wide the diffusion of a localized sound. The order depending signals are weighted and appear in a logarithmic way to have linear changes.

Usage
```
_ : wider(n,w) : _
```
Where:
• n: the order
• w: the width value between 0 - 1

(ho.)map
It simulate the distance of the source by applying a gain on the signal and a wider processing on the soundfield.

Usage
```
map(n, x, r, a)
```
Where:
• n: the order
• x: the signal
• r: the radius
• a: the angle in radian
(ho.)*rotate*
Rotates the sound field.

**Usage**

```markdown
_ : rotate(n, a) :
```

Where:

- `n`: the order
- `a`: the angle in radian

### 3D functions

```
//———————————————————-//
```

(ho.)*encoder3D*

Ambisonic encoder. Encodes a signal in the circular harmonics domain depending on an order of decomposition, an angle and an elevation.

**Usage**

```markdown
encoder3D(n, x, a, e) :
```

Where:

- `n`: the order
- `x`: the signal
- `a`: the angle
- `e`: the elevation

(ho.)*optimBasic3D*

The basic optimization has no effect and should be used for a perfect sphere of loudspeakers with one listener at the perfect center loudspeakers array.

**Usage**

```markdown
_ : optimBasic3D(n) :
```

Where:

- `n`: the order
(ho.)optimMaxRe3D

The maxRe optimization optimize energy vector. It should be used for an auditory confined in the center of the loudspeakers array.

Usage

_ : optimMaxRe3D(n) : _

Where:

• n: the order

______________________________

(ho.)optimInPhase3D

The inPhase Optimization optimize energy vector and put all loudspeakers signals n phase. It should be used for an auditory.

Usage

" optimInPhase3D(n) : _ "

here:

n: the order

______________________________

interpolators.lib

A library to handle interpolation in Faust. Its official prefix is it.

(it.)interpolate_linear

Linear interpolation between 2 values

Usage

interpolate_linear(dv,v0,v1) : _

Where:

• dv: in the fractional value in [0..1] range
• v0: is the first value
• v1: is the second value
**Reference:**

https://github.com/jamoma/JamomaCore/blob/master/Foundation/library/includes/TTInterpolate.h

(\textit{it.})\texttt{interpolate\_cosine}

Cosine interpolation between 2 values

**Usage**

\texttt{interpolate\_cosine(dv,v0,v1)} : _

Where:

- \texttt{dv}: in the fractional value in [0..1] range
- \texttt{v0}: is the first value
- \texttt{v1}: is the second value

**Reference:**

https://github.com/jamoma/JamomaCore/blob/master/Foundation/library/includes/TTInterpolate.h

(\textit{it.})\texttt{interpolate\_cubic}

Cubic interpolation between 4 values

**Usage**

\texttt{interpolate\_cubic(dv,v0,v1,v2,v3)} : _

Where:

- \texttt{dv}: in the fractional value in [0..1] range
- \texttt{v0}: is the first value
- \texttt{v1}: is the second value
- \texttt{v2}: is the third value
- \texttt{v3}: is the fourth value

**Reference:**

https://www.paulinternet.nl/?page=bicubic
(it.) interpolator_linear

Linear interpolator for a ‘gen’ circuit triggered by an ‘idv’ input to generate values

Usage

interpolator_linear(gen, idv) : _,_,... (equal to N = outputs(gen))

Where:

• gen: a circuit with an ‘idv’ reader input that produces N outputs
• idv: a fractional read index expressed as a float value, or a (int,frac) pair
  (see float.lib and double.lib)

_____________________

(it.) interpolator_cosine

Cosine interpolator for a ‘gen’ circuit triggered by an ‘idv’ input to generate values

Usage

interpolator_cosine(gen, idv) : _,_,... (equal to N = outputs(gen))

Where:

• gen: a circuit with an ‘idv’ reader input that produces N outputs
• idv: a fractional read index expressed as a float value, or a (int,frac) pair
  (see float.lib and double.lib)

_____________________

(it.) interpolator_cubic

Cubic interpolator for a ‘gen’ circuit triggered by an ‘idv’ input to generate values

Usage

interpolator_cubic(gen, idv) : _,_,... (equal to N = outputs(gen))

Where:

• gen: a circuit with an ‘idv’ reader input that produces N outputs
• idv: a fractional read index expressed as a float value, or a (int,frac) pair
  (see float.lib and double.lib)
(it.) interpolator_select

Generic configurable interpolator (with selector between in [0..3]). The value 3 is used for no interpolation.

Usage

interpolator_select(gen, idv, sel) : _,_,... (equal to N = outputs(gen))

Where:

- `gen`: a circuit with an `idv` reader input that produces N outputs
- `idv`: a fractional read index expressed as a float value, or a (int,frac) pair (see float.lib and double.lib)
- `sel`: an interpolation algorithm selector in [0..3] (0 = linear, 1 = cosine, 2 = cubic, 3 = nointerp)

--------------------

maths.lib

Mathematic library for Faust. Its official prefix is ma.

Functions Reference

(ma.)SR

Current sampling rate. Constant during program execution.

Usage

SR : _

--------------------

(ma.)BS

Current block-size. Can change during the execution.

Usage

BS : _

--------------------

(ma.)PI

Constant PI in double precision.
Usage

PI : _

(ma.) EPSILON
Constant EPSILON in simple/double/quad precision.

Usage

EPSILON : _

(ma.) MIN
Constant MIN in simple/double/quad precision (minimal positive value).

Usage

MIN : _

(ma.) INFINITY
Constant INFINITY in simple/double/quad precision (maximal positive value).

Usage

INFINITY : _

(ma.) FTZ
Flush to zero: force samples under the “maximum subnormal number” to be zero. Usually not needed in C++ because the architecture file take care of this, but can be useful in JavaScript for instance.

Usage

_ : ftz : _

See : http://docs.oracle.com/cd/E19957-01/806-3568/ncg_math.html
(ma.)neg
Invert the sign (-x) of a signal.

Usage
_ : neg : _

________________________

(ma.)sub(x,y)
Subtract x and y.

________________________

(ma.)inv
Compute the inverse (1/x) of the input signal.

Usage
_ : inv : _

________________________

(ma.)cbrt
Computes the cube root of of the input signal.

Usage
_ : cbrt : _

________________________

(ma.)hypot
Computes the euclidian distance of the two input signals sqrt(xx+yy) without undue overflow or underflow.

Usage
_,_ : hypot : _

________________________

(ma.)ldexp
 Takes two input signals: x and n, and multiplies x by 2 to the power n.
Usage

\_\_ : ldexp : _

\(\text{(ma.)scalb}\)
Takes two input signals: x and n, and multiplies x by 2 to the power n.

Usage

\_\_ : scalb : _

\(\text{(ma.)log1p}\)
Computes \(\log(1 + x)\) without undue loss of accuracy when x is nearly zero.

Usage

\_ : log1p : _

\(\text{(ma.)logb}\)
Return exponent of the input signal as a floating-point number.

Usage

\_ : logb : _

\(\text{(ma.)ilogb}\)
Return exponent of the input signal as an integer number.

Usage

\_ : ilogb : _

\(\text{(ma.)log2}\)
Returns the base 2 logarithm of x.
Usage
_ : log2 : _

(ma.) expm1
Return exponent of the input signal minus 1 with better precision.

Usage
_ : expm1 : _

(ma.) acosh
Computes the principle value of the inverse hyperbolic cosine of the input signal.

Usage
_ : acosh : _

(ma.) asinh
Computes the inverse hyperbolic sine of the input signal.

Usage
_ : asinh : _

(ma.) atanh
Computes the inverse hyperbolic tangent of the input signal.

Usage
_ : atanh : _

(ma.) sinh
Computes the hyperbolic sine of the input signal.
Usage
_ : sinh : _

(ma.)cosh
Computes the hyperbolic cosine of the input signal.

Usage
_ : cosh : _

(ma.)tanh
Computes the hyperbolic tangent of the input signal.

Usage
_ : tanh : _

(ma.)erf
Computes the error function of the input signal.

Usage
_ : erf : _

(ma.)erfc
Computes the complementary error function of the input signal.

Usage
_ : erfc : _

(ma.)gamma
Computes the gamma function of the input signal.
Usage
_ : gamma : _

(ma.)lgamma
Calculates the natural logarithm of the absolute value of the gamma function of the input signal.

Usage
_ : lgamma : _

(ma.)J0
Computes the Bessel function of the first kind of order 0 of the input signal.

Usage
_ : J0 : _

(ma.)J1
Computes the Bessel function of the first kind of order 1 of the input signal.

Usage
_ : J1 : _

(ma.)Jn
Computes the Bessel function of the first kind of order n (first input signal) of the second input signal.

Usage
_,_ : Jn : _

(ma.)Y0
Computes the linearly independent Bessel function of the second kind of order 0 of the input signal.
Usage
_ : Y0 : _

(ma.)Y1
Computes the linearly independent Bessel function of the second kind of order 1 of the input signal.

Usage
_ : Y0 : _

(ma.)Yn
Computes the linearly independent Bessel function of the second kind of order n (first input signal) of the second input signal.

Usage
_,_ : Yn : _

(ma.)fabs, (ma.)fmax, (ma.)fmin
Just for compatibility...
fabs = abs
fmax = max
fmin = min

(ma.)np2
Gives the next power of 2 of x.

Usage
np2(n) : _
Where:
• n: an integer
(ma.)frac
Gives the fractional part of n.

Usage
frac(n) : _
Where:
  • n: a decimal number

(modulo)
Modulus operation.

Usage
modulo(x,N) : _
Where:
  • x: the numerator
  • N: the denominator

(isnan)
Return non-zero if x is a NaN.

Usage
isnan(x)
_ : isnan : _
Where:
  • x: signal to analyse

(isinf)
Return non-zero if x is a positive or negative infinity.
Usage
isinf(x)
_ : isinf : _
Where:
• x: signal to analyse

(ma.)chebychev
Chebychev transformation of order n.

Usage
_ : chebychev(n) : _
Where:
• n: the order of the polynomial

Semantics
T[0](x) = 1,
T[1](x) = x,
T[n](x) = 2x*T[n-1](x) - T[n-2](x)

Reference
http://en.wikipedia.org/wiki/Chebyshev_polynomial

(ma.)chebychevpoly
Linear combination of the first Chebyshev polynomials.

Usage
_ : chebychevpoly((c0,c1,...,cn)) : _
Where:
• cn: the different Chebychev polynomials such that: chebychevpoly((c0,c1,...,cn)) = Sum of chebychev(i)*ci

Reference
Negated first-order difference.

Usage

_ : diffn : _

The signum function signum(x) is defined as -1 for x<0, 0 for x==0, and 1 for x>0.

Usage

_ : signum : _

The nextpow2(x) returns the lowest integer m such that \(2^m \geq x\).

Usage

\(2^{\text{nextpow2}(n)}\)

Useful for allocating delay lines, e.g.,

delay(\(2^{\text{nextpow2}(\text{maxDelayNeeded})}\), \text{currentDelay});

This library contains a collection of audio effects. Its official prefix is ef.

Cubic nonlinearity distortion. cubicnl is a standard Faust library.

Usage:

_ : cubicnl(drive,offset) : _

_ : cubicnl_nodc(drive,offset) : _

Where:
• drive: distortion amount, between 0 and 1
• offset: constant added before nonlinearity to give even harmonics. Note: offset can introduce a nonzero mean - feed cubicnl output to dcblocker to remove this.

References:
• https://ccrma.stanford.edu/~jos/pasp/Cubic_Soft_Clipper.html
• https://ccrma.stanford.edu/~jos/pasp/Nonlinear_Distortion.html

(ef.) gate_mono
Mono signal gate. gate_mono is a standard Faust function.

Usage
_ : gate_mono(thresh,att,hold,rel) : _

Where:
• thresh: dB level threshold above which gate opens (e.g., -60 dB)
• att: attack time = time constant (sec) for gate to open (e.g., 0.0001 s = 0.1 ms)
• hold: hold time = time (sec) gate stays open after signal level < thresh (e.g., 0.1 s)
• rel: release time = time constant (sec) for gate to close (e.g., 0.020 s = 20 ms)

References
• http://en.wikipedia.org/wiki/Noise_gate
• http://www.soundonsound.com/sos/apr01/articles/advanced.asp
• http://en.wikipedia.org/wiki/Gating_(sound_engineering)

(ef.) gate_stereo
Stereo signal gates. gate_stereo is a standard Faust function.

Usage
_,_ : gate_stereo(thresh,att,hold,rel) : _,_

Where:
• thresh: dB level threshold above which gate opens (e.g., -60 dB)
• att: attack time = time constant (sec) for gate to open (e.g., 0.0001 s = 0.1 ms)
• **hold**: hold time = time (sec) gate stays open after signal level < thresh (e.g., 0.1 s)
• **rel**: release time = time constant (sec) for gate to close (e.g., 0.020 s = 20 ms)

**References**
- [http://www.soundonsound.com/sos/apr01/articles/advanced.asp](http://www.soundonsound.com/sos/apr01/articles/advanced.asp)

---

**Filtering**

(ef.) **speakerbp**

Dirt-simple speaker simulator (overall bandpass eq with observed roll-offs above and below the passband).

Low-frequency speaker model = +12 dB/octave slope breaking to flat near f1. Implemented using two dc blockers in series.

High-frequency model = -24 dB/octave slope implemented using a fourth-order Butterworth lowpass.

Example based on measured Celestion G12 (12' speaker):

*speakerbp* is a standard Faust function

**Usage**

```faust
speakerbp(f1,f2)
```

_ : speakerbp(130,5000) : _

---

(ef.) **piano_dispersion_filter**

Piano dispersion allpass filter in closed form.

**Usage**

```faust
piano_dispersion_filter(M,B,f0)
```

_ : piano Dispersion Filter(1,B,f0) : +(totalDelay),_ : fdelay(maxDelay) : _

Where:

- **M**: number of first-order allpass sections (compile-time only) Keep below 20. 8 is typical for medium-sized piano strings.
- **B**: string inharmonicity coefficient (0.0001 is typical)
- **f0**: fundamental frequency in Hz
**Outputs**

- MINUS the estimated delay at \( f_0 \) of allpass chain in samples, provided in negative form to facilitate subtraction from delay-line length.
- Output signal from allpass chain

**Reference**

- “Dispersion Modeling in Waveguide Piano Synthesis Using Tunable Allpass Filters”, by Jukka Rauhala and Vesa Valimaki, DAFX-2006, pp. 71-76
- http://www.dafx.ca/proceedings/papers/p_071.pdf (An erratum in Eq. (7) is corrected in Dr. Rauhala’s encompassing dissertation (and below).)
- http://www.acoustics.hut.fi/research/asp/piano/

---

(ef.) **stereo_width**

Stereo Width effect using the Blumlein Shuffler technique. `stereo_width` is a standard Faust function.

**Usage**

```
_:_ : stereo_width(w) : _,_
```

Where:

- \( w \): stereo width between 0 and 1

At \( w=0 \), the output signal is mono ((left+right)/2 in both channels). At \( w=1 \), there is no effect (original stereo image). Thus, \( w \) between 0 and 1 varies stereo width from 0 to “original”.

**Reference**

- “Applications of Blumlein Shuffling to Stereo Microphone Techniques”
  Michael A. Gerzon, JAES vol. 42, no. 6, June 1994

---

(ef.) **mesh_square**

Square Rectangular Digital Waveguide Mesh.

**Usage**

```
bus(4*N) : mesh_square(N) : bus(4*N);
```

Where:
• \( N \): number of nodes along each edge - a power of two (1, 2, 4, 8, \ldots)

Reference

Signal Order In and Out
The mesh is constructed recursively using 2x2 embeddings. Thus, the top level of \( \text{mesh\_square}(M) \) is a block 2x2 mesh, where each block is a \( \text{mesh}(M/2) \). Let these blocks be numbered 1, 2, 3, 4 in the geometry NW, NE, SW, SE, i.e., as 1 2 3 4. Each block has four vector inputs and four vector outputs, where the length of each vector is \( M/2 \). Label the input vectors as Ni, Ei, Wi, Si, i.e., as the inputs from the North, East South, and West, and similarly for the outputs. Then, for example, the upper left input block of \( M/2 \) signals is labeled 1Ni. Most of the connections are internal, such as 1Eo -> 2Wi. The \( 8 \times (M/2) \) input signals are grouped in the order 1Ni 2Ni 3Si 4Si 1Wi 3Wi 2Ei 4Ei and the output signals are 1No 1Wo 2No 2Eo 3So 3Wo 4So 4Eo or

In: 1No 1Wo 2No 2Eo 3So 3Wo 4So 4Eo
Out: 1Ni 2Ni 3Si 4Si 1Wi 3Wi 2Ei 4Ei

Thus, the inputs are grouped by direction N,S,W,E, while the outputs are grouped by block number 1, 2, 3, 4, which can also be interpreted as directions NW, NE, SW, SE. A simple program illustrating these orderings is \( \text{process} = \text{mesh\_square}(2); \).

