The

pluggo

Plug-in Reference Guide

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Introduction

About This Manual

The Pluggo Plug-in Reference Guide provides an alphabetical description of all the plug-ins included with **pluggo**. Each plug-in description usually consists of four parts:

- A *What It Does* section that presents an overview of the plug-in and its function in life.
- A picture of the interface of the plug-in
- A listing of all the parameters and, if applicable, special interface controls (*Interface Elements*). The listing is called *Visible Parameters* because they are listed in the pop-up menu of a Modulator plug-in.
- An *Insights* section that provides some hints about how the plug-in can be used and configured.

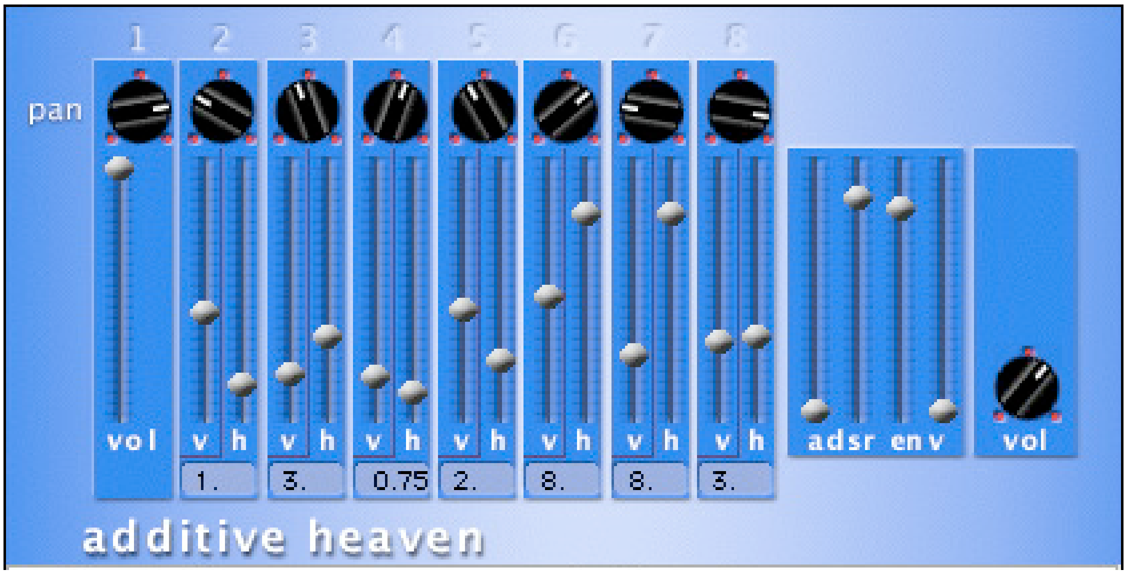
additive heaven

category: synthesis

input: MIDI | *cpu: light/medium* | *audio output: mono*

What It Does

additive heaven is an additive synthesizer featuring eight harmonic voices with individual settings for volume, harmonic and pan position. This synthesizer also includes an amplitude envelope that affects all eight voices, and an overall volume control. All of the synthesis elements can be modified with MIDI continuous controllers.



Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Harmonic Volume (voices 1–8)	0 (0%)	127 (100%)	21-28 (1-8)	Sets the volume of each of the harmonic voices.
Harmonic (voices 2-8)	0 (0.1 x)	127 (10.0 x)	14-20 (2-8)	Sets the harmonic (frequency multiplier) for each of the voices. The harmonic for the first voice is preset to 1.0. The harmonic settings, along with the volume settings, determine the timbre of the synthesizer.
Pan Position (voices 1–8)	0 (left)	127 (right)	53-60 (1-8)	Adjusts the pan setting of each of the harmonic voices. A value of 64 (a vertical knob orientation) represent panning to the center.
Attack	0 (2 ms)	127 (8002 ms)	73	Sets the amplitude envelope attack rate. When a note is played on the synthesizer, the amplitude envelope is triggered, and the attack time determines how long it takes the amplitude to move from zero to its maximum value.

Name	*Min*	*Max*	*CC#*	*Description*
Decay	0 (2 ms)	127 (6002 ms)	75	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the amplitude to move to its sustain level.
Sustain	0 (0%)	127 (100%)	76	Sets the amplitude envelope sustain level. The amplitude will remain at this level until the note is released. The actual sustain level is modified by the key velocity received over MIDI.
Release	0 (5 ms)	127 (10,005 ms)	77	Sets the amplitude envelope release rate. This is the amount of time it takes for the amplitude to move to zero after the note has been released.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.

Insights

- Additive synthesizers are often used for spacey or bell-like tones. The basis of this synthesis method is the addition of harmonic (in tune) or enharmonic (out of tune) tones to a base waveform, creating a more complex waveform as a result.
- Few additive synthesizers have provided individual voice panning, but the effect allows string and bell tones to be widened without the negative effects of pitch or phase alteration (as would be found with chorus or delay effects)

analogue drums

category: synthesis

input: MIDI *cpu: light* *audio output: mono*

What It Does

analogue drums is a tasty analog drum emulator, with synthesized bass, snare, hi-hat and tom voices. Each voice is wildly programmable, with timbres from rattling to blipping. All of the user interface elements can be modified using MIDI continuous controllers.

Visible Parameters

Name	*Min*	*Max*	*MIDI CC#*	*Description*
Bass Drum Tune	0 (20 Hz)	127 (147 Hz)	14	Sets the tuning of the bass drum.
Bass Drum Tone	0 (0 Hz)	127 (254 Hz)	15	Applies a downward pitch bend to the bass drum. As the setting is increased, the initial frequency of a drum hit will increase.
Bass Drum Decay	0 (0 ms)	127 (3000 ms)	16	Sets the rate at which a bass drum hit will decay to silence.
Bass Drum Attack	0 (0 %)	127 (100 %)	17	Adjusts the amount of “bite” that is added to the core bass timbre. This tone is a mixture of pink noise and sawtooth wave, and can provide definition to the initial drum attack.
Snare Drum Tune	0 (100 Hz)	127 (1270 Hz)	19	Sets the tuning of the snare drum “body” tone.

Name	*Min*	*Max*	*MIDI CC#*	*Description*
Snare Drum Snap	0 (0%)	127 (100%)	21	Adjusts the amount of "snap" that is added to the core snare timbre. The snap setting adds definition to the attack of a snare drum hit.
Snare Drum Decay	0 (0 ms)	127 (3000 ms)	20	Sets the rate at which a snare drum hit will decay to silence.
Snare Drum Hicut	0 (700 Hz)	127 (5780 Hz)	22	Applies a low-pass filter to the snare drum output. A low setting will dampen most of the high frequency content of the snare drum.
Hi-Hat Tune	0 (0 Hz)	127 (1000 Hz)	24	Sets the tuning of the Hi-Hat. Since the Hi-Hat tone is created using a "noise" waveform, there is no true tonal center. This adjustment "offsets" the frequency of the noise by the amount selected.
Hi-Hat C Decay	0 (0 ms)	127 (1000 ms)	25	Sets the rate at which a closed hi-hat strike will decay to silence.
Hi-Hat O Decay	0 (0 ms)	127 (3000 ms)	26	Sets the rate at which an open hi-hat strike will decay to silence.
Tom Low (Tuning)	0 (30 Hz)	127 (157 Hz)	28	Sets the tuning of the low tom-tom.
Tom Mid (Tuning)	0 (100 Hz)	127 (354 Hz)	29	Sets the tuning of the middle tom-tom.
Tom Hi (Tuning)	0 (200 Hz)	127 (581 Hz)	30	Sets the tuning of the high tom-tom.
Tom Decay	0 (0 ms)	127 (3000 ms)	31	Sets the rate at which a tom-tom strike will decay to silence. This setting is shared by all three tom-toms.
Bass Drum Volume	0 (0%)	127 (100%)	18	Adjusts the relative volume of the bass drum.
Snare Drum Volume	0 (0%)	127 (100%)	23	Adjusts the relative volume of the snare drum.
Hi-Hat Volume	0 (0%)	127 (100%)	27	Adjusts the relative volume of the closed hi-hat.
Tom Volume	0 (0%)	127 (100%)	32	Adjusts the relative volume of the tom-toms.
Indicator Lights	off	on	N/A	Each indicator will light when its respective drum is struck.
Drum Pads	off	on	N/A	In addition to using MIDI to strike the drums, you can click on the drum pads to sound each voice. Clicking on the pad will produce a drum hit at velocity 127.

analogue drums

category: synthesis

input: MIDI | *cpu: light* | *audio output: mono*

Name	*Min*	*Max*	*MIDI CC#*	*Description*
Volume	0	127	7	Adjusts the global level of the output signal.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the kick drum tuning.