Example
Reflectively terminated mesh impulsed at one corner:

\[
\text{mesh\_square\_test}(N,x) = \text{mesh\_square}(N)-(\text{busi}(4*N,x)) // \text{input to corner}
\text{with \{ busi(N,x) = bus(N) : par(i,N,*(-1)) : par(i,N-1,_), +(x); \}};
\text{process} = 1-1' : \text{mesh\_square\_test}(4); // \text{all modes excited forever}
\]

In this simple example, the mesh edges are connected as follows:

1No -> 1Ni, 1Wo -> 2Ni, 2No -> 3Si, 2Eo -> 4Si,
3So -> 1Wi, 3Wo -> 3Wi, 4So -> 2Ei, 4Eo -> 4Ei

A routing matrix can be used to obtain other connection geometries.

(ef.) reverseEchoN(nChans, delay)

Reverse echo effect
Usage

_ : ef.reverseEchoN(N,delay) : si.bus(N)

Where:

- `N`: Number of channels desired (1 or more)
- `delay`: echo delay (integer power of 2)

Demo

_ : dm.reverseEchoN(N) : _,_

Description

The effect uses N instances of reverseDelayRamped at different phases.

(ef.)reverseDelayRamped(delay,phase)

Reverse delay with amplitude ramp

Usage

_ : ef.reverseDelayRamped(delay,phase) : _

Where:

- `delay`: echo delay (integer power of 2)
- `phase`: float between 0 and 1 giving ramp delay phase*delay

Demo

_ : dm.reverseEchoN(N) : _,_

(ef.)uniformPanToStereo(nChans)

Pan nChans channels to the stereo field, spread uniformly left to right

Usage

si.bus(N) : ef.uniformPanToStereo(N) : _,_

Where:

- `N`: Number of input channels to pan down to stereo
Demo

```plaintext
_ : dm.reverseEchoN(N) : _,_
```

Time Based

(ef.)echo

A simple echo effect.

echo is a standard Faust function

Usage

```plaintext
_ : echo(maxDuration,duration,feedback) : _
```

Where:

- `maxDuration`: the max echo duration in seconds
- `duration`: the echo duration in seconds
- `feedback`: the feedback coefficient

Pitch Shifting

(ef.)transpose

A simple pitch shifter based on 2 delay lines. transpose is a standard Faust function.

Usage

```plaintext
_ : transpose(w, x, s) : _
```

Where:

- `w`: the window length (samples)
- `x`: crossfade duration duration (samples)
- `s`: shift (semitones)

noises.lib

Faust Noise Generator Library. Its official prefix is `no`.
Functions Reference

(no.)noise
White noise generator (outputs random number between -1 and 1). Noise is a standard Faust function.

Usage
noise : _

(no.)multirandom
Generates multiple decorrelated random numbers in parallel.

Usage
multirandom(n) : si.bus(n)
Where:
• n: the number of decorrelated random numbers in parallel

(no.)multinoise
Generates multiple decorrelated noises in parallel.

Usage
multinoise(n) : si.bus(n)
Where:
• n: the number of decorrelated random numbers in parallel

(no.)noises
TODO.

(no.)pink_noise
Pink noise (1/f noise) generator (third-order approximation) pink_noise is a standard Faust function.
Usage

pink_noise : _;

Reference:
https://ccrma.stanford.edu/~jos/sasp/Example_Synthesis_1_F_Noise.html

(no.)pink_noise_vm
Multi pink noise generator.

Usage

pink_noise_vm(N) : _;

Where:

• N: number of latched white-noise processes to sum, not to exceed sizeof(int) in C++ (typically 32).

References

• http://www.dsprelated.com/showarticle/908.php

(no.)lfnoise, (no.)lfnoise0 and (no.)lfnoiseN
Low-frequency noise generators (Butterworth-filtered downsampled white noise).

Usage

lfnoise0(rate) : _; // new random number every int(SR/rate) samples or so
lfnoiseN(N,rate) : _; // same as "lfnoise0(rate) : lowpass(N,rate)" [see filters.lib]
lfnoise(rate) : _; // same as "lfnoise0(rate) : seq(i,5,lowpass(N,rate))" (no overshoot)

Example

(view waveforms in faust2octave):

ratre = SR/100.0; // new random value every 100 samples (SR from music.lib)
process = lfnoise0(ratre), // sampled/held noise (piecewise constant)
lfnoiseN(3,rate), // lfnoise0 smoothed by 3rd order Butterworth LPF
lfnoise(rate); // lfnoise0 smoothed with no overshoot
sparse_noise_vm

sparse noise generator.

Usage

sparse_noise(f0) : _;

Where:

- f0: average frequency of noise impulses per second

Random impulses in the amplitude range -1 to 1 are generated at an average rate of f0 impulses per second.

Reference

- See velvet_noise

velvet_noise_vm

velvet noise generator.

Usage

velvet_noise(amp,f0) : _;

Where:

- amp: amplitude of noise impulses (positive and negative)
- f0: average frequency of noise impulses per second

Reference


gnoise

approximate zero-mean, unit-variance Gaussian white noise generator.

Usage

gnoise(N) : _;

Where:
• \( N \): number of uniform random numbers added to approximate Gaussian white noise

Reference
• See Central Limit Theorem

oscillators.lib
This library contains a collection of sound generators. Its official prefix is `os`.

Wave-Table-Based Oscillators

(os.)sinwaveform
Sine waveform ready to use with a `rdtable`.

Usage
```
sinwaveform(tablesize) : _
```
Where:
• `tablesize`: the table size

(os.)coswaveform
Cosine waveform ready to use with a `rdtable`.

Usage
```
coswaveform(tablesize) : _
```
Where:
• `tablesize`: the table size

(os.)phasor
A simple phasor to be used with a `rdtable`. `phasor` is a standard Faust function.
Usage
phasor(tablesize,freq) : _
Where:
• tablesize: the table size
• freq: the frequency of the phasor (Hz)

(os.)hs_phasor
Hardsyncing phasor to be used with an rdtable.

Usage
hs_phasor(tablesize,freq,c) : _
Where:
• tablesize: the table size
• freq: the frequency of the phasor (Hz)
• c: a clock signal, c>0 resets phase to 0

(os.)oscsin
Sine wave oscillator. oscsin is a standard Faust function.

Usage
oscsin(freq) : _
Where:
• freq: the frequency of the wave (Hz)

(os.)hs_oscsin
Sin lookup table with hardsyncing phase.

Usage
hs_oscsin(freq,c) : _
Where:
• freq: the fundamental frequency of the phasor
• c: a clock signal, c>0 resets phase to 0
(os.)osc cos
Cosine wave oscillator.

Usage
osc cos(freq) : _
Where:
• freq: the frequency of the wave (Hz)

(os.)osc p
A sine wave generator with controllable phase.

Usage
osc p(freq,p) : _
Where:
• freq: the frequency of the wave (Hz)
• p: the phase in radian

(os.)osci
Interpolated phase sine wave oscillator.

Usage
osci(freq) : _
Where:
• freq: the frequency of the wave (Hz)

LFOs
Low-Frequency Oscillators (LFOs) have prefix lf_ (no aliasing suppression, which is not audible at LF).
(os.)lf_imptrain
Unit-amplitude low-frequency impulse train. \texttt{lf_imptrain} is a standard Faust function.

Usage
\texttt{lf_imptrain(freq)} :
Where:
- \texttt{freq}: frequency in Hz

____________________

(os.)lf_pulsetrainpos
Unit-amplitude nonnegative LF pulse train, duty cycle between 0 and 1.

Usage
\texttt{lf_pulsetrainpos(freq,duty)} :
Where:
- \texttt{freq}: frequency in Hz
- \texttt{duty}: duty cycle between 0 and 1

____________________

(os.)lf_pulsetrain
Unit-amplitude zero-mean LF pulse train, duty cycle between 0 and 1.

Usage
\texttt{lf_pulsetrain(freq,duty)} :
Where:
- \texttt{freq}: frequency in Hz
- \texttt{duty}: duty cycle between 0 and 1

____________________

(os.)lf_squarewavepos
Positive LF square wave in \([0,1]\)
Usage
lf_squarewavepos(freq) : _
Where:
  • freq: frequency in Hz

________________________

(os.)lf_squarewave
Zero-mean unit-amplitude LF square wave. lf_squarewave is a standard Faust function.

Usage
lf_squarewave(freq) : _
Where:
  • freq: frequency in Hz

________________________

(os.)lf_trianglepos
Positive unit-amplitude LF positive triangle wave.

Usage
lf_trianglepos(freq) : _
Where:
  • freq: frequency in Hz

________________________

(os.)lf_triangle
Positive unit-amplitude LF triangle wave lf_triangle is a standard Faust function.

Usage
lf_triangle(freq) : _
Where:
  • freq: frequency in Hz

________________________
Low Frequency Sawtooths

Sawtooth waveform oscillators for virtual analog synthesis et al. The ‘simple’ versions (lf_rawsaw, lf_sawpos and saw1), are mere samplings of the ideal continuous-time (“analog”) waveforms. While simple, the aliasing due to sampling is quite audible. The differentiated polynomial waveform family (saw2, sawN, and derived functions) do some extra processing to suppress aliasing (not audible for very low fundamental frequencies). According to Lehtonen et al. (JASA 2012), the aliasing of saw2 should be inaudible at fundamental frequencies below 2 kHz or so, for a 44.1 kHz sampling rate and 60 dB SPL presentation level; fundamentals 415 and below required no aliasing suppression (i.e., saw1 is ok).

(os.)lf_rawsaw

Simple sawtooth waveform oscillator between 0 and period in samples.

Usage
lf_rawsaw(periodsamps)

Where:

• periodsamps: number of periods per samples

(os.)lf_sawpos_phase

Simple sawtooth waveform oscillator between 0 and 1 with phase control.

Usage
lf_sawpos_phase(freq,phase)

Where:

• freq: frequency
• phase: phase

(os.)lf_sawpos

Simple sawtooth waveform oscillator between 0 and 1.

Usage
lf_sawpos(freq)

Where:
- freq: frequency

(\text{os.})\text{lf\_saw}

Simple sawtooth waveform. \text{lf\_saw} is a standard Faust function.

**Usage**

\text{lf\_saw}(freq)

Where:

- freq: frequency

---

**Bandlimited Sawtooth**

//———\text{(os.)sawN}——— Bandlimited Sawtooth

\text{sawN}(N,freq), \text{sawNp}, \text{saw2dpw}(freq), \text{saw2}(freq), \text{saw3}(freq), \text{saw4}(freq), \text{saw5}(freq), \text{saw6}(freq), \text{sawtooth}(freq), \text{saw2f2}(freq) \text{saw2f4}(freq)

**Method 1 (saw2)**

Polynomial Transition Regions (PTR) (for aliasing suppression).

**References**


**Method 2 (sawN)**

Differentiated Polynomial Waves (DPW) (for aliasing suppression).

**Reference**

Other Cases

Correction-filtered versions of saw2: saw2f2, saw2f4. The correction filter compensates “droop” near half the sampling rate. See reference for sawN.

Usage

sawN(N,freq) : _
sawNp(N,freq,phase) : _
saw2dpw(freq) : _
saw2(freq) : _
saw3(freq) : _ // based on sawN
saw4(freq) : _ // based on sawN
saw5(freq) : _ // based on sawN
saw6(freq) : _ // based on sawN
sawtooth(freq) : _ // = saw2
saw2f2(freq) : _
saw2f4(freq) : _

Where:

• N: polynomial order
• freq: frequency in Hz
• phase: phase

(os.)sawNp

TODO: MarkDown doc in comments

______________________________

(os.)saw2dpw

TODO: MarkDown doc in comments

______________________________

(os.)saw3

TODO: MarkDown doc in comments

______________________________

(os.)sawtooth

Alias-free sawtooth wave. 2nd order interpolation (based on saw2). sawtooth is a standard Faust function.
Usage

`sawtooth(freq)` : _

Where:

- `freq`: frequency

---

(sos.) `saw2f2`

TODO: MarkDown doc in comments

---

(sos.) `saw2f4`

TODO: MarkDown doc in comments

---

**Bandlimited Pulse, Square, and Impulse Trains**

Bandlimited Pulse, Square, and Impulse Trains.

`pulsetrainN, pulsetrain, squareN, square, imptrain, imptrainN, triangle, triangleN`

All are zero-mean and meant to oscillate in the audio frequency range. Use simpler sample-rounded `lf_*` versions above for LFOs.

**Usage**

`pulsetrainN(N,freq,duty)` : _
`pulsetrain(freq, duty)` : _ // = `pulsetrainN(2)`
`squareN(N, freq)` : _
`square` : _ // = `squareN(2)`
`imptrainN(N,freq)` : _
`imptrain` : _ // = `imptrainN(2)`
`triangleN(N,freq)` : _
`triangle` : _ // = `triangleN(2)`

Where:

- `N`: polynomial order
- `freq`: frequency in Hz

(sos.) `pulsetrainN`

TODO: MarkDown doc in comments

---
(os.)pulsetrain

Bandlimited pulse train oscillator. Based on \texttt{pulsetrainN(2)}. \texttt{pulsetrain} is a standard Faust function.

Usage

\texttt{pulsetrain(freq, duty)} : _

Where:

- \texttt{freq}: frequency
- \texttt{duty}: duty cycle between 0 and 1

_____________________________

(os.)squareN

TODO: MarkDown doc in comments

_____________________________

(os.)square

Bandlimited square wave oscillator. Based on \texttt{squareN(2)}. \texttt{square} is a standard Faust function.

Usage

\texttt{square(freq)} : _

Where:

- \texttt{freq}: frequency

_____________________________

(os.)impulse

One-time impulse generated when the Faust process is started. \texttt{impulse} is a standard Faust function.

Usage

\texttt{impulse} : _

_____________________________

(os.)imptrainN

TODO: MarkDown doc in comments

_____________________________
(os.)imptrain
Bandlimited impulse train generator. Based on imptrainN(2). imptrain is a standard Faust function.

Usage
imptrain(freq) : _
Where:
  * freq: frequency

(os.)triangleN
TODO: MarkDown doc in comments

(os.)triangle
Bandlimited triangle wave oscillator. Based on triangleN(2). triangle is a standard Faust function.

Usage
triangle(freq) : _
Where:
  * freq: frequency

Filter-Based Oscillators
Filter-Based Oscillators

Usage
osc[b|r|rs|rc|s|w](f), where f = frequency in Hz.

References

(os.)oscb
Sinusoidal oscillator based on the biquad.
Usage
oscb(freq) : _
Where:
• freq: frequency

(os.)oscrq
Sinusoidal (sine and cosine) oscillator based on 2D vector rotation, = undamped “coupled-form” resonator = lossless 2nd-order normalized ladder filter.

Usage
oscrq(freq) : _,_  
Where:
• freq: frequency

Reference
• https://ccrma.stanford.edu/~jos/pasp/Normalized_Scattering_Junctions.html

(os.)oscrs
Sinusoidal (sine) oscillator based on 2D vector rotation, = undamped “coupled-form” resonator = lossless 2nd-order normalized ladder filter.

Usage
oscrs(freq) : _
Where:
• freq: frequency

Reference
• https://ccrma.stanford.edu/~jos/pasp/Normalized_Scattering_Junctions.html
(os.)oscrc
Sinusoidal (cosine) oscillator based on 2D vector rotation, = undamped “coupled-form” resonator = lossless 2nd-order normalized ladder filter.

Usage
oscrc(freq) : _
Where:
  • freq: frequency

Reference
  • https://ccrma.stanford.edu/~jos/pasp/Normalized_Scattering_Junctions.html

(os.)ocs
Sinusoidal oscillator based on the state variable filter = undamped “modified-coupled-form” resonator = “magic circle” algorithm used in graphics.

(os.)osc
Default sine wave oscillator (same as oscsin). osc is a standard Faust function.

Usage
osc(freq) : _
Where:
  • freq: the frequency of the wave (Hz)

Waveguide-Resonator-Based Oscillators
Sinusoidal oscillator based on the waveguide resonator wgr.

(os.)oscw
Sinusoidal oscillator based on the waveguide resonator wgr. Unit-amplitude cosine oscillator.
Usage
oscwc(freq) : _

Where:
• **freq**: frequency

Reference
• https://ccrma.stanford.edu/~jos/pasp/Digital_Waveguide_Oscillator.html

---

(os.)oscws

Sinusoidal oscillator based on the waveguide resonator wgr. Unit-amplitude sine oscillator.

Usage
oscws(freq) : _

Where:
• **freq**: frequency

Reference
• https://ccrma.stanford.edu/~jos/pasp/Digital_Waveguide_Oscillator.html

---

(os.)oscwq

Sinusoidal oscillator based on the waveguide resonator wgr. Unit-amplitude cosine and sine (quadrature) oscillator.

Usage
oscwq(freq) : _

Where:
• **freq**: frequency
Reference


(os.) oscw
Sinusoidal oscillator based on the waveguide resonator wgr. Unit-amplitude cosine oscillator (default).

Usage
oscw(freq) : _
Where:
- freq: frequency

Reference

Casio CZ Oscillators
Oscillators that mimics some of the Casio CZ oscillators.

(os.) CZsaw
Oscillator that mimics the Casio CZ saw oscillator CZsaw is a standard Faust function.

Usage
CZsaw(fund,index) : _
Where:
- fund: a saw-tooth waveform between 0 and 1 that the oscillator slaves to
- index: the brightness of the oscillator, 0 to 1. 0 = sine-wave, 1 = saw-wave

(os.) CZsquare
Oscillator that mimics the Casio CZ square oscillator CZsquare is a standard Faust function.
Usage

CZsquare(fund,index) : _

Where:

- **fund**: a saw-tooth waveform between 0 and 1 that the oscillator slaves to
- **index**: the brightness of the oscillator, 0 to 1. 0 = sine-wave, 1 = square-wave

---

(os.)CZpulse

Oscillator that mimics the Casio CZ pulse oscillator **CZpulse** is a standard Faust function.

Usage

CZpulse(fund,index) : _

Where:

- **fund**: a saw-tooth waveform between 0 and 1 that the oscillator slaves to
- **index**: the brightness of the oscillator, 0 gives a sine-wave, 1 is closer to a pulse

---

(os.)CZsinePulse

Oscillator that mimics the Casio CZ sine/pulse oscillator **CZsinePulse** is a standard Faust function.

Usage

CZsinePulse(fund,index) : _

Where:

- **fund**: a saw-tooth waveform between 0 and 1 that the oscillator slaves to
- **index**: the brightness of the oscillator, 0 gives a sine-wave, 1 is a sine minus a pulse

---

(os.)CZhalfSine

Oscillator that mimics the Casio CZ half sine oscillator **CZhalfSine** is a standard Faust function.
Usage
CZhalfSine(fund,index) : _
Where:

- fund: a saw-tooth waveform between 0 and 1 that the oscillator slaves to
- index: the brightness of the oscillator, 0 gives a sine-wave, 1 is somewhere between a saw and a square

(os.)CZresSaw
Oscillator that mimics the Casio CZ resonant saw-tooth oscillator CZresSaw is a standard Faust function.

Usage
CZresSaw(fund,res) : _
Where:

- fund: a saw-tooth waveform between 0 and 1 that the oscillator slaves to
- res: the frequency of resonance as a factor of the fundamental pitch.

(os.)CZresTriangle
Oscillator that mimics the Casio CZ resonant triangle oscillator CZresTriangle is a standard Faust function.

Usage
CZresTriangle(fund,res) : _
Where:

- fund: a saw-tooth waveform between 0 and 1 that the oscillator slaves to
- res: the frequency of resonance as a factor of the fundamental pitch.

(os.)CZresTrap
Oscillator that mimics the Casio CZ resonant trapeze oscillator CZresTrap is a standard Faust function.
Usage
CZresTrap(fund, res) : _

Where:
- **fund**: a saw-tooth waveform between 0 and 1 that the oscillator slaves to
- **res**: the frequency of resonance as a factor of the fundamental pitch.

---

**PolyBLEP-Based Oscillators**

(os.)polyblep

PolyBLEP residual function - used for smoothing steps in the audio signal.

Usage
polyblep(Q, phase) : _

Where:
- **Q**: smoothing factor between 0 and 0.5. Determines how far from the ends of the phase interval the quadratic function is used.
- **phase**: normalised phase (between 0 and 1)

---

(os.)polyblep_saw

Sawtooth oscillator with suppressed aliasing (using polyBLEP)

Usage
polyblep_saw(f) : _

Where:
- **f**: frequency in Hz

---

(os.)polyblep_square

Square wave oscillator with suppressed aliasing (using polyBLEP)

Usage
polyblep_square(f) : _

Where:
- **f**: frequency in Hz
(os.)polyblep_triangle
Triangle wave oscillator with suppressed aliasing (using polyBLEP)

Usage
polyblep_triangle(f) : _
Where:
  • f: frequency in Hz

Filter-Based Oscillators
(os.)quadosc
Sinusoidal oscillator based on QuadOsc by Martin Vicanek

Usage
quadosc(freq) : _
where
  • freq: frequency in Hz

Reference
  • https://vicanek.de/articles/QuadOsc.pdf

phaflangers.lib
A library of phasor and flanger effects. Its official prefix is pf.

Functions Reference
(pf.)flanger_mono
Mono flanging effect.
Usage:

_ : flanger_mono(dmax, curdel, depth, fb, invert) : _;

Where:

- **dmax**: maximum delay-line length (power of 2) - 10 ms typical
- **curdel**: current dynamic delay (not to exceed dmax)
- **depth**: effect strength between 0 and 1 (1 typical)
- **fb**: feedback gain between 0 and 1 (0 typical)
- **invert**: 0 for normal, 1 to invert sign of flanging sum

Reference

https://ccrma.stanford.edu/~jos/pasp/Flanging.html

(pf.) flanger_stereo

Stereo flanging effect. flanger_stereo is a standard Faust function.

Usage:

_ ,_ : flanger_stereo(dmax, curdel1, curdel2, depth, fb, invert) : _ ,_;

Where:

- **dmax**: maximum delay-line length (power of 2) - 10 ms typical
- **curdel**: current dynamic delay (not to exceed dmax)
- **depth**: effect strength between 0 and 1 (1 typical)
- **fb**: feedback gain between 0 and 1 (0 typical)
- **invert**: 0 for normal, 1 to invert sign of flanging sum

Reference

https://ccrma.stanford.edu/~jos/pasp/Flanging.html

(pf.) phaser2_mono

Mono phasing effect.

Phaser

_ : phaser2_mono(Notches, phase, width, frqmin, fratio, frqmax, speed, depth, fb, invert) : _;

Where:

- **Notches**: number of spectral notches (MACRO ARGUMENT - not a signal)
- **phase**: phase of the oscillator (0-1)
- **width**: approximate width of spectral notches in Hz
- **frqmin**: approximate minimum frequency of first spectral notch in Hz
- **fratio**: ratio of adjacent notch frequencies
- **frqmax**: approximate maximum frequency of first spectral notch in Hz
- **speed**: LFO frequency in Hz (rate of periodic notch sweep cycles)
- **depth**: effect strength between 0 and 1 (1 typical) (aka “intensity”) when depth=2, “vibrato mode” is obtained (pure allpass chain)
- **fb**: feedback gain between -1 and 1 (0 typical)
- **invert**: 0 for normal, 1 to invert sign of flanging sum

**Reference:**
- https://ccrma.stanford.edu/~jos/pasp/Phasing.html

---

**(pf.)** _phaser2_stereo_

Stereo phasing effect. _phaser2_stereo_ is a standard Faust function.

**Phaser**

```faust
_ : phaser2_stereo(Notches,phase,width,frqmin,fratio,frqmax,speed,depth,fb,invert) : _;
```

Where:

- **Notches**: number of spectral notches (MACRO ARGUMENT - not a signal)
- **phase**: phase of the oscillator (0-1)
- **width**: approximate width of spectral notches in Hz
- **frqmin**: approximate minimum frequency of first spectral notch in Hz
- **fratio**: ratio of adjacent notch frequencies
- **frqmax**: approximate maximum frequency of first spectral notch in Hz
- **speed**: LFO frequency in Hz (rate of periodic notch sweep cycles)
- **depth**: effect strength between 0 and 1 (1 typical) (aka “intensity”) when depth=2, “vibrato mode” is obtained (pure allpass chain)
- **fb**: feedback gain between -1 and 1 (0 typical)
- **invert**: 0 for normal, 1 to invert sign of flanging sum

**Reference:**
- https://ccrma.stanford.edu/~jos/pasp/Phasing.html
physmodels.lib

Faust physical modeling library; Its official prefix is \texttt{pm}.

This library provides an environment to facilitate physical modeling of musical instruments. It contains dozens of functions implementing low and high level elements going from a simple waveguide to fully operational models with built-in UI, etc.

It is organized as follows:

- Global Variables: Useful pre-defined variables for physical modeling (e.g., speed of sound, etc.).
- Conversion Tools: Conversion functions specific to physical modeling (e.g., length to frequency, etc.).
- Bidirectional Utilities: Functions to create bidirectional block diagrams for physical modeling.
- Basic Elements: waveguides, specific types of filters, etc.
- String Instruments: various types of strings (e.g., steel, nylon, etc.), bridges, guitars, etc.
- Bowed String Instruments: parts and models specific to bowed string instruments (e.g., bows, bridges, violins, etc.).
- Wind Instrument: parts and models specific to wind string instruments (e.g., reeds, mouthpieces, flutes, clarinets, etc.).
- Exciters: pluck generators, “blowers”, etc.
- Modal Percussions: percussion instruments based on modal models.
- Vocal Synthesis: functions for various vocal synthesis techniques (e.g., fof, source/filter, etc.) and vocal synthesizers.
- Misc Functions: any other functions that don’t fit in the previous category (e.g., nonlinear filters, etc.).

This library is part of the Faust Physical Modeling ToolKit. More information on how to use this library can be found on this page: https://ccrma.stanford.edu/~rmichon/pmFaust. Tutorials on how to make physical models of musical instruments using Faust can be found here as well.

**Global Variables**

Useful pre-defined variables for physical modeling.
(pm.)speedOfSound
Speed of sound in meters per second (340m/s).

____________________________

(pm.)maxLength
The default maximum length (3) in meters of strings and tubes used in this library. This variable should be overridden to allow longer strings or tubes.

____________________________

Conversion Tools
Useful conversion tools for physical modeling.

(pm.)f2l
Frequency to length in meters.

Usage
f2l(freq) : distanceInMeters
Where:
- freq: the frequency

____________________________

(pm.)l2f
Length in meters to frequency.

Usage
l2f(length) : freq
Where:
- length: length/distance in meters

____________________________

(pm.)l2s
Length in meters to number of samples.
Usage
l2s(l) : numberOfSamples
Where:
• l: length in meters

Bidirectional Utilities
Set of fundamental functions to create bi-directional block diagrams in Faust. These elements are used as the basis of this library to connect high level elements (e.g., mouthpieces, strings, bridge, instrument body, etc.). Each block has 3 inputs and 3 outputs. The first input/output carry left going waves, the second input/output carry right going waves, and the third input/output is used to carry any potential output signal to the end of the algorithm.

(pm.)basicBlock
Empty bidirectional block to be used with chain: 3 signals ins and 3 signals out.

Usage
chain(basicBlock : basicBlock : etc.)

(pm.)chain
Creates a chain of bidirectional blocks. Blocks must have 3 inputs and outputs. The first input/output carry left going waves, the second input/output carry right going waves, and the third input/output is used to carry any potential output signal to the end of the algorithm. The implied one sample delay created by the ~ operator is generalized to the left and right going waves. Thus, n blocks in chain() will add an n samples delay to both left and right going waves.

Usage
leftGoingWaves,rightGoingWaves,mixedOutput : chain( A : B ) : leftGoingWaves,rightGoingWaves
with{
    A = _,_,_
    B = _,_,_
};

(pm.)inLeftWave
Adds a signal to left going waves anywhere in a chain of blocks.
Usage

\[ \text{model}(x) = \text{chain}(A : \text{inLeftWave}(x) : B) \]

Where \( A \) and \( B \) are bidirectional blocks and \( x \) is the signal added to left going waves in that chain.

\[
\text{(pm.)inRightWave}
\]

Adds a signal to right going waves anywhere in a \text{chain} of blocks.

Usage

\[ \text{model}(x) = \text{chain}(A : \text{inRightWave}(x) : B) \]

Where \( A \) and \( B \) are bidirectional blocks and \( x \) is the signal added to right going waves in that chain.

\[
\text{(pm.)in}
\]

Adds a signal to left and right going waves anywhere in a \text{chain} of blocks.

Usage

\[ \text{model}(x) = \text{chain}(A : \text{in}(x) : B) \]

Where \( A \) and \( B \) are bidirectional blocks and \( x \) is the signal added to left and right going waves in that chain.

\[
\text{(pm.)outLeftWave}
\]

Sends the signal of left going waves to the output channel of the \text{chain}.

Usage

\[ \text{chain}(A : \text{outLeftWave} : B) \]

Where \( A \) and \( B \) are bidirectional blocks.

\[
\text{(pm.)outRightWave}
\]

Sends the signal of right going waves to the output channel of the \text{chain}. 
Usage

chain(A : outRightWave : B)
Where A and B are bidirectional blocks.

/pm.)/out
Sends the signal of right and left going waves to the output channel of the chain.

Usage

chain(A : out : B)
Where A and B are bidirectional blocks.

/pm.)/terminations
Creates terminations on both sides of a chain without closing the inputs and outputs of the bidirectional signals chain. As for chain, this function adds a 1 sample delay to the bidirectional signal, both ways. Of courses, this function can be nested within a chain.

Usage

terminations(a,b,c)
with{
  a = *(-1); // left termination
  b = chain(D : E : F); // bidirectional chain of blocks (D, E, F, etc.)
  c = *(-1); // right termination
};

/pm.)/lTermination
Creates a termination on the left side of a chain without closing the inputs and outputs of the bidirectional signals chain. This function adds a 1 sample delay near the termination and can be nested within another chain.

Usage

lTerminations(a,b)
with{
  a = *(-1); // left termination
  b = chain(D : E : F); // bidirectional chain of blocks (D, E, F, etc.)
};
(pm.) **rTermination**

Creates a termination on the right side of a chain without closing the inputs and outputs of the bidirectional signals chain. This function adds a 1 sample delay near the termination and can be nested within another chain.

**Usage**

```plaintext
rTerminations(b,c) with{
    b = chain(D : E : F); // bidirectional chain of blocks (D, E, F, etc.)
    c = *(-1); // right termination
}
```

(pm.) **closeIns**

Closes the inputs of a bidirectional chain in all directions.

**Usage**

```plaintext
closeIns : chain(...) : _,_,_
```

(pm.) **closeOuts**

Closes the outputs of a bidirectional chain in all directions except for the main signal output (3d output).

**Usage**

```plaintext
_,_,_ : chain(...) : _
```

(pm.) **endChain**

Closes the inputs and outputs of a bidirectional chain in all directions except for the main signal output (3d output).

**Usage**

```plaintext
endChain(chain(...)) : _
```
Basic Elements

Basic elements for physical modeling (e.g., waveguides, specific filters, etc.).

(pm.) waveguideN

A series of waveguide functions based on various types of delays (see fdelay[n]).

List of functions

- waveguideUd: unit delay waveguide
- waveguideFd: fractional delay waveguide
- waveguideFd2: second order fractional delay waveguide
- waveguideFd4: fourth order fractional delay waveguide

Usage

chain(A : waveguideUd(nMax,n) : B)

Where:

- nMax: the maximum length of the delays in the waveguide
- n: the length of the delay lines in samples.

(pm.) waveguide

Standard pm.lib waveguide (based on waveguideFd4).