Insights

- The MIDI notes used to fire the drums sounds are:

Bass Drum:	36
Snare Drum:	38
Closed Hi-Hat:	42
Open Hi-Hat:	46
Low Tom-Tom:	41
Mid Tom-Tom:	43
High Tom-Tom:	45

- Using the modulation wheel to control the bass drum tuning can allow you to track the bass drum to a bass synth in your track. While subtle, this type of effect can help a track sound like it is "locked in" to the bass groove.

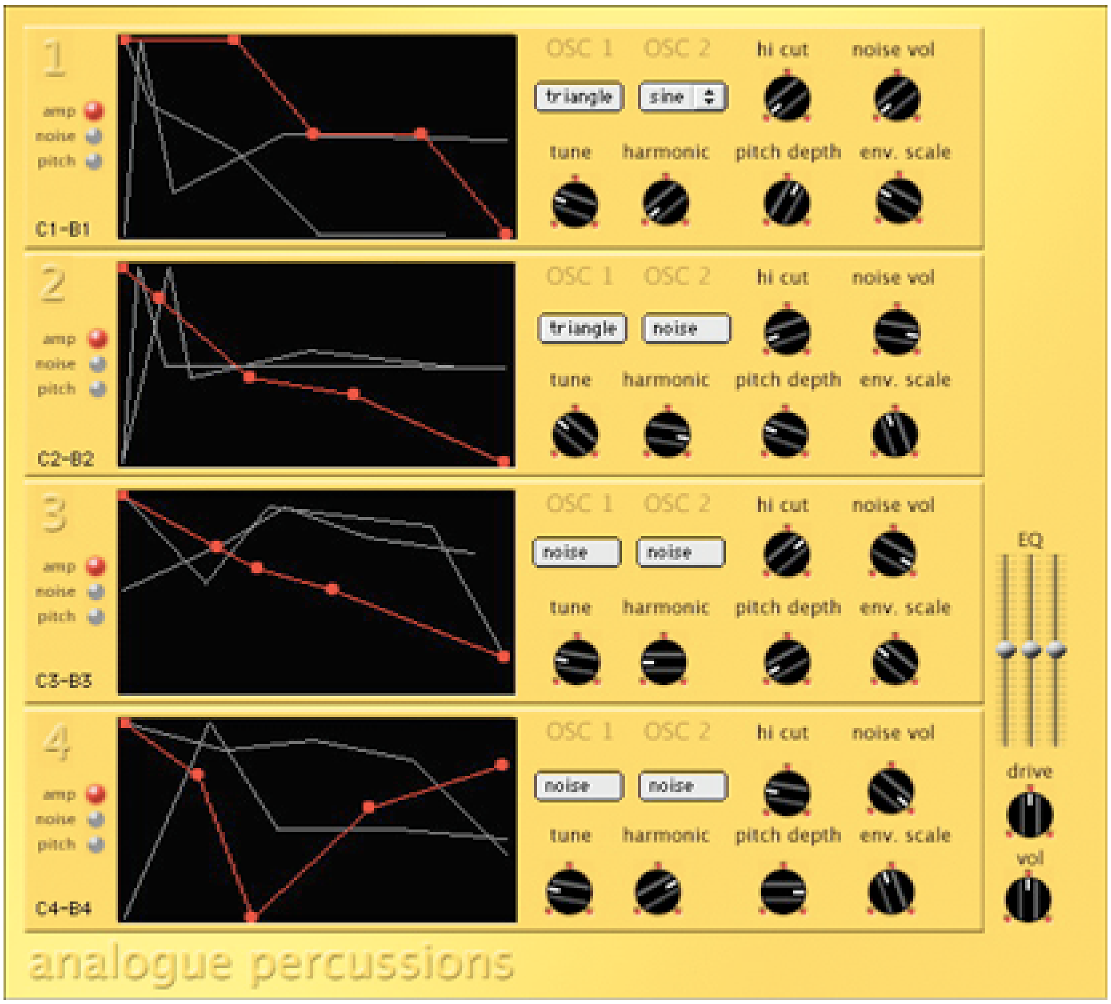
category: synthesis

analogue percussion

input: MIDI *cpu: light* *audio output: mono/stereo*

What It Does

analogue percussion is a versatile analog-style percussion synthesizer with four voices – each one spanning an octave of MIDI input. The main editing function of analogue percussion is provided by a multi-envelope editor. The envelope selector to the left of the display activates one of the five-point envelopes, which control voice amplitude, noise level and oscillator pitch.



Visible Channel Parameters

Name	*Min*	*Max*	*Units*	*Description*
Oscillator 1 waveform	0 (sine)	4 (noise)		Selects the waveform used by the primary oscillator. Options include sine, triangle, saw, square and noise waveforms.

analogue percussion

category: synthesis

input: MIDI *cpu: light* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Oscillator 2 waveform	0 (sine)	4 (noise)		Selects the waveform used by the secondary oscillator. Options are sine, triangle, saw, square and noise.
Hi-Cut	0	10,000	Hz	Applies a hi-pass filter across the output of the noise generator. This allows some tailoring of the noise characteristic.
Noise Volume	0	100	percent	Sets the level of the noise generator, as well as the secondary oscillator (OSC 2).
Tuning	+0	+127	MIDI note	Alters the tuning of the two oscillators by adding a value to the incoming MIDI note value. The 0-127 provides over 10 octaves of pitch increase to the oscillator tuning.
Harmonic	0x	12.7x		Sets the harmonic multiplier of the secondary oscillator (which provides both output tone and frequency modulation of the primary oscillator). Very high settings will provide a wide range of aliasing noises and strange FM overtones.
Pitch Depth	0	100	percent	Adjusts the amount of effect that the pitch envelope will have on the primary oscillator.
Envelope Scale	0	3.2	seconds	Determines the length of time that is represented by the envelope editing display. The actual duration of the envelope can be shorter, since the final segment of the envelope does not have to extend completely to the right of the display.

Visible Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
EQ - Low	-6	+6	dB	Provides a 6 dB boost or cut at 200 Hz.
EQ - Mid	-6	+6	dB	Provides a 6 dB boost or cut at 1000 Hz.
EQ - High	-6	+6	dB	Provides a 6 dB boost or cut at 5000 Hz.
Drive	0	100	percent	Applies a non-linear “overdrive” circuit to the mixed output of the device.
Volume	0	100	percent	Adjusts the global level of the output signal.

Interface Elements

- **Active Envelope Selection**: For each voice, you can select the active envelope - amp (amplitude), noise (noise level) or pitch (Osc 1/2 base pitch). The selected envelope will be displayed in red, and will be editable.
- **Envelope Editor**: The active envelope is modified in the Envelope Editor. Each envelope is a five-point envelope, with no sustain point. The duration of the editor is based on the Envelope Scale control (discussed below), and the magnitude of envelopes are 0.0 –1.0 (amp and noise envelopes) and -12.0 –12.0 semitones (pitch envelope).

Insights

- The secondary oscillator (OSC 2) is not simply mixed into the output signal – its use is much more complex. First, the secondary oscillator is used as an FM source for the primary oscillator (OSC 1). Secondly, this oscillator is mixed with the output of the noise generator to create the “noise output”, which is controlled by the noise envelope, hi-cut filter and noise volume settings. It’s best to think of the secondary oscillator as an “influencing agent” on the tone, rather than a direct sound source.
- The use of octave stretches for each voice provides a lot of latitude for creating tonal or semi-tonal percussion. Several of the presets provide examples of tonality in a percussive setting.
- While the voices are individually generated, they are based on single-cycle waveform samples. When using high tuning or harmonic settings, unpredictable aliasing can occur, occasionally with interesting results. The presets *Harley Set* (voices 3 and 4) and *Just A Bad Idea* (voices 1 and 2) show some of the results of using extreme tuning and harmonic settings.
- The envelope setting are not affected by the Randomize All and Evolve All commands in the pop-up menu. While Randomizing can still be useful (since it will alter the oscillator and mix controls), randomization of the envelopes is too unpredictable to be usable.