Usage

chain(A : waveguide(nMax,n) : B)

Where:

- nMax: the maximum length of the delays in the waveguide
- n: the length of the delay lines in samples.

(pm.) bridgeFilter

Generic two zeros bridge FIR filter (as implemented in the STK) that can be used to implement the reflectance violin, guitar, etc. bridges.

Usage

_ : bridge(brightness,absorption) : _

Where:
• **brightness**: controls the damping of high frequencies (0-1)
• **absorption**: controls the absorption of the bridge and thus the t60 of the string plugged to it (0-1) (1 = 20 seconds)

(pm.)**modeFilter**

Resonant bandpass filter that can be used to implement a single resonance (mode).

**Usage**

```
_: modeFilter(freq,t60,gain) : _
```

Where:

• **freq**: mode frequency
• **t60**: mode resonance duration (in seconds)
• **gain**: mode gain (0-1)

(pm.)**stringSegment**

A string segment without terminations (just a simple waveguide).

**Usage**

```
chain(A : stringSegment(maxLength,length) : B)
```

Where:

• **maxLength**: the maximum length of the string in meters (should be static)
• **length**: the length of the string in meters

(pm.)**openString**

A bidirectional block implementing a basic “generic” string with a selectable excitation position. Lowpass filters are built-in and allow to simulate the effect of dispersion on the sound and thus to change the “stiffness” of the string.
Usage

chain(... : openString(length, stiffness, pluckPosition, excitation) : ...)

Where:

- `length`: the length of the string in meters
- `stiffness`: the stiffness of the string (0-1) (1 for max stiffness)
- `pluckPosition`: excitation position (0-1) (1 is bottom)
- `excitation`: the excitation signal

(pm.) nylonString

A bidirectional block implementing a basic nylon string with selectable excitation position. This element is based on `openString` and has a fix stiffness corresponding to that of a nylon string.

Usage

chain(... : nylonString(length, pluckPosition, excitation) : ...)

Where:

- `length`: the length of the string in meters
- `pluckPosition`: excitation position (0-1) (1 is bottom)
- `excitation`: the excitation signal

/pm.) steelString

A bidirectional block implementing a basic steel string with selectable excitation position. This element is based on `openString` and has a fix stiffness corresponding to that of a steel string.

Usage

chain(... : steelString(length, pluckPosition, excitation) : ...)

Where:

- `length`: the length of the string in meters
- `pluckPosition`: excitation position (0-1) (1 is bottom)
- `excitation`: the excitation signal
(pm.)openStringPick
A bidirectional block implementing a “generic” string with selectable excitation position. It also has a built-in pickup whose position is the same as the excitation position. Thus, moving the excitation position will also move the pickup.

Usage
chain(... : openStringPick(length,stiffness,pluckPosition,excitation) : ...)

Where:
- length: the length of the string in meters
- stiffness: the stiffness of the string (0-1) (1 for max stiffness)
- pluckPosition: excitation position (0-1) (1 is bottom)
- excitation: the excitation signal

(pm.)openStringPickUp
A bidirectional block implementing a “generic” string with selectable excitation position and stiffness. It also has a built-in pickup whose position can be independently selected. The only constraint is that the pickup has to be placed after the excitation position.

Usage
chain(... : openStringPickUp(length,stiffness,pluckPosition,excitation) : ...)

Where:
- length: the length of the string in meters
- stiffness: the stiffness of the string (0-1) (1 for max stiffness)
- pluckPosition: pluck position between the top of the string and the pickup (0-1) (1 for same as pickup position)
- pickupPosition: position of the pickup on the string (0-1) (1 is bottom)
- excitation: the excitation signal

(pm.)openStringPickDown
A bidirectional block implementing a “generic” string with selectable excitation position and stiffness. It also has a built-in pickup whose position can be independently selected. The only constraint is that the pickup has to be placed before the excitation position.
Usage

chain(... : openStringPickDown(length, stiffness, pluckPosition, excitation) : ...)

Where:

- **length**: the length of the string in meters
- **stiffness**: the stiffness of the string (0-1) (1 for max stiffness)
- **pluckPosition**: pluck position on the string (0-1) (1 is bottom)
- **pickupPosition**: position of the pickup between the top of the string and the excitation position (0-1) (1 is excitation position)
- **excitation**: the excitation signal

(pm.)ksReflexionFilter

The “typical” one-zero Karplus-strong feedforward reflexion filter. This filter will be typically used in a termination (see below).

Usage

terminations(_, chain(...), ksReflexionFilter)

(pm.)rStringRigidTermination

Bidirectional block implementing a right rigid string termination (no damping, just phase inversion).

Usage

chain(rStringRigidTermination : stringSegment : ...)

(pm.)lStringRigidTermination

Bidirectional block implementing a left rigid string termination (no damping, just phase inversion).

Usage

chain(... : stringSegment : lStringRigidTermination)
(pm.)elecGuitarBridge
Bidirectional block implementing a simple electric guitar bridge. This block is based on bridgeFilter. The bridge doesn’t implement transmittance since it is not meant to be connected to a body (unlike acoustic guitar). It also partially sets the resonance duration of the string with the nuts used on the other side.

Usage
chain(... : stringSegment : elecGuitarBridge)

(pm.)elecGuitarNuts
Bidirectional block implementing a simple electric guitar nuts. This block is based on bridgeFilter and does essentially the same thing as elecGuitarBridge, but on the other side of the chain. It also partially sets the resonance duration of the string with the bridge used on the other side.

Usage
chain(elecGuitarNuts : stringSegment : ...)

(pm.)guitarBridge
Bidirectional block implementing a simple acoustic guitar bridge. This bridge damps more high frequencies than elecGuitarBridge and implements a transmittance filter. It also partially sets the resonance duration of the string with the nuts used on the other side.

Usage
chain(... : stringSegment : guitarBridge)

(pm.)guitarNuts
Bidirectional block implementing a simple acoustic guitar nuts. This nuts damps more high frequencies than elecGuitarNuts and implements a transmittance filter. It also partially sets the resonance duration of the string with the bridge used on the other side.

Usage
chain(guitarNuts : stringSegment : ...)
An “ideal” string with rigid terminations and where the plucking position and the pick-up position are the same. Since terminations are rigid, this string will ring forever.

Usage

idealString(length, reflexion, xPosition, excitation)

With:
- **length**: the length of the string in meters
- **pluckPosition**: the plucking position (0.001-0.999)
- **excitation**: the input signal for the excitation.

A Karplus-Strong string (in that case, the string is implemented as a one dimension waveguide).

Usage

ks(length, damping, excitation)

Where:
- **length**: the length of the string in meters
- **damping**: string damping (0-1)
- **excitation**: excitation signal

Ready-to-use, MIDI-enabled Karplus-Strong string with built-in UI.

Usage

ks_ui_MIDI

A simple electric guitar model (without audio effects, of course) with selectable pluck position. This model implements a single string. Additional strings should be created by making a polyphonic applications out of this function. Pitch is changed by changing the length of the string and not through a finger model.
Usage
elecGuitarModel(length,pluckPosition,mute,excitation) : _

Where:
- **length**: the length of the string in meters
- **pluckPosition**: pluck position (0-1) (1 is on the bridge)
- **mute**: mute coefficient (1 for no mute and 0 for instant mute)
- **excitation**: excitation signal

(pm.)elecGuitar

A simple electric guitar model with steel strings (based on elecGuitarModel) implementing an excitation model. This model implements a single string. Additional strings should be created by making a polyphonic applications out of this function.

Usage
elecGuitar(length,pluckPosition,trigger) : _

Where:
- **length**: the length of the string in meters
- **pluckPosition**: pluck position (0-1) (1 is on the bridge)
- **mute**: mute coefficient (1 for no mute and 0 for instant mute)
- **gain**: gain of the pluck (0-1)
- **trigger**: trigger signal (1 for on, 0 for off)

(pm.)elecGuitar_ui_MIDI

Ready-to-use MIDI-enabled electric guitar physical model with built-in UI.

Usage
elecGuitar_ui_MIDI : _

(pm.)guitarBody

WARNING: not implemented yet! Bidirectional block implementing a simple acoustic guitar body.
Usage
chain(... : guitarBody)

(pm.)guitarModel
A simple acoustic guitar model with steel strings and selectable excitation position. This model implements a single string. Additional strings should be created by making a polyphonic applications out of this function. Pitch is changed by changing the length of the string and not through a finger model. WARNING: this function doesn’t currently implement a body (just strings and bridge).

Usage
guitarModel(length,pluckPosition,excitation) : _
Where:
- length: the length of the string in meters
- pluckPosition: pluck position (0-1) (1 is on the bridge)
- excitation: excitation signal

(pm.)guitar
A simple acoustic guitar model with steel strings (based on guitarModel) implementing an excitation model. This model implements a single string. Additional strings should be created by making a polyphonic applications out of this function.

Usage
guitar(length,pluckPosition,trigger) : _
Where:
- length: the length of the string in meters
- pluckPosition: pluck position (0-1) (1 is on the bridge)
- gain: gain of the excitation
- trigger: trigger signal (1 for on, 0 for off)

(pm.)guitar_ui_MIDI
Ready-to-use MIDI-enabled steel strings acoustic guitar physical model with built-in UI.
Usage
guitar_ui_MIDI : _

(p.m.)nylonGuitarModel
A simple acoustic guitar model with nylon strings and selectable excitation position. This model implements a single string. Additional strings should be created by making a polyphonic applications out of this function. Pitch is changed by changing the length of the string and not through a finger model. WARNING: this function doesn’t currently implement a body (just strings and bridge).

Usage
nylonGuitarModel(length,pluckPosition,excitation) : _
Where:
- **length**: the length of the string in meters
- **pluckPosition**: pluck position (0-1) (1 is on the bridge)
- **excitation**: excitation signal

(p.m.)nylonGuitar
A simple acoustic guitar model with steel strings (based on nylonGuitarModel) implementing an excitation model. This model implements a single string. Additional strings should be created by making a polyphonic applications out of this function.

Usage
nylonGuitar(length,pluckPosition,trigger) : _
Where:
- **length**: the length of the string in meters
- **pluckPosition**: pluck position (0-1) (1 is on the bridge)
- **gain**: gain of the excitation (0-1)
- **trigger**: trigger signal (1 for on, 0 for off)

(p.m.)nylonGuitar_ui_MIDI
Ready-to-use MIDI-enabled nylon strings acoustic guitar physical model with built-in UI.
Usage
nylonGuitar_ui_MIDI : _

________________________

(pm.) modeInterpRes

Modular string instrument resonator based on IR measurements made on 3D printed models. The 2D space allowing for the control of the shape and the scale of the model is enabled by interpolating between modes parameters. More information about this technique/project can be found here: https://ccrma.stanford.edu/~rmichon/3dPrintingModeling/.

Usage
_ : modeInterpRes(nModes,x,y) : _

Where:

- **nModes**: number of modeled modes (40 max)
- **x**: shape of the resonator (0: square, 1: square with rounded corners, 2: round)
- **y**: scale of the resonator (0: small, 1: medium, 2: large)

________________________

(pm.) modularInterpBody

Bidirectional block implementing a modular string instrument resonator (see modeInterpRes).

Usage
chain(... : modularInterpBody(nModes,shape,scale) : ...)

Where:

- **nModes**: number of modeled modes (40 max)
- **shape**: shape of the resonator (0: square, 1: square with rounded corners, 2: round)
- **scale**: scale of the resonator (0: small, 1: medium, 2: large)

________________________

(pm.) modularInterpStringModel

String instrument model with a modular body (see modeInterpRes and https://ccrma.stanford.edu/~rmichon/3dPrintingModeling/).
**Usage**

`modularInterpStringModel(length, pluckPosition, shape, scale, bodyExcitation, stringExcitation)`

Where:
- **stringLength**: the length of the string in meters
- **pluckPosition**: pluck position (0-1) (1 is on the bridge)
- **shape**: shape of the resonator (0: square, 1: square with rounded corners, 2: round)
- **scale**: scale of the resonator (0: small, 1: medium, 2: large)
- **bodyExcitation**: excitation signal for the body
- **stringExcitation**: excitation signal for the string

---

*(pm.)* `modularInterpInstr`

String instrument with a modular body (see `modeInterpRes` and [https://ccrma.stanford.edu/~rmichon/3dPrintingModeling/]).

**Usage**

`modularInterpInstr(stringLength, pluckPosition, shape, scale, gain, tapBody, triggerString)`

Where:
- **stringLength**: the length of the string in meters
- **pluckPosition**: pluck position (0-1) (1 is on the bridge)
- **shape**: shape of the resonator (0: square, 1: square with rounded corners, 2: round)
- **scale**: scale of the resonator (0: small, 1: medium, 2: large)
- **gain**: of the string excitation
- **tapBody**: send an impulse in the body of the instrument where the string is connected (1 for on, 0 for off)
- **triggerString**: trigger signal for the string (1 for on, 0 for off)

---

*(pm.)* `modularInterpInstr_ui_MIDI`

Ready-to-use MIDI-enabled string instrument with a modular body (see `modeInterpRes` and [https://ccrma.stanford.edu/~rmichon/3dPrintingModeling/] with built-in UI.

**Usage**

`modularInterpInstr_ui_MIDI`
Bowed String Instruments

Low and high level basic string instruments parts. Most of the elements in this section can be used in a bidirectional chain.

(pm.)bowTable

Extremely basic bow table that can be used to implement a wide range of bow types for many different bowed string instruments (violin, cello, etc.).

Usage

excitation : bowTable(offset, slope) : _

Where:

• excitation: an excitation signal
• offset: table offset
• slope: table slope

(pm.)violinBowTable

Violin bow table based on bowTable.

Usage

bowVelocity : violinBowTable(bowPressure) : _

Where:

• bowVelocity: velocity of the bow/excitation signal (0-1)
• bowPressure: bow pressure on the string (0-1)

(pm.)bowInteraction

Bidirectional block implementing the interaction of a bow in a chain.

Usage

chain(... : stringSegment : bowInteraction(bowTable) : stringSegment : ...)
(pm.)violinBow
Bidirectional block implementing a violin bow and its interaction with a string.

Usage
chain(... : stringSegment : violinBow(bowPressure,bowVelocity) : stringSegment : ...) 
Where:
- bowVelocity: velocity of the bow / excitation signal (0-1)
- bowPressure: bow pressure on the string (0-1)

(pm.)violinBowedString
Violin bowed string bidirectional block with controllable bow position. Terminations are not implemented in this model.

Usage
chain(nuts : violinBowedString(stringLength,bowPressure,bowVelocity,bowPosition) : bridge)
Where:
- stringLength: the length of the string in meters
- bowVelocity: velocity of the bow / excitation signal (0-1)
- bowPressure: bow pressure on the string (0-1)
- bowPosition: the position of the bow on the string (0-1)

(pm.)violinNuts
Bidirectional block implementing simple violin nuts. This function is based on bridgeFilter.

Usage
chain(violinNuts : stringSegment : ...)

(pm.)violinBridge
Bidirectional block implementing a simple violin bridge. This function is based on bridgeFilter.
Usage

chain(... : stringSegment : violinBridge

(pm.)violinBody
Bidirectional block implementing a simple violin body (just a simple resonant
lowpass filter).

Usage

chain(... : stringSegment : violinBridge : violinBody

(pm.)violinModel
Ready-to-use simple violin physical model. This model implements a single
string. Additional strings should be created by making a polyphonic applications
out of this function. Pitch is changed by changing the length of the string (and
not through a finger model).

Usage

violinModel(stringLength,bowPressure,bowVelocity,bridgeReflexion,
bridgeAbsorption,bowPosition) : _

Where:

- **stringLength**: the length of the string in meters
- **bowVelocity**: velocity of the bow / excitation signal (0-1)
- **bowPressure**: bow pressure on the string (0-1))
- **bowPosition**: the position of the bow on the string (0-1)

(pm.)violin_ui
Ready-to-use violin physical model with built-in UI.

Usage

violinModel_ui : _

(pm.)violin_ui_MIDI
Ready-to-use MIDI-enabled violin physical model with built-in UI.
Usage

violin_ui_MIDI : _

Wind Instruments

Low and high level basic wind instruments parts. Most of the elements in this section can be used in a bidirectional chain.

(pm.)openTube

A tube segment without terminations (same as stringSegment).

Usage

chain(A : openTube(maxLength,length) : B)

Where:

• maxLength: the maximum length of the tube in meters (should be static)
• length: the length of the tube in meters

(pm.)reedTable

Extremely basic reed table that can be used to implement a wide range of single reed types for many different instruments (saxophone, clarinet, etc.).

Usage

excitation : reedTable(offset,slope) : _

Where:

• excitation: an excitation signal
• offset: table offset
• slope: table slope

(pm.)fluteJetTable

Extremely basic flute jet table.

Usage

excitation : fluteJetTable : _

Where:
• excitation: an excitation signal

------------------------------------------------------------------

(pm.)brassLipsTable

Simple brass lips/mouthpiece table. Since this implementation is very basic and that the lips and tube of the instrument are coupled to each other, the length of that tube must be provided here.

Usage

excitation : brassLipsTable(tubeLength,lipsTension) : _

Where:

• excitation: an excitation signal (can be DC)
• tubeLength: length in meters of the tube connected to the mouthpiece
• lipsTension: tension of the lips (0-1) (default: 0.5)

------------------------------------------------------------------

(pm.)clarinetReed

Clarinet reed based on reedTable with controllable stiffness.

Usage

excitation : clarinetReed(stiffness) : _

Where:

• excitation: an excitation signal
• stiffness: reed stiffness (0-1)

------------------------------------------------------------------

(pm.)clarinetMouthPiece

Bidirectional block implementing a clarinet mouthpiece as well as the various interactions happening with traveling waves. This element is ready to be plugged to a tube...

Usage

chain(clarinetMouthPiece(reedStiffness,pressure) : tube : etc.)

Where:

• pressure: the pressure of the air flow (DC) created by the virtual performer (0-1). This can also be any kind of signal that will directly injected in the mouthpiece (e.g., breath noise, etc.).
- **reedStiffness**: reed stiffness (0-1)

(pm.) **brassLips**
Bidirectional block implementing a brass mouthpiece as well as the various interactions happening with traveling waves. This element is ready to be plugged to a tube...

**Usage**

chain(brassLips(tubeLength, lipsTension, pressure) : tube : etc.)

Where:

- **tubeLength**: length in meters of the tube connected to the mouthpiece
- **lipsTension**: tension of the lips (0-1) (default: 0.5)
- **pressure**: the pressure of the air flow (DC) created by the virtual performer (0-1). This can also be any kind of signal that will directly injected in the mouthpiece (e.g., breath noise, etc.).

(pm.) **fluteEmbouchure**
Bidirectional block implementing a flute embouchure as well as the various interactions happening with traveling waves. This element is ready to be plugged between tubes segments...

**Usage**

chain(... : tube : fluteEmbouchure(pressure) : tube : etc.)

Where:

- **pressure**: the pressure of the air flow (DC) created by the virtual performer (0-1). This can also be any kind of signal that will directly injected in the mouthpiece (e.g., breath noise, etc.).

(pm.) **wBell**
Generic wind instrument bell bidirectional block that should be placed at the end of a chain.

**Usage**

chain(... : wBell(opening))

Where:
• opening: the “opening” of bell (0-1)

(pm.) fluteHead
Simple flute head implementing waves reflexion.

Usage
chain(fluteHead : tube : ...)

(pm.) fluteFoot
Simple flute foot implementing waves reflexion and dispersion.

Usage
chain(... : tube : fluteFoot)

(pm.) clarinetModel
A simple clarinet physical model without tone holes (pitch is changed by changing the length of the tube of the instrument).

Usage
clarinetModel(length,pressure,reedStiffness,bellOpening) : _

Where:
• tubeLength: the length of the tube in meters
• pressure: the pressure of the air flow created by the virtual performer (0-1). This can also be any kind of signal that will directly injected in the mouthpiece (e.g., breath noise, etc.).
• reedStiffness: reed stiffness (0-1)
• bellOpening: the opening of bell (0-1)

(pm.) clarinetModel_ui
Same as clarinetModel but with a built-in UI. This function doesn’t implement a virtual “blower”, thus pressure remains an argument here.
Usage
clarinetModel_ui(pressure) : _

Where:

- **pressure**: the pressure of the air flow created by the virtual performer (0-1). This can also be any kind of signal that will be directly injected in the mouthpiece (e.g., breath noise, etc).

(p.m.) clarinet_ui
Ready-to-use clarinet physical model with built-in UI based on clarinetModel.

Usage
clarinet_ui : _

______________________________

(p.m.) clarinet_ui_MIDI
Ready-to-use MIDI compliant clarinet physical model with built-in UI.

Usage
clarinet_ui_MIDI : _

______________________________

(p.m.) brassModel
A simple generic brass instrument physical model without pistons (pitch is changed by changing the length of the tube of the instrument). This model is kind of hard to control and might not sound very good if bad parameters are given to it...

Usage
brassModel(tubeLength, lipsTension, mute, pressure) : _

Where:

- **tubeLength**: the length of the tube in meters
- **lipsTension**: tension of the lips (0-1) (default: 0.5)
- **mute**: mute opening at the end of the instrument (0-1) (default: 0.5)
- **pressure**: the pressure of the air flow created by the virtual performer (0-1). This can also be any kind of signal that will directly injected in the mouthpiece (e.g., breath noise, etc.).
(pm.)brassModel_ui

Same as brassModel but with a built-in UI. This function doesn’t implement a virtual “blower”, thus pressure remains an argument here.

Usage
brassModel_ui(pressure) : _

Where:

• pressure: the pressure of the air flow created by the virtual performer (0-1). This can also be any kind of signal that will be directly injected in the mouthpiece (e.g., breath noise, etc.).

(pm.)brass_ui

Ready-to-use brass instrument physical model with built-in UI based on brassModel.

Usage
brass_ui : _

-----------------------------

(pm.)brass_ui_MIDI

Ready-to-use MIDI-controllable brass instrument physical model with built-in UI.

Usage
brass_ui_MIDI : _

-----------------------------

(pm.)fluteModel

A simple generic flute instrument physical model without tone holes (pitch is changed by changing the length of the tube of the instrument).

Usage
fluteModel(tubeLength,mouthPosition,pressure) : _

Where:
- **tubeLength**: the length of the tube in meters
- **mouthPosition**: position of the mouth on the embouchure (0-1) (default: 0.5)
- **pressure**: the pressure of the air flow created by the virtual performer (0-1). This can also be any kind of signal that will directly injected in the mouthpiece (e.g., breath noise, etc.).

---

**pm.)fluteModel_ui**

Same as **fluteModel** but with a built-in UI. This function doesn’t implement a virtual “blower”, thus **pressure** remains an argument here.

**Usage**

```plaintext
fluteModel_ui(pressure) : _
```

Where:

- **pressure**: the pressure of the air flow created by the virtual performer (0-1). This can also be any kind of signal that will be directly injected in the mouthpiece (e.g., breath noise, etc.).

---

**pm.)flute_ui**

Ready-to-use flute physical model with built-in UI based on **fluteModel**.

**Usage**

```plaintext
flute_ui : _
```

---

**pm.)flute_ui_MIDI**

Ready-to-use MIDI-controllable flute physical model with built-in UI.

**Usage**

```plaintext
flute_ui_MIDI : _
```

---

**Exciters**

Various kind of excitation signal generators.
(pm.)impulseExcitation
Creates an impulse excitation of one sample.

Usage
```javascript
gate = button('gate');
impulseExcitation(gate) : chain;
```
Where:
- gate: a gate button

(pm.)strikeModel
Creates a filtered noise excitation.

Usage
```javascript
gate = button('gate');
strikeModel(LPcutoff,HPcutoff,sharpness,gain,gate) : chain;
```
Where:
- HPcutoff: highpass cutoff frequency
- LPcutoff: lowpass cutoff frequency
- sharpness: sharpness of the attack and release (0-1)
- gain: gain of the excitation
- gate: a gate button/trigger signal (0/1)

(pm.)strike
Strikes generator with controllable excitation position.

Usage
```javascript
gate = button('gate');
strike(exPos,sharpness,gain,gate) : chain;
```
Where:
- exPos: excitation position wiht 0: for max low freqs and 1: for max high freqs. So, on membrane for example, 0 would be the middle and 1 the edge
- sharpness: sharpness of the attack and release (0-1)
- gain: gain of the excitation
- gate: a gate button/trigger signal (0/1)
(pm.)pluckString

Creates a plucking excitation signal.

Usage

```javascript
trigger = button('gate');
pluckString(stringLength, cutoff, maxFreq, sharpness, trigger)
```

Where:

- **stringLength**: length of the string to pluck
- **cutoff**: cutoff ratio (1 for default)
- **maxFreq**: max frequency ratio (1 for default)
- **sharpness**: sharpness of the attack and release (1 for default)
- **gain**: gain of the excitation (0-1)
- **trigger**: trigger signal (1 for on, 0 for off)

---

(pm.)blower

A virtual blower creating a DC signal with some breath noise in it.

Usage

```javascript
blower(pressure, breathGain, breathCutoff) : _
```

Where:

- **pressure**: pressure (0-1)
- **breathGain**: breath noise gain (0-1) (recommended: 0.005)
- **breathCutoff**: breath cutoff frequency (Hz) (recommended: 2000)

---

(pm.)blower_ui

Same as **blower** but with a built-in UI.

Usage

```javascript
blower : somethingToBeBlown
```

---

**Modal Percussions**

High and low level functions for modal synthesis of percussion instruments.
Dirt-simple djembe modal physical model. Mode parameters are empirically calculated and don’t correspond to any measurements or 3D model. They kind of sound good though :).

**Usage**

djembeModel(freq)

Where:

- excitation: excitation signal
- freq: fundamental frequency of the bar

This model also implements a virtual “exciter”.

**Usage**
djembe(freq,strikePosition,strikeSharpness,gain,trigger)

Where:

- freq: fundamental frequency of the model
- strikePosition: strike position (0 for the middle of the membrane and 1 for the edge)
- strikeSharpness: sharpness of the strike (0-1, default: 0.5)
- gain: gain of the strike
- trigger: trigger signal (0: off, 1: on)

Simple MIDI controllable djembe physical model with built-in UI.

**Usage**
djembe_ui_MIDI

________________________
(pm.) marimbaBarModel

Generic marimba tone bar modal model.

This model was generated using mesh2faust from a 3D CAD model of a marimba tone bar (libraries/modalmodels/marimbaBar). The corresponding CAD model is that of a C2 tone bar (original fundamental frequency: ~65Hz). While marimbaBarModel allows to translate the harmonic content of the generated sound by providing a frequency (freq), mode transposition has limits and the model will sound less and less like a marimba tone bar as it diverges from C2. To make an accurate model of a marimba, we’d want to have an independent model for each bar...

This model contains 5 excitation positions going linearly from the center bottom to the center top of the bar. Obviously, a model with more excitation position could be regenerated using mesh2faust.

Usage

excitation : marimbaBarModel(freq,exPos,t60,t60DecayRatio,t60DecaySlope)

Where:
- excitation: excitation signal
- freq: fundamental frequency of the bar
- exPos: excitation position (0-4)
- t60: T60 in seconds (recommended value: 0.1)
- t60DecayRatio: T60 decay ratio (recommended value: 1)
- t60DecaySlope: T60 decay slope (recommended value: 5)

(pm.) marimbaResTube

Simple marimba resonance tube.

Usage

marimbaResTube(tubeLength,excitation)

Where:
- tubeLength: the length of the tube in meters
- excitation: the excitation signal (audio in)

(pm.) marimbaModel

Simple marimba physical model implementing a single tone bar connected to tube. This model is scalable and can be adapted to any size of bar/tube (see marimbaBarModel to know more about the limitations of this type of system).
Usage

excitation : marimbaModel(freq,exPos) : 

Where:

• freq: the frequency of the bar/tube couple
• exPos: excitation position (0-4)

(pm.)marimba

Simple marimba physical model implementing a single tone bar connected to tube. This model is scalable and can be adapted to any size of bar/tube (see marimbaBarModel to know more about the limitations of this type of system).

This function also implement a virtual exciter to drive the model.

Usage

excitation : marimba(freq,strikePosition,strikeCutoff,strikeSharpness,gain,trigger) : 

Where:

• excitation: the excitation signal
• freq: the frequency of the bar/tube couple
• strikePosition: strike position (0-4)
• strikeCutoff: cuttoff frequency of the strike generator (recommended: ~7000Hz)
• strikeSharpness: sharpness of the strike (recommended: ~0.25)
• gain: gain of the strike (0-1)
• trigger signal (0: off, 1: on)

(pm.)marimba_ui_MIDI

Simple MIDI controllable marimba physical model with built-in UI implementing a single tone bar connected to tube. This model is scalable and can be adapted to any size of bar/tube (see marimbaBarModel to know more about the limitations of this type of system).

Usage

marimba_ui_MIDI : 


churchBellModel

Generic church bell modal model generated by mesh2faust from libraries/modalmodels/churchBell.


Model height is 301 mm.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using mesh2faust.

Usage

excitation : churchBellModel(nModes,exPos,t60,t60DecayRatio,t60DecaySlope)

Where:

- excitation: the excitation signal
- nModes: number of synthesized modes (max: 50)
- exPos: excitation position (0-6)
- t60: T60 in seconds (recommended value: 0.1)
- t60DecayRatio: T60 decay ratio (recommended value: 1)
- t60DecaySlope: T60 decay slope (recommended value: 5)

churchBell

Generic church bell modal model.


Model height is 301 mm.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using mesh2faust.

This function also implement a virtual exciter to drive the model.

Usage

excitation : churchBell(strikePosition,strikeCutoff,strikeSharpness,gain,trigger) : _

Where:

- excitation: the excitation signal
- strikePosition: strike position (0-6)
- strikeCutoff: cuttoff frequency of the strike generator (recommended: ~7000Hz)
• **strikeSharpness**: sharpness of the strike (recommended: ~0.25)
• **gain**: gain of the strike (0-1)
• **trigger** signal (0: off, 1: on)

---

(pm.) **churchBell_ui**

Church bell physical model based on `churchBell` with built-in UI.

**Usage**

```plaintext```

churchBell_ui : _
```

---

(pm.) **englishBellModel**

English church bell modal model generated by `mesh2faust` from `libraries/modalmodels/englishBell`.


Model height is 1 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using `mesh2faust`.

**Usage**

```plaintext```

excitation : englishBellModel(nModes,exPos,t60,t60DecayRatio,t60DecaySlope)
```

Where:

• **excitation**: the excitation signal
• **nModes**: number of synthesized modes (max: 50)
• **exPos**: excitation position (0-6)
• **t60**: T60 in seconds (recommended value: 0.1)
• **t60DecayRatio**: T60 decay ratio (recommended value: 1)
• **t60DecaySlope**: T60 decay slope (recommended value: 5)

---

(pm.) **englishBell**

English church bell modal model.

Model height is 1 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using mesh2faust.

This function also implement a virtual exciter to drive the model.

Usage

```latex
excitation : englishBell(strikePosition,strikeCutoff,strikeSharpness,gain,trigger) :
```

Where:

- `excitation`: the excitation signal
- `strikePosition`: strike position (0-6)
- `strikeCutoff`: cutoff frequency of the strike generator (recommended: ~7000Hz)
- `strikeSharpness`: sharpness of the strike (recommended: ~0.25)
- `gain`: gain of the strike (0-1)
- `trigger`: signal (0: off, 1: on)

---

(pm.)englishBell_ui

English church bell physical model based on englishBell with built-in UI.

Usage

```latex
englishBell_ui :
```

---

(pm.)frenchBellModel

French church bell modal model generated by mesh2faust from libraries/modalmodels/frenchBell.


Model height is 1 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using mesh2faust.

Usage

```latex
excitation : frenchBellModel(nModes,exPos,t60,t60DecayRatio,t60DecaySlope)
```
Where:

- **excitation**: the excitation signal
- **nModes**: number of synthesized modes (max: 50)
- **exPos**: excitation position (0-6)
- **t60**: T60 in seconds (recommended value: 0.1)
- **t60DecayRatio**: T60 decay ratio (recommended value: 1)
- **t60DecaySlope**: T60 decay slope (recommended value: 5)

---

**(pm.)frenchBell**

French church bell modal model.


Model height is 1 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using `mesh2faust`.

This function also implement a virtual exciter to drive the model.