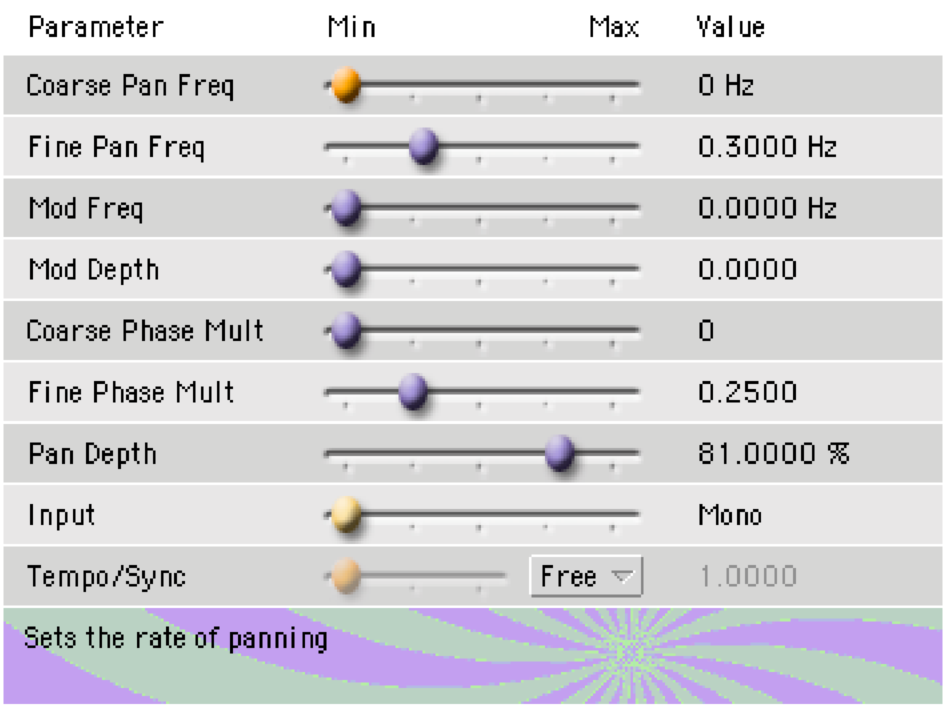
Audio Rate Pan

accepts sync | *category: multichannel*

Audio input: mono or stereo | *cpu: light* | *audio output: stereo*

What It Does

Audio Rate Pan pans a sound from left to right by changing the amplitude of the input in the output channels. You have control over the panning frequency, as well as the ability to modulate the panning frequency itself. This creates complex effects that can vary over a period of several seconds. *Audio Rate Pan* also features a “phase multiplier” control that changes the panning curve from a sine wave to more complex shapes. At higher rates of panning as well as higher phase multiplier values, the effect takes on the character of ring modulation.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Coarse Pan Freq	0	75	Hz	Sets the integer component of the rate of panning—how often the sound switches between the left and right output channels. This parameter is disabled when the Sync mode is set to Host, Plug, or UDT.
Coarse Pan Freq Mult	0	25		This parameter is displayed as a slider under Coarse Pan Freq when the Sync mode is set to Host, Plug, or UDT. It sets a multiplication factor on the Coarse Pan Freq Units parameter to produce the effective rate of panning relative to the tempo. For example, a value of 2 when the Units parameter is 1/4 gives an effective panning interval of a half note at the current tempo. This parameter is disabled when the Sync mode is set to Free.

Name	*Min*	*Max*	*Units*	*Description*
Coarse Pan Note Value	1	1/64t		This parameter is displayed as a pop-up menu under Coarse Pan Freq when the Sync mode is set to Host, Plug, or UDT. It sets a base note duration value that determines the effective rate of panning relative to the tempo. The note duration value is multiplied by the Coarse Pan Freq Mult parameter. For example, a value of 1/4 when the Mult is set to 2 gives an effective panning interval of a half note at the current tempo. This parameter is disabled when the Sync mode is set to Free.
Fine Pan Freq	0	1	Hz	Sets the fractional component of the rate of panning. This parameter is disabled when the Sync mode is set to Host or Plug.
Mod Freq	0	10	Hz	The rate at which the panning frequency is modulated. This parameter is disabled when the Sync mode is set to Host or Plug.
Mod Depth	0	4		The amount by which the oscillator whose frequency is set by Mod Freq modulates the panning frequency. This parameter is disabled when the Sync mode is set to Host or Plug.
Coarse Phase Mult	0	125		Values greater than 1 are similar in effect to a multiplier on the coarse panning frequency and create ring modulation effects.
Fine Phase Mult	0	1		A value of 0.25 is a normal sine wave pan, other values create irregular pan trajectories. The effect is non-linear, we recommend experimenting to find interesting settings.
Pan Depth	0	100	percent	Sets how much of the amplitude of the original signal is affected by the pan. A value 0 means the pan has no effect, while a value of 100 means that the pan reduces the signal to zero amplitude in one channel when it has moved to the opposite channel.
Input	mono	stereo		When set to mono, the left (mono) input is panned between the left and right outputs. When set to stereo, each input channel is panned “in place.” Is that really panning? It sounds pretty good.
Tempo	1	300	BPM	Tempo is displayed as a slider in the Tempo/Sync parameter. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the rate of panning. When in Host and Plug modes, the tempo is an indicator that cannot be changed--it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range and Audio Rate Pan will calculate the panning rate based on the values of the Coarse Pan Freq Mult and Coarse Pan Freq Units parameters. This parameter is disabled when the Sync mode is set to Free.

Audio Rate Pan

accepts sync | *category: multichannel*

Audio input: mono or stereo | *cpu: light* | *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Sync Mode	Free	UDT		Sync Mode is a pop-up menu in the Tempo/Sync parameter. There are four modes of sync available: • Free mode lets you set the panning rate in Hertz independent of the host sequencer. • Host, mode synchronizes panning to the host tempo. • Plug mode synchronizes panning to the beat output of PluggoSync • UDT (User-Defined Tempo) mode is equivalent to Free mode but allows you to set the rate of panning in terms of a tempo and note unit values. Note: Host mode is only available in VST 2.0 and MAS applications that support it.

Insights

- The *Punchy 47* preset demonstrates the way in which rapid amplitude modulation set with the Coarse Phase Mult can have a filtering or EQ effect. In this case, bass frequencies are emphasized.
- The *Stereo Shudder* preset demonstrates the complex modulation you can get from using the Mod Freq and Mod Depth parameters.
- The *Queasy Pan* preset uses a Fine Phase Mult of something other than 0.25 to produce an irregular kind of panning that never seems to repeat.

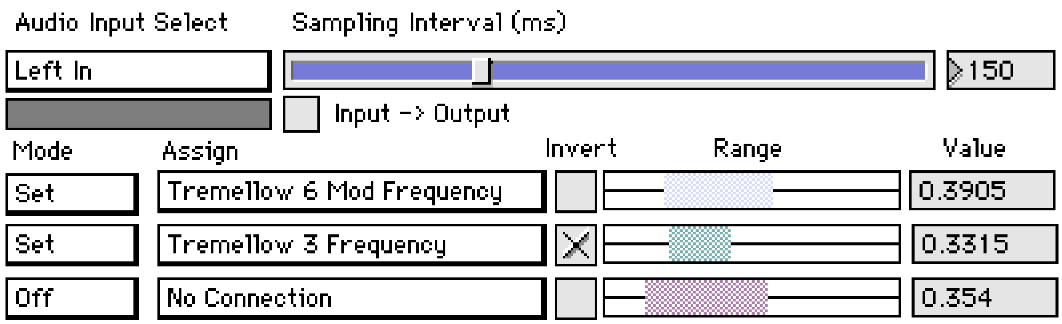
category: modulator

Audio2Control

audio input: mono *cpu: light* *audio output: none*

What It Does

Audio2Control samples an input audio signal at a regular rate that is much lower than the audio sampling rate. It lets you assign this downsampled signal to the parameters of other plug-ins. The data can be scaled and inverted before being sent as a source of modulation.



Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Source	Left	Noise		Selects the audio input to be used for sampling.
Sampling interval	5	500	ms	Sets the time between samples. Lower values could slow your CPU significantly.
Input -> Output	Off	On		Sets whether the audio input to the plug-in is passed through to the audio output. *Audio2Control* produces no audio output in and of itself.

Parameters for Each Modulation Destination

Name	*Min*	*Max*	*Units*	*Description*
Mode	Off	Scale		When the mode is Set (the most common), the modulator's value directly sets the assigned parameter. When the mode is Offset, the modulator's value is added to the current value of the parameter (before modulation). When the mode is Scale, the modulator's value multiplies the current value of the parameter. When the mode is Off, no changes to the parameter occur.
Assign				Choose a plug-in and a parameter to modulate from this menu. The parameters listed will depend on the plug-ins that are currently inserted.
Invert	Off	On		When checked, the sampled value is inverted. In Audio2Control this means that small sample values are made the largest values, and vice versa.