**Usage**

```python
excitation : frenchBell(strikePosition, strikeCutoff, strikeSharpness, gain, trigger) :
```

Where:

- **excitation**: the excitation signal
- **strikePosition**: strike position (0-6)
- **strikeCutoff**: cutoff frequency of the strike generator (recommended: ~7000Hz)
- **strikeSharpness**: sharpness of the strike (recommended: ~0.25)
- **gain**: gain of the strike (0-1)
- **trigger** signal (0: off, 1: on)

---

**(pm.)frenchBell_ui**

French church bell physical model based on `frenchBell` with built-in UI.

**Usage**

```python
frenchBell_ui :
```
German church bell modal model generated by mesh2faust from libraries/modalmodels/germanBell.


Model height is 1 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using mesh2faust.

Usage

excitation : germanBellModel(nModes,exPos,t60,t60DecayRatio,t60DecaySlope)

Where:

- excitation: the excitation signal
- nModes: number of synthesized modes (max: 50)
- exPos: excitation position (0-6)
- t60: T60 in seconds (recommended value: 0.1)
- t60DecayRatio: T60 decay ratio (recommended value: 1)
- t60DecaySlope: T60 decay slope (recommended value: 5)

German church bell modal model.


Model height is 1 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using mesh2faust.

This function also implement a virtual exciter to drive the model.

Usage

excitation : germanBell(strikePosition,strikeCutoff,strikeSharpness,gain,trigger) : _

Where:

- excitation: the excitation signal
- strikePosition: strike position (0-6)
- **strikeCutoff**: cutoff frequency of the strike generator (recommended: ~7000Hz)
- **strikeSharpness**: sharpness of the strike (recommended: ~0.25)
- **gain**: gain of the strike (0-1)
- **trigger** signal (0: off, 1: on)

---

**(pm.) germanBell_ui**

German church bell physical model based on germanBell with built-in UI.

**Usage**

`germanBell_ui : _`

---

**(pm.) russianBellModel**

Russian church bell modal model generated by `mesh2faust` from `libraries/modalmodels/russianBell`.


Model height is 2 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using `mesh2faust`.

**Usage**

`excitation : russianBellModel(nModes,exPos,t60,t60DecayRatio,t60DecaySlope)`

Where:

- **excitation**: the excitation signal
- **nModes**: number of synthesized modes (max: 50)
- **exPos**: excitation position (0-6)
- **t60**: T60 in seconds (recommended value: 0.1)
- **t60DecayRatio**: T60 decay ratio (recommended value: 1)
- **t60DecaySlope**: T60 decay slope (recommended value: 5)

---

**(pm.) russianBell**

Russian church bell modal model.

Model height is 2 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using mesh2faust.

This function also implement a virtual exciter to drive the model.

**Usage**

```plaintext
excitation : russianBell(strikePosition, strikeCutoff, strikeSharpness, gain, trigger) : _
```

Where:

- **excitation**: the excitation signal
- **strikePosition**: strike position (0-6)
- **strikeCutoff**: cutoff frequency of the strike generator (recommended: ~7000Hz)
- **strikeSharpness**: sharpness of the strike (recommended: ~0.25)
- **gain**: gain of the strike (0-1)
- **trigger**: signal (0: off, 1: on)

---

*(pm.*)russianBell_ui

Russian church bell physical model based on russianBell with built-in UI.

**Usage**

```plaintext
russianBell_ui : _
```

---

*(pm.*)standardBellModel

Standard church bell modal model generated by mesh2faust from libraries/modalmodels/standardBell.


Model height is 1.8 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using mesh2faust.
Usage

excitation : standardBellModel(nModes,exPos,t60,t60DecayRatio,t60DecaySlope)

Where:

- excitation: the excitation signal
- nModes: number of synthesized modes (max: 50)
- exPos: excitation position (0-6)
- t60: T60 in seconds (recommended value: 0.1)
- t60DecayRatio: T60 decay ratio (recommended value: 1)
- t60DecaySlope: T60 decay slope (recommended value: 5)

(pm.) standardBell

Standard church bell modal model.


Model height is 1.8 m.

This model contains 7 excitation positions going linearly from the bottom to the top of the bell. Obviously, a model with more excitation position could be regenerated using mesh2faust.

This function also implement a virtual exciter to drive the model.

Usage

excitation : standardBell(strikePosition,strikeCutoff,strikeSharpness,gain,trigger) :

Where:

- excitation: the excitation signal
- strikePosition: strike position (0-6)
- strikeCutoff: cuttoff frequency of the strike generator (recommended: ~7000Hz)
- strikeSharpness: shaarpness of the strike (recommended: ~0.25)
- gain: gain of the strike (0-1)
- trigger signal (0: off, 1: on)

(pm.) standardBell_ui

Standard church bell physical model based on standardBell with built-in UI.
Usage
standardBell_ui : _

Vocal Synthesis
Vocal synthesizer functions (source/filter, fof, etc.).

(formantValues)
Formant data values.
The formant data used here come from the CSOUND manual http://www.csounds.com/manual/html/.

Usage
ba.take(j+1,formantValues.f(i)) : _
ba.take(j+1,formantValues.g(i)) : _
ba.take(j+1,formantValues.bw(i)) : _

Where:
• i: formant number
• j: (voiceType*nFormants)+vowel
• voiceType: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
• vowel: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)

(voiceGender)
Calculate the gender for the provided voiceType value. (0: male, 1: female)

Usage
voiceGender(voiceType) : _

Where:
• voiceType: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)

(skirtWidthMultiplier)
Calculates value to multiply bandwidth to obtain skirtwidth for a Fof filter.
Usage

skirtWidthMultiplier(vowel,freq,gender) : _

Where:

- **vowel**: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- **freq**: the fundamental frequency of the excitation signal
- **gender**: gender of the voice used in the fof filter (0: male, 1: female)

(pm.) autobendFreq

Autobends the center frequencies of formants 1 and 2 based on the fundamental frequency of the excitation signal and leaves all other formant frequencies unchanged. Ported from chant-lib. Reference: https://ccrma.stanford.edu/~rmichon/chantLib/.

Usage

_: autobendFreq(n,freq,voiceType) : _

Where:

- **n**: formant index
- **freq**: the fundamental frequency of the excitation signal
- **voiceType**: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- **input** is the center frequency of the corresponding formant

(pm.) vocalEffort

Changes the gains of the formants based on the fundamental frequency of the excitation signal. Higher formants are reinforced for higher fundamental frequencies. Ported from chant-lib. Reference: https://ccrma.stanford.edu/~rmichon/chantLib/.

Usage

_: vocalEffort(freq,gender) : _

Where:

- **freq**: the fundamental frequency of the excitation signal
- **gender**: the gender of the voice type (0: male, 1: female)
- **input** is the linear amplitude of the formant
(pm.)fof


Usage

_ : fof(fc,bw,a,g) : _

Where:

- \( fc \): formant center frequency,
- \( bw \): formant bandwidth (Hz),
- \( sw \): formant skirtwidth (Hz)
- \( g \): linear scale factor (\( g = 1 \) gives 0dB amplitude response at \( fc \))
- input is an impulse signal to excite filter

(fofSH)

FOF with sample and hold used on \( bw \) and a parameter used in the filter-cycling FOF function fofCycle. Reference: https://ccrma.stanford.edu/~mjolsen/pdfs/smc2016_MOlsenFOF.pdf.

Usage

_ : fofSH(fc,bw,a,g) : _

Where: all parameters same as for fof

(fofCycle)

FOF implementation where time-varying filter parameter noise is mitigated by using a cycle of \( n \) sample and hold FOF filters. Reference: https://ccrma.stanford.edu/~mjolsen/pdfs/smc2016_MOlsenFOF.pdf.

Usage

_ : fofCycle(fc,bw,a,g,n) : _

Where:

- \( n \): the number of FOF filters to cycle through
- all other parameters are same as for fof
(pm.)fofSmooth

FOF implementation where time-varying filter parameter noise is mitigated by lowpass filtering the filter parameters \( bw \) and \( a \) with smooth.

Usage

\[ \_ : \text{fofSmooth}(fc,bw,sw,g,\tau) : \_ \]

Where:

- \( \tau \): the desired smoothing time constant in seconds
- all other parameters are same as for fof

____________________________

(pm.)formantFilterFofCycle

Formant filter based on a single FOF filter. Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. A cycle of \( n \) fof filters with sample-and-hold is used so that the fof filter parameters can be varied in realtime. This technique is more robust but more computationally expensive than formantFilterFofSmooth. Voice type can be selected but must correspond to the frequency range of the provided source to be realistic.

Usage

\[ \_ : \text{formantFilterFofCycle}(\text{voiceType},\text{vowel},\text{nFormants},i,freq) : \_ \]

Where:

- \( \text{voiceType} \): the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- \( \text{vowel} \): the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- \( \text{nFormants} \): number of formant regions in frequency domain, typically 5
- \( i \): formant number (i.e. 0 - 4) used to index formant data value arrays
- \( \text{freq} \): fundamental frequency of excitation signal. Used to calculate rise time of envelope

____________________________

(pm.)formantFilterFofSmooth

Formant filter based on a single FOF filter. Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Fof filter parameters are lowpass filtered to mitigate possible noise from varying them in realtime. Voice type can be selected but must correspond to the frequency range of the provided source to be realistic.
Usage

formantFilterFofSmooth(voiceType, vowel, nFormants, i, freq)

Where:

- **voiceType**: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- **vowel**: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- **nFormants**: number of formant regions in frequency domain, typically 5
- **i**: formant number (i.e. 1 - 5) used to index formant data value arrays
- **freq**: fundamental frequency of excitation signal. Used to calculate rise time of envelope

(formantFilterBP

Formant filter based on a single resonant bandpass filter. Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Voice type can be selected but must correspond to the frequency range of the provided source to be realistic.

Usage

formantFilterBP(voiceType, vowel, nFormants, i, freq)

Where:

- **voiceType**: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- **vowel**: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- **nFormants**: number of formant regions in frequency domain, typically 5
- **i**: formant index used to index formant data value arrays
- **freq**: fundamental frequency of excitation signal.

(formantFilterbank

Formant filterbank which can use different types of filterbank functions and different excitation signals. Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Voice type can be selected but must correspond to the frequency range of the provided source to be realistic.

Usage

formantFilterbank(voiceType, vowel, formantGen, freq)

Where:
- **voiceType**: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- **vowel**: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- **formantGen**: the specific formant filterbank function (i.e. FormantFilterbankBP, FormantFilterbankFof,...)
- **freq**: fundamental frequency of excitation signal. Needed for FOF version to calculate rise time of envelope

---

**(pm.)formantFilterbankFofCycle**

Formant filterbank based on a bank of fof filters. Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Voice type can be selected but must correspond to the frequency range of the provided source to be realistic.

**Usage**

```
_ : formantFilterbankFofCycle(voiceType, vowel, freq) : _
```

Where:

- **voiceType**: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- **vowel**: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- **freq**: the fundamental frequency of the excitation signal. Needed to calculate the skirtwidth of the FOF envelopes and for the autobendFreq and vocalEffort functions

---

**(pm.)formantFilterbankFofSmooth**

Formant filterbank based on a bank of fof filters. Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Voice type can be selected but must correspond to the frequency range of the provided source to be realistic.

**Usage**

```
_ : formantFilterbankFofSmooth(voiceType, vowel, freq) : _
```

Where:

- **voiceType**: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- **vowel**: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
• **freq**: the fundamental frequency of the excitation signal. Needed to calculate the skirtwidth of the FOF envelopes and for the autobendFreq and vocalEffort functions

---

**formantFilterbankBP**

Formant filterbank based on a bank of resonant bandpass filters. Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Voice type can be selected but must correspond to the frequency range of the provided source to be realistic.

**Usage**

```plaintext
_: formantFilterbankBP(voiceType,vowel) : _
```

Where:

- **voiceType**: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- **vowel**: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- **freq**: the fundamental frequency of the excitation signal. Needed for the autobendFreq and vocalEffort functions

---

**SFFormantModel**

Simple formant/vocal synthesizer based on a source/filter model. The source and filterbank must be specified by the user. The filterbank must take the same input parameters as `formantFilterbank` (**BP/FofCycle/FofSmooth**). Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Voice type can be selected but must correspond to the frequency range of the synthesized voice to be realistic.

**Usage**

```plaintext
SFFormantModel(voiceType,vowel,exType,freq,gain,source,filterbank,isFof) : _
```

Where:

- **voiceType**: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- **vowel**: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- **exType**: voice vs. fricative sound ratio (0-1 where 1 is 100% fricative)
- **freq**: the fundamental frequency of the source signal
- **gain**: linear gain multiplier to multiply the source by
- **isFof**: whether model is FOF based (0: no, 1: yes)
(pm.)SFFormantModelFofCycle

Simple formant/vocal synthesizer based on a source/filter model. The source is just a periodic impulse and the “filter” is a bank of FOF filters. Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Voice type can be selected but must correspond to the frequency range of the synthesized voice to be realistic. This model does not work with noise in the source signal so exType has been removed and model does not depend on SFFormantModel function.

Usage

SFFormantModelFofCycle(voiceType,vowel,freq,gain) : _

Where:

- `voiceType`: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- `vowel`: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- `freq`: the fundamental frequency of the source signal
- `gain`: linear gain multiplier to multiply the source by

(pm.)SFFormantModelFofSmooth

Simple formant/vocal synthesizer based on a source/filter model. The source is just a periodic impulse and the “filter” is a bank of FOF filters. Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Voice type can be selected but must correspond to the frequency range of the synthesized voice to be realistic.

Usage

SFFormantModelFofSmooth(voiceType,vowel,freq,gain) : _

Where:

- `voiceType`: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- `vowel`: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- `freq`: the fundamental frequency of the source signal
- `gain`: linear gain multiplier to multiply the source by

(pm.)SFFormantModelBP

Simple formant/vocal synthesizer based on a source/filter model. The source is just a sawtooth wave and the “filter” is a bank of resonant bandpass filters.  

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Formant parameters are linearly interpolated allowing to go smoothly from one vowel to another. Voice type can be selected but must correspond to the frequency range of the synthesized voice to be realistic.

The formant data used here come from the CSOUND manual http://www.csounds.com/manual/html/.

**Usage**

\[
\text{SFFormantModelBP(voiceType,vowel,exType,freq,gain)} : _ \\
\]

Where:

- **voiceType**: the voice type (0: alto, 1: bass, 2: countertenor, 3: soprano, 4: tenor)
- **vowel**: the vowel (0: a, 1: e, 2: i, 3: o, 4: u)
- **exType**: voice vs. fricative sound ratio (0-1 where 1 is 100% fricative)
- **freq**: the fundamental frequency of the source signal
- **gain**: linear gain multiplier to multiply the source by

________________________

\[
\text{(pm.)SFFormantModelFofCycle_ui} \\
\]

Ready-to-use source-filter vocal synthesizer with built-in user interface.

**Usage**

\[
\text{SFFormantModelFofCycle_ui} : _ \\
\]

________________________

\[
\text{(pm.)SFFormantModelFofSmooth_ui} \\
\]

Ready-to-use source-filter vocal synthesizer with built-in user interface.

**Usage**

\[
\text{SFFormantModelFofSmooth_ui} : _ \\
\]

________________________

\[
\text{SFFormantModelBP_ui} \\
\]

Ready-to-use source-filter vocal synthesizer with built-in user interface.

**Usage**

\[
\text{SFFormantModelBP_ui} : _ \\
\]
(pm.)SFFormantModelFofCycle_ui_MIDI
Ready-to-use MIDI-controllable source-filter vocal synthesizer.

Usage
SFFormantModelFofCycle_ui_MIDI : _

________________________________________________________

(pm.)SFFormantModelFofSmooth_ui_MIDI
Ready-to-use MIDI-controllable source-filter vocal synthesizer.

Usage
SFFormantModelFofSmooth_ui_MIDI : _

________________________________________________________

(pm.)SFFormantModelBP_ui_MIDI
Ready-to-use MIDI-controllable source-filter vocal synthesizer.

Usage
SFFormantModelBP_ui_MIDI : _

________________________________________________________

Misc Functions
Various miscellaneous functions.

(pm.)allpassNL
Bidirectional block adding nonlinearities in both directions in a chain. Nonlinearities are created by modulating the coefficients of a passive allpass filter by the signal it is processing.

Usage
chain(... : allpassNL(nonlinearity) : ...)

Where:
- nonlinearity: amount of nonlinearity to be added (0-1)
modalModel

// Implement multiple resonance modes using resonant bandpass filters.