Audio2Control

	accepts sync	*category: modulator*
audio input: mono	*cpu: light*	*audio output: none*

Name	*Min*	*Max*	*Units*	*Description*
Range Low Value	0	1		Use the range bar to scale the modulator's output data. The low value in the range is the minimum output value sent to the assigned parameter.
Range High Value	0	1		Use the range bar to scale the modulator's output data. The high value in the range is the maximum output value sent to the assigned parameter.

Insights

- One practical use of *Audio2Control* would be in conjunction with audio tracks that you don't listen to—rather, you use them to record some type of control data, playing them back only through *Audio2Control* to be used for plug-in parameter modulation. The *Control2Audio* plug-in can convert a control data into an audio signal of this type.
- On the other hand, you might find *Audio2Control* as a very suitable parameter modulator. The effect of modulating a parameter with sampled audio tends to have a random character that isn't purely "noisy" and retains some vague connection with the original signal. You might try modulating a parameter of an effect on one track with the sampled audio of another completely unrelated track.

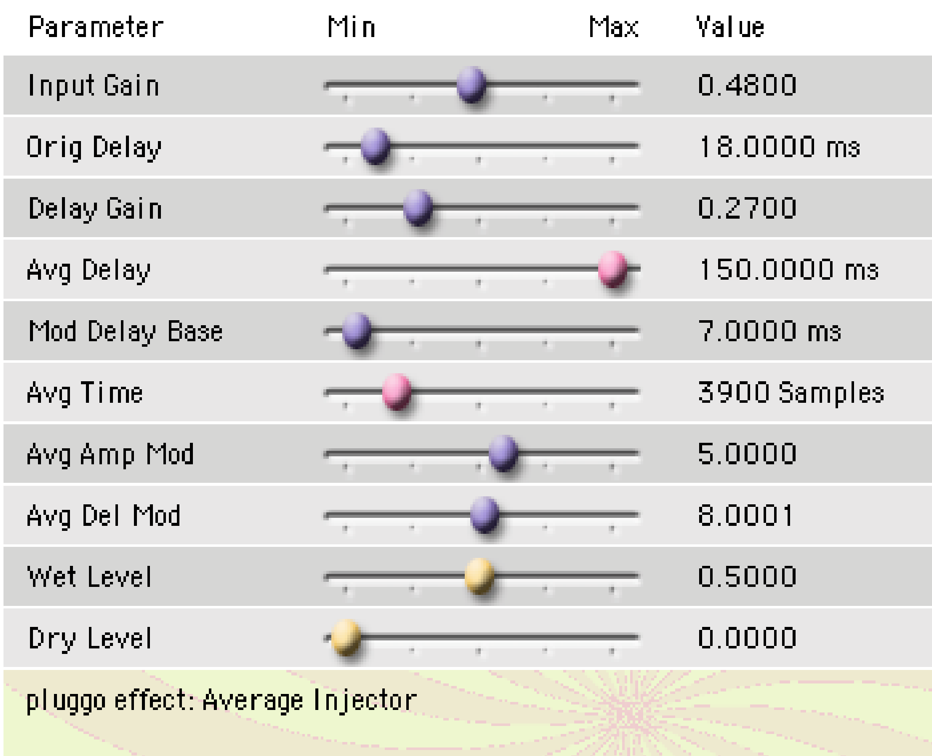
categories: delay, distortion

Average Injector

audio input: mono	*cpu: light*	*audio output: mono*

What It Does

Average Injector modulates the delay time and amplitude of its input based on an average of the signal's amplitude. Parameters include the time over which the signal is averaged, as well as the offset between the average computation and when the average-based modulation is applied. You also have control over the amount of average-based modulation that is applied to the input signal. Just to be arbitrary, the modulated signal is subtracted from the delayed original signal.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Input Gain	0	1		Reduces the input level before any processing is applied. Increasing this level can exaggerate the effect of the average computation.
Orig Delay	0	150	ms	Sets the delay of the unprocessed input signal, from which the average-modulated signal will be subtracted.
Delay Gain	0	1		Gain on the average-modulated delay that is subtracted from the original signal.

Average Injector

categories: delay, distortion

audio input: mono | *cpu: light* | *audio output: mono*

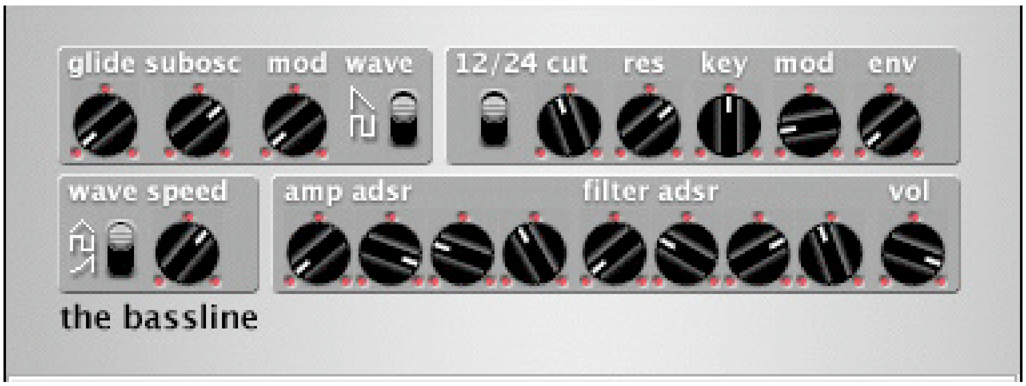
Name	*Min*	*Max*	*Units*	*Description*
Avg Delay	0	150	ms	Delay before computing the signal average. If this value is 0, the modulation effects of the average computation will be heard roughly at the same time as the changes in the average themselves. As the delay time increases, the modulation effects of the average computation will be delayed relative to when they are heard in the original signal.
Mod Delay Base	0	150	ms	The delay time around which the signal average's value modulates. This delayed signal is subtracted from the original delayed signal.
Avg Time	1	20000	samples	Time over which the signal average is computer. Larger values will produce smaller, less sudden fluctuations. Smaller values will tend to make the output resemble noise.
Avg Amp Mod	-25	25		Sets the level of amplitude modulation caused by changes in the value of the average computation. Negative values invert the amplitude's effects, in which case higher signal average values produce smaller output amplitudes.
Avg Del Mod	-150	150		Sets the amount of delay time modulation caused by changes in the value of the average computation. Negative values invert the amplitude's effects, in which case higher signal average values produce smaller output amplitudes.
Wet Level	0	1		Sets the overall level of the effect.
Dry Level	0	1		Sets the level of the unprocessed input signal.

Insights

- *Average Injector* is rather arbitrary combination of processing elements: compression, delays, and phase inversion. Start exploring by adjusting the AvgTime parameter, which affects the signal averaging that begins the modulation process. Shorter times create noisier modulations, and longer times more subtle changes. From there, listen to what happens when you change the AvgAmpMod and AvgDelMod parameters.
- As with a number of other plug-ins, *Average Injector* was developed while listening to jazz recordings. In particular, we suggest applying the effect to trumpet sounds or other lead instruments, where it has the ability to make acoustic instruments sound synthetic. The preset *Yeah Yeah* is an example of this. You may have to adjust the Input Gain parameter to achieve a decent synthetic effect.

What It Does

bassline is a simple but powerful analog-style bass synthesizer. It features a single oscillator design (with sub-oscillator), 2- and 4-pole low-pass resonant filter, integral LFO and independent amplitude and filter envelopes.



Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Glide	0 (2 ms)	127 (256 ms)	14	Sets the amount of time for the pitch to reach a MIDI note value.
Sub-osc Level	0 (0%)	127 (100%)	15	Determines the mix level of the sub-oscillator (a square wave tone one octave below the MIDI pitch).
Oscillator Modulation	0 (0 cents)	127 (1200 cents)	16	Sets the amount of LFO modulation on the oscillator pitch.
Oscillator Waveform	0 (square)	1 (saw)	17	Adjusts the waveform used by the oscillator.
Filter Type	0 (12 dB)	1 (24 dB)	18	Determines the filter type, with either 12 dB (2-pole) or 24 dB (4-pole) filter types available.
Filter Cutoff	0 (0 Hz)	127 (10 kHz)	19	Sets the cutoff frequency of the filter.
Filter Resonance	0 (0.0)	127 (1.0)	20	Sets the resonance value (sometimes called Q or pre-emphasis) of the filter. Resonance values range from 0.0 (no resonance) to 1.0 (self-oscillation).
Filter Key Follow	0 (0%)	127 (1000%)	21	Adjusts the amount that the filter cutoff setting will “follow” the current incoming MIDI