Usage

_ : modalModel(n, freqs, t60s, gains) : _

Where:

• n: number of given modes
• freqs: list of filter center frequencies
• t60s: list of mode resonance durations (in seconds)
• gains: list of mode gains (0-1)

For example, to generate a model with 2 modes (440 Hz and 660 Hz, a fifth) where the higher one decays faster and is attenuated:

os.impulse : modalModel(2, (440, 660),
(0.5, 0.25),
(ba.db2linear(-1), ba.db2linear(-6)) : _

Further reading: Grumiaux et. al., 2017: Impulse-Response and CAD-Mod//
el-Based Physical Modeling in Faust

platform.lib

A library to handle platform specific code in Faust. Its official prefix is pl.

(pl.)SR

Current sampling rate (between 1Hz and 192000Hz). Constant during program execution.

(reducemaps.lib

A library to handle reduce/map kind of operation in Faust. Its official prefix is rm.
reduce

Fold-like high order function. Apply a binary operation on a block of consecutive samples of a signal. For example: reduce(max,128) will compute the maximum of each block of 128 samples. Please note that the resulting value, while produced continuously, will be constant for the duration of a block. A new value is only produced at the end of a block. Note also that blocks should be of at least one sample (n>0).

Usage
reduce(op, n, x)

reducemap

Like reduce but a foo function is applied to the result. From a mathematical point of view: reducemap(op,foo,n) is equivalent to reduce(op,n):foo but more efficient.

Usage
reducemap (op, foo, n, x)

reverbs.lib

A library of reverb effects. Its official prefix is re.

Schroeder Reverberators

jcrev

This artificial reverb takes a mono signal and outputs stereo (satrev) and quad (jcrev). They were implemented by John Chowning in the MUS10 computer-music language (descended from Music V by Max Mathews). They are Schroeder Reverberators, well tuned for their size. Nowadays, the more expensive freeverb is more commonly used (see the Faust examples directory).

jcrev reverb below was made from a listing of “RV”, dated April 14, 1972, which was recovered from an old SAIL DART backup tape. John Chowning thinks this might be the one that became the well known and often copied JCREV.

jcrev is a standard Faust function
Usage

_ : jcrev : _,_,_,_

(re.)satrev

This artificial reverberator takes a mono signal and outputs stereo (satrev) and quad (jcrev). They were implemented by John Chowning in the MUS10 computer-music language (descended from Music V by Max Mathews). They are Schroeder Reverberators, well tuned for their size. Nowadays, the more expensive freeverb is more commonly used (see the Faust examples directory).

satrev was made from a listing of “SATREV”, dated May 15, 1971, which was recovered from an old SAIL DART backup tape. John Chowning thinks this might be the one used on his often-heard brass canon sound examples, one of which can be found at https://ccrma.stanford.edu/~jos/wav/FM_BrassCanon2.wav.

Usage

_ : satrev : _,_

Feedback Delay Network (FDN) Reverberators

(re.)fdnrev0

Pure Feedback Delay Network Reverberator (generalized for easy scaling). fdnrev0 is a standard Faust function.

Usage

<1,2,4,...,N signals> <:
fdnrev0(MAXDELAY,delays,BBSO,freqs,durs,loopgainmax,nonl) :>
<1,2,4,...,N signals>

Where:

- N: 2, 4, 8, ... (power of 2)
- MAXDELAY: power of 2 at least as large as longest delay-line length
- delays: N delay lines, N a power of 2, lengths preferably coprime
- BBSO: odd positive integer = order of bandsplit desired at freqs
- freqs: NB-1 crossover frequencies separating desired frequency bands
- durs: NB decay times (t60) desired for the various bands
- loopgainmax: scalar gain between 0 and 1 used to “squelch” the reverb
- nonl: nonlinearity (0 to 0.999..., 0 being linear)
Reference
https://ccrma.stanford.edu/~jos/pasp/FDN_Reverberation.html

(re.)zita_rev_fdn
Internal 8x8 late-reverberation FDN used in the FOSS Linux reverb zita-rev1 by Fons Adriaensen fons@linuxaudio.org. This is an FDN reverb with allpass comb filters in each feedback delay in addition to the damping filters.

Usage
bus(8) : zita_rev_fdn(f1,f2,t60dc,t60m,fsmax) : bus(8)

Where:
- f1: crossover frequency (Hz) separating dc and midrange frequencies
- f2: frequency (Hz) above f1 where T60 = t60m/2 (see below)
- t60dc: desired decay time (t60) at frequency 0 (sec)
- t60m: desired decay time (t60) at midrange frequencies (sec)
- fsmax: maximum sampling rate to be used (Hz)

Reference

(re.)zita_rev1_stereo
Extend zita_rev_fdn to include zita_rev1 input/output mapping in stereo mode. zita_rev1_stereo is a standard Faust function.

Usage
_ ,_ : zita_rev1_stereo(rdel,f1,f2,t60dc,t60m,fsmax) : _ ,_

Where:
rdel = delay (in ms) before reverberation begins (e.g., 0 to ~100 ms) (remaining args and refs as for zita_rev_fdn above)

(re.)zita_rev1_ambi
Extend zita_rev_fdn to include zita_rev1 input/output mapping in “ambisonics mode”, as provided in the Linux C++ version.
Usage

_: zita_rev1_ambi(rgxyz, rdel, f1, f2, t60dc, t60m, fsmax) : _,_,_,_

Where:

rgxyz = relative gain of lanes 1,4,2 to lane 0 in output (e.g., -9 to 9) (remaining args and references as for zita_rev1_stereo above)

Freeverb

(re.) mono_freeverb

A simple Schroeder reverberator primarily developed by “Jezar at Dreampoint” that is extensively used in the free-software world. It uses four Schroeder allpasses in series and eight parallel Schroeder-Moorer filtered-feedback comb-filters for each audio channel, and is said to be especially well tuned.

mono_freeverb is a standard Faust function.

Usage

_: mono_freeverb(fb1, fb2, damp, spread) : _;

Where:

- fb1: coefficient of the lowpass comb filters (0-1)
- fb2: coefficient of the allpass comb filters (0-1)
- damp: damping of the lowpass comb filter (0-1)
- spread: spatial spread in number of samples (for stereo)

License

While this version is licensed LGPL (with exception) along with other GRAME library functions, the file freeverb.dsp in the examples directory of older Faust distributions, such as faust-0.9.85, was released under the BSD license, which is less restrictive.

(re.) stereo_freeverb

A simple Schroeder reverberator primarily developed by “Jezar at Dreampoint” that is extensively used in the free-software world. It uses four Schroeder allpasses in series and eight parallel Schroeder-Moorer filtered-feedback comb-filters for each audio channel, and is said to be especially well tuned.
Usage

\_\_ : stereo_freeverb(fb1, fb2, damp, spread) : \_\_;

Where:

- \textit{fb1}: coefficient of the lowpass comb filters (0-1)
- \textit{fb2}: coefficient of the allpass comb filters (0-1)
- \textit{damp}: damping of the lowpass comb filter (0-1)
- \textit{spread}: spatial spread in number of samples (for stereo)

\begin{lstlisting}[language=faust]
routes.lib
\end{lstlisting}

A library to handle signal routing in Faust. Its official prefix is \texttt{ro}.

**Functions Reference**

\texttt{(ro.)cross}

Cross \textit{n} signals: \((x_1, x_2, \ldots, x_n) \rightarrow (x_n, \ldots, x_2, x_1)\). \texttt{cross} is a standard Faust function.

**Usage**

\texttt{cross(n)}

\_\_ : \texttt{cross(3)} : \_\_;

Where:

- \textit{n}: number of signals (int, must be known at compile time)

**Note**

Special case: \texttt{cross2}:

\texttt{cross2 = \_\_, cross(2), \_};

\begin{lstlisting}[language=faust]
(ro.)crossnn
\end{lstlisting}

Cross two \texttt{bus(n)}s.

**Usage**

\_\_, \ldots : \texttt{crossmm(n)} : \_\_, \ldots;

Where:

- \textit{n}: the number of signals in the \texttt{bus}
(ro.)crossn1
Cross bus(n) and bus(1).

Usage
_,_,... : crossn1(n) : _,_,...
Where:
  • n: the number of signals in the first bus

(ro.)interleave
Interleave rowcol cables from column order to row order. input : x(0), x(1), x(2)
  ...., x(rowcol-1) output: x(0+0row), x(0+1row), x(0+2row), ...., x(1+0row),
  x(1+1row), x(1+2row), ...

Usage
_,_,_,_,_,_ : interleave(row,column) : _,_,_,_,_,_
Where:
  • row: the number of row (int, known at compile time)
  • column: the number of column (int, known at compile time)

(ro.)butterfly
Addition (first half) then subtraction (second half) of interleaved signals.

Usage
_,_,_,_ : butterfly(n) : _,_,_,_
Where:
  • n: size of the butterfly (n is int, even and known at compile time)

(ro.)hadamard
Hadamard matrix function of size \( n = 2^k \).
Usage

-_,_,_,_ : hadamard(n) : _,_,_,_
Where:
  • n: $2^k$, size of the matrix (int, must be known at compile time)

Note:
Implementation contributed by Remy Muller.

__________________________

(ro.)recursivize
Create a recursion from two arbitrary processors p and q.

Usage

-_,-,-,- : recursivize(p,q) : _,_
Where:
  • p: the forward arbitrary processor
  • q: the feedback arbitrary processor

__________________________

signals.lib
A library of basic elements to handle signals in Faust. Its official prefix is si.

Functions Reference

(si.)bus
n parallel cables. bus is a standard Faust function.

Usage

bus(n)
bus(4) : _,_,_,_
Where:
  • n: is an integer known at compile time that indicates the number of parallel cables.

__________________________
(si.)block
Block - terminate n signals. block is a standard Faust function.

Usage
`,`,... : block(n) : `,`,...
Where:
  • n: the number of signals to be blocked

---

(si.)interpolate
Linear interpolation between two signals.

Usage
`,` : interpolate(i) : _
Where:
  • i: interpolation control between 0 and 1 (0: first input; 1: second input)

---

(si.)smoo
Smoothing function based on smooth ideal to smooth UI signals (sliders, etc.) down. smoo is a standard Faust function.

Usage
hslider(...) : smoo;

---

(si.)polySmooth
A smoothing function based on smooth that doesn’t smooth when a trigger signal is given. This is very useful when making polyphonic synthesizer to make sure that the value of the parameter is the right one when the note is started.

Usage
hslider(...) : polySmooth(g,s,d) : _
Where:
  • g: the gate/trigger signal used when making polyphonic synths
  • s: the smoothness (see smooth)
• d: the number of samples to wait before the signal start being smoothed after g switched to 1

_________________________

(sia.)smoothAndH
A smoothing function based on smooth that holds its output signal when a trigger is sent to it. This feature is convenient when implementing polyphonic instruments to prevent some smoothed parameter to change when a note-off event is sent.

Usage

hslider(...) : smoothAndH(g,s) : _

Where:
• g: the hold signal (0 for hold, 1 for bypass)
• s: the smoothness (see smooth)

_________________________

(sia.)bsmooth
Block smooth linear interpolation during a block of samples.

Usage

hslider(...) : bsmooth : _

_________________________

(sia.)dot
Dot product for two vectors of size n.

Usage

_,_,_,_,_,_ : dot(n) : _

Where:
• n: size of the vectors (int, must be known at compile time)

_________________________

(sia.)smooth
Exponential smoothing by a unity-dec-gain one-pole lowpass. smooth is a standard Faust function.
Usage:
_ : smooth(tau2pole(tau)) : _
Where:
  • tau: desired smoothing time constant in seconds, or

hslider(...) : smooth(s) : _
Where:
  • s: smoothness between 0 and 1. s=0 for no smoothing, s=0.999 is “very smooth”, s>1 is unstable, and s=1 yields the zero signal for all inputs. The exponential time-constant is approximately 1/(1-s) samples, when s is close to (but less than) 1.

Reference:
https://ccrma.stanford.edu/~jos/mdft/Convolution_Example_2_ADSR.html

(si.)cbus
n parallel cables for complex signals. cbus is a standard Faust function.

Usage
cbus(n)
cbus(4) : (r0,i0), (r1,i1), (r2,i2), (r3,i3)
Where:
  • n: is an integer known at compile time that indicates the number of parallel cables.
  • each complex number is represented by two real signals as (real,imag)

(si.)cmul
multiply two complex signals pointwise. cmul is a standard Faust function.

Usage
(r1,i1) : cmul(r2,i2) : (_,_);
Where:
  • Each complex number is represented by two real signals as (real,imag), so
  • (r1,i1) = real and imaginary parts of signal 1
  • (r2,i2) = real and imaginary parts of signal 2
(si.)cconj
complex conjugation of a (complex) signal. cconj is a standard Faust function.

Usage

```
(r1,i1) : cconj : (_,_);
```

Where:
- Each complex number is represented by two real signals as (real,imag), so
- \((r1,i1)\) = real and imaginary parts of the input signal
- \((r1,-i1)\) = real and imaginary parts of the output signal

----------------------------------------

(si.)lag_ud
Lag filter with separate times for up and down.

Usage

```
_ : lag_ud(up, dn) : _;
```

----------------------------------------

(si.)rev
Reverse the input signal by blocks of \(N>0\) samples. rev(1) is the indentity function. rev(\(N\)) has a latency of \(N-1\) samples.

Usage

```
_ : rev(N) : _;
```

Where:
- \(N\): the block size

----------------------------------------

soundfiles.lib
A library to handle soundfiles in Faust. Its official prefix is so.
Functions Reference

(so.)loop

Play a soundfile in a loop taking into account its sampling rate loop is a standard Faust function.

Usage
loop(sf, part)

Where:
• sf: the soundfile
• part: the part in the soundfile list of sounds

(so.)loop_speed

Play a soundfile in a loop taking into account its sampling rate, with speed control loop_speed is a standard Faust function.

Usage
loop_speed(sf, part, speed)

Where:
• sf: the soundfile
• part: the part in the soundfile list of sounds
• speed: the speed between 0 and n

(so.)loop_speed_level

Play a soundfile in a loop taking into account its sampling rate, with speed and level controls loop_speed_level is a standard Faust function.

Usage
loop_speed_level(sf, part, speed, level)

Where:
• sf: the soundfile
• part: the part in the soundfile list of sounds
• speed: the speed between 0 and n
• level: the volume between 0 and n
spats.lib

This library contains a collection of tools for sound spatialization. Its official prefix is sp.

(sp.)panner
A simple linear stereo panner. panner is a standard Faust function.

Usage

_ : panner(g) : _,_

Where:

• g: the panning (0-1)

(sp.)spat
GMEM SPAT: n-outputs spatializer. spat is a standard Faust function.

Usage

_ : spat(n,r,d) : _,_,...

Where:

• n: number of outputs
• r: rotation (between 0 et 1)
• d: distance of the source (between 0 et 1)

(sp.)stereoize
Transform an arbitrary processor p into a stereo processor with 2 inputs and 2 outputs.

Usage

_,_ : stereoize(p) : _,_

Where:

• p: the arbitrary processor
synths.lib

This library contains a collection of synthesizers. Its official prefix is **sy**.

*(sy.)* **popFilterPerc**

A simple percussion instrument based on a “popped” resonant bandpass filter. **popFilterPerc** is a standard Faust function.

Usage

```faust
popFilterDrum(freq,q,gate) : _;
```

Where:

- **freq**: the resonance frequency of the instrument
- **q**: the q of the res filter (typically, 5 is a good value)
- **gate**: the trigger signal (0 or 1)

*(sy.)* **dubDub**

A simple synth based on a sawtooth wave filtered by a resonant lowpass. **dubDub** is a standard Faust function.

Usage

```faust
dubDub(freq,ctFreq,q,gate) : _;
```

Where:

- **freq**: frequency of the sawtooth
- **ctFreq**: cutoff frequency of the filter
- **q**: Q of the filter
- **gate**: the trigger signal (0 or 1)

*(sy.)* **sawTrombone**

A simple trombone based on a lowpassed sawtooth wave. **sawTrombone** is a standard Faust function.

Usage

```faust
sawTrombone(att,freq,gain,gate) : _
```

Where:

- **att**: exponential attack duration in s (typically 0.01)
- **freq**: the frequency
• **gain**: the gain (0-1)
• **gate**: the gate (0 or 1)

---

**(combString)**

Simplest string physical model ever based on a comb filter. **combString** is a standard Faust function.

**Usage**

```faust
combString(freq, res, gate) : _;
```

Where:

• **freq**: the frequency of the string
• **res**: string T60 (resonance time) in second
• **gate**: trigger signal (0 or 1)

---

**additiveDrum**

A simple drum using additive synthesis. **additiveDrum** is a standard Faust function.

**Usage**

```faust
additiveDrum(freq, freqRatio, gain, harmDec, att, rel, gate) : _
```

Where:

• **freq**: the resonance frequency of the drum
• **freqRatio**: a list of ratio to choose the frequency of the mode in function of **freq** e.g.(1 1.2 1.5 ...). The first element should always be one (fundamental).
• **gain**: the gain of each mode as a list (1 0.9 0.8 ...). The first element is the gain of the fundamental.
• **harmDec**: harmonic decay ratio (0-1): configure the speed at which higher modes decay compare to lower modes.
• **att**: attack duration in second
• **rel**: release duration in second
• **gate**: trigger signal (0 or 1)

---

**fm**

An FM synthesizer with an arbitrary number of modulators connected as a sequence. **fm** is a standard Faust function.
Usage

freqs = (300,400,...);
indices = (20,...);
fm(freqs,indices) : _

Where:
  • freqs: a list of frequencies where the first one is the frequency of the
carrier and the others, the frequency of the modulator(s)
  • indices: the indices of modulation (Nfreqs-1)

vaeffects.lib

A library of virtual analog filter effects. Its official prefix is ve.

Moog Filters

(ve.)moog_vcf

Moog “Voltage Controlled Filter” (VCF) in “analog” form. Moog VCF implemented using the same logical block diagram as the classic analog circuit. As such, it neglects the one-sample delay associated with the feedback path around the four one-poles. This extra delay alters the response, especially at high frequencies (see reference [1] for details). See moog_vcf_2b below for a more accurate implementation.

Usage

moog_vcf(res,fr)

Where:
  • res: normalized amount of corner-resonance between 0 and 1 (0 is no
resonance, 1 is maximum)
  • fr: corner-resonance frequency in Hz (less than SR/6.3 or so)

References

  • https://ccrma.stanford.edu/~stilti/papers/moogvcf.pdf
  • https://ccrma.stanford.edu/~jos/pasp/vegf.html

(ve.)moog_vcf_2b[n]

Moog “Voltage Controlled Filter” (VCF) as two biquads. Implementation of the ideal Moog VCF transfer function factored into second-order sections. As
a result, it is more accurate than moog_vcf above, but its coefficient formulas are more complex when one or both parameters are varied. Here, res is the fourth root of that in moog_vcf, so, as the sampling rate approaches infinity, moog_vcf(res,fr) becomes equivalent to moog_vcf_2b[n](res^4,fr) (when res and fr are constant). moog_vcf_2b uses two direct-form biquads (tf2). moog_vcf_2bn uses two protected normalized-ladder biquads (tf2np).

Usage
moog_vcf_2b(res,fr)
moog_vcf_2bn(res,fr)

Where:
- res: normalized amount of corner-resonance between 0 and 1 (0 is min resonance, 1 is maximum)
- fr: corner-resonance frequency in Hz

(ve.)moogLadder
Virtual analog model of the 4th-order Moog Ladder, which is arguably the most well-known ladder filter in analog synthesizers. Several 1st-order filters are cascaded in series. Feedback is then used, in part, to control the cut-off frequency and the resonance.

This filter was implemented in Faust by Eric Tarr during the 2019 Embedded DSP With Faust Workshop.

References
- https://www.willpirkle.com/706-2/

Usage
_ : moogLadder(normFreq,Q) : _

Where:
- normFreq: normalized frequency (0-1)
- Q: q

(ve.)moogHalfLadder
Virtual analog model of the 2nd-order Moog Half Ladder (simplified version of (ve.)moogLadder). Several 1st-order filters are cascaded in series. Feedback is then used, in part, to control the cut-off frequency and the resonance.
This filter was implemented in Faust by Eric Tarr during the 2019 Embedded DSP With Faust Workshop.

References
- https://www.willpirkle.com/app-notes/virtual-analog-moog-half-ladder-filter

Usage
```
_ : moogHalfLadder(normFreq,Q) : _
```
Where:
- `normFreq`: normalized frequency (0-1)
- `Q`: `q`

...(ve.)diodeLadder

4th order virtual analog diode ladder filter. In addition to the individual states used within each independent 1st-order filter, there are also additional feedback paths found in the block diagram. These feedback paths are labeled as connecting states. Rather than separately storing these connecting states in the Faust implementation, they are simply implicitly calculated by tracing back to the other states (s1,s2,s3,s4) each recursive step.

This filter was implemented in Faust by Eric Tarr during the 2019 Embedded DSP With Faust Workshop.

References

Usage
```
_ : diodeLadder(normFreq,Q) : _
```
Where:
- `normFreq`: normalized frequency (0-1)
- `Q`: `q`
**Korg 35 Filters**

The following filters are virtual analog models of the Korg 35 low-pass filter and high-pass filter found in the MS-10 and MS-20 synthesizers. The virtual analog models for the LPF and HPF are different, making these filters more interesting than simply tapping different states of the same circuit.

These filters were implemented in Faust by Eric Tarr during the 2019 Embedded DSP With Faust Workshop.

**Filter history:**

https://secretlifeofsynthesizers.com/the-korg-35-filter/

(ve.)korg35LPF

Virtual analog models of the Korg 35 low-pass filter found in the MS-10 and MS-20 synthesizers.

**Usage**

_ : korg35LPF(normFreq,Q) : _

Where:

- normFreq: normalized frequency (0-1)
- Q: q

________________________________________________________________________

(ve.)korg35HPF

Virtual analog models of the Korg 35 high-pass filter found in the MS-10 and MS-20 synthesizers.

**Usage**

_ : korg35HPF(normFreq,Q) : _

Where:

- normFreq: normalized frequency (0-1)
- Q: q

________________________________________________________________________

**Oberheim Filters**

The following filter (4 types) is an implementation of the virtual analog model described in Section 7.2 of the Will Pirkle book, *Designing Software Synthesizer Plug-ins in C++*. It is based on the block diagram in Figure 7.5.
The Oberheim filter is a state-variable filter with soft-clipping distortion within the circuit.

In many VA filters, distortion is accomplished using the “tanh” function. For this Faust implementation, that distortion function was replaced with the (ef.)cubicnl function.

(ve.)oberheim
Generic multi-outputs Oberheim filter (see description above).

Usage
_ : oberheim(normFreq,Q) : _,_,_,_  
Where:
• normFreq: normalized frequency (0-1)
• Q: q

(ve.)oberheimBSF
Band-Stop Oberheim filter (see description above).

Usage
_ : oberheimBSF(normFreq,Q) : _  
Where:
• normFreq: normalized frequency (0-1)
• Q: q

(ve.)oberheimBPF
Band-Pass Oberheim filter (see description above).

Usage
_ : oberheimBPF(normFreq,Q) : _  
Where:
• normFreq: normalized frequency (0-1)
• Q: q

233
(ve.) oberheimHPF
High-Pass Oberheim filter (see description above).

Usage
_ : oberheimHPF(normFreq,Q) : _

Where:
- normFreq: normalized frequency (0-1)
- Q: q

(ve.) oberheimLPF
Low-Pass Oberheim filter (see description above).

Usage
_ : oberheimLPF(normFreq,Q) : _

Where:
- normFreq: normalized frequency (0-1)
- Q: q

Sallen Key Filters
The following filters were implemented based on VA models of synthesizer filters.
The modeling approach is based on a Topology Preserving Transform (TPT) to resolve the delay-free feedback loop in the corresponding analog filters.
The primary processing block used to build other filters (Moog, Korg, etc.) is based on a 1st-order Sallen-Key filter.
The filters included in this script are 1st-order LPF/HPF and 2nd-order state-variable filters capable of LPF, HPF, and BPF.

Resources:
• Description and diagrams of 1st- and 2nd-order TPT filters: https://www.willpirkle.com/706-2/

(ve.)sallenKeyOnePole
Sallen-Key generic One Pole filter (see description above).

For the Faust implementation of this filter, recursion (\texttt{letrec}) is used for storing filter “states”. The output (e.g. \texttt{y}) is calculated by using the input signal and the previous states of the filter. During the current recursive step, the states of the filter (e.g. \texttt{s}) for the next step are also calculated. Admittedly, this is not an efficient way to implement a filter because it requires independently calculating the output and each state during each recursive step. However, it works as a way to store and use “states” within the constraints of Faust.

(ve.)sallenKeyOnePoleLPF
Sallen-Key One Pole lowpass filter (see description above).

\textbf{Usage}

\_ : sallenKeyOnePoleLPF(normFreq) : \\_

Where:

• \textit{normFreq}: normalized frequency (0-1)

(ve.)sallenKeyOnePoleHPF
Sallen-Key One Pole Highpass filter (see description above). The dry input signal is routed in parallel to the output. The LPF’d signal is subtracted from the input so that the HPF remains.

\textbf{Usage}

\_ : sallenKeyOnePoleHPF(normFreq) : \\_

Where:

• \textit{normFreq}: normalized frequency (0-1)

(ve.)sallenKey2ndOrder
Sallen-Key generic multi-outputs 2nd order filter.

This is a 2nd-order Sallen-Key state-variable filter. The idea is that by “tapping” into different points in the circuit, different filters (LPF,BPF,HPF) can be achieved. See Figure 4.6 of https://www.willpirkle.com/706-2/
This is also a good example of the next step for generalizing the Faust programming approach used for all these VA filters. In this case, there are three things to calculate each recursive step \((y,s1,s2)\). For each thing, the circuit is only calculated up to that point.

Comparing the LPF to BPF, the output signal \((y)\) is calculated similarly. Except, the output of the BPF stops earlier in the circuit. Similarly, the states \((s1\) and \(s2)\) only differ in that \(s2\) includes a couple more terms beyond what is used for \(s1\).

**Usage**

\[
_ : \text{sallenKey2ndOrder}(\text{normFreq}, Q) : _ , _ , _
\]

Where:

- \text{normFreq}: normalized frequency (0-1)
- \text{Q}: q

\[
(\text{ve.})\text{sallenKey2ndOrderLPF}
\]

Sallen-Key 2nd order lowpass filter (see description above).

**Usage**

\[
_ : \text{sallenKey2ndOrderLPF}(\text{normFreq}, Q) : _
\]

Where:

- \text{normFreq}: normalized frequency (0-1)
- \text{Q}: q

\[
(\text{ve.})\text{sallenKey2ndOrderBPF}
\]

Sallen-Key 2nd order bandpass filter (see description above).

**Usage**

\[
_ : \text{sallenKey2ndOrderBPF}(\text{normFreq}, Q) : _
\]

Where:

- \text{normFreq}: normalized frequency (0-1)
- \text{Q}: q
(ve.) **sallenKey2ndOrderHPF**

Sallen-Key 2nd order highpass filter (see description above).

**Usage**

```
_ : sallenKey2ndOrderHPF(normFreq,Q) :_
```

Where:

- `normFreq`: normalized frequency (0-1)
- `Q`: q

---

**Effects**

(ve.) **wah4**

Wah effect, 4th order. `wah4` is a standard Faust function.

**Usage**

```
_ : wah4(fr) :_
```

Where:

- `fr`: resonance frequency in Hz

**Reference**

https://ccrma.stanford.edu/~jos/pasp/vegf.html

---

(ve.) **autowah**

Auto-wah effect. `autowah` is a standard Faust function.

**Usage**

```
_ : autowah(level) :_
```

Where:

- `level`: amount of effect desired (0 to 1).

---

(ve.) **crybaby**

Digitized CryBaby wah pedal. `crybaby` is a standard Faust function.
Usage

_ : crybaby(wah) : _

Where:

- **wah**: “pedal angle” from 0 to 1

Reference

https://ccrma.stanford.edu/~jos/pasp/vegf.html

(ve.)vocoder

A very simple vocoder where the spectrum of the modulation signal is analyzed using a filter bank. **vocoder** is a standard Faust function.

Usage

_ : vocoder(nBands,att,rel,BWRatio,source,excitation) : _

Where:

- **nBands**: Number of vocoder bands
- **att**: Attack time in seconds
- **rel**: Release time in seconds
- **BWRatio**: Coefficient to adjust the bandwidth of each band (0.1 - 2)
- **source**: Modulation signal
- **excitation**: Excitation/Carrier signal

version.lib

Semantic versioning for the Faust libraries. Its official prefix is v1.

(v1.)version

Return the version number of the Faust standard libraries.

Usage

version : _,_,_
webaudio.lib

An implementation of the web audio API filters. Its official prefix is `wa`.

`(wa.)lowpass2`

Standard second-order resonant lowpass filter with 12dB/octave rolloff. Frequencies below the cutoff pass through; frequencies above it are attenuated.

Usage

`_: lowpass2(f0, Q, dtune) _:`

Where:

- `f0`: cutoff frequency in Hz
- `Q`: the quality factor
- `dtune`: detuning of the frequency in cents

Reference

https://searchfox.org/mozilla-central/source/dom/media/webaudio/blink/Biquad.cpp#98

__________________________

`(wa.)highpass2`

Standard second-order resonant highpass filter with 12dB/octave rolloff. Frequencies below the cutoff are attenuated; frequencies above it pass through.

Usage

`_: highpass2(f0, Q, dtune) _:`

Where:

- `f0`: cutoff frequency in Hz
- `Q`: the quality factor
- `dtune`: detuning of the frequency in cents

Reference

https://searchfox.org/mozilla-central/source/dom/media/webaudio/blink/Biquad.cpp#127

__________________________
(wa.)bandpass2

Standard second-order bandpass filter. Frequencies outside the given range of frequencies are attenuated; the frequencies inside it pass through.

Usage

`_: bandpass2(f0, Q, dtune) :_

Where:

- `f0`: cutoff frequency in Hz
- `Q`: the quality factor
- `dtune`: detuning of the frequency in cents

Reference

https://searchfox.org/mozilla-central/source/dom/media/webaudio/blink/Biquad.cpp#334

-------------

(wa.)notch2

Standard notch filter, also called a band-stop or band-rejection filter. It is the opposite of a bandpass filter: frequencies outside the given range of frequencies pass through, frequencies inside it are attenuated.

Usage

`_: notch2(f0, Q, dtune) :_

Where:

- `f0`: cutoff frequency in Hz
- `Q`: the quality factor
- `dtune`: detuning of the frequency in cents

Reference

https://searchfox.org/mozilla-central/source/dom/media/webaudio/blink/Biquad.cpp#301

-------------

(wa.)allpass2

Standard second-order allpass filter. It lets all frequencies through, but changes the phase-relationship between the various frequencies.
Usage

_: allpass2(f0, Q, dtune) :_

Where:

- \( f_0 \): cutoff frequency in Hz
- \( Q \): the quality factor
- \( dtune \): detuning of the frequency in cents

Reference

https://searchfox.org/mozilla-central/source/dom/media/webaudio/blink/
Biquad.cpp#268

_____________________________

_(wa.)peaking2_

Frequencies inside the range get a boost or an attenuation; frequencies outside it are unchanged.

Usage

_: peaking2(f0, gain, Q, dtune) :_

Where:

- \( f_0 \): cutoff frequency in Hz
- \( gain \): the gain in dB
- \( Q \): the quality factor
- \( dtune \): detuning of the frequency in cents

Reference

https://searchfox.org/mozilla-central/source/dom/media/webaudio/blink/
Biquad.cpp#233

_____________________________

_(wa.)lowshelf2_

Standard second-order lowshelf filter. Frequencies lower than the frequency get a boost, or an attenuation, frequencies over it are unchanged.

_: lowshelf2(f0, gain, dtune) :_

Where:

- \( f_0 \): cutoff frequency in Hz
- \( gain \): the gain in dB
- \( dtune \): detuning of the frequency in cents
Standard second-order highshelf filter. Frequencies higher than the frequency get a boost or an attenuation, frequencies lower than it are unchanged.

_: highshelf2(f0, gain, dtune) :_

Where:

• f0: cutoff frequency in Hz
• gain: the gain in dB
• dtune: detuning of the frequency in cents

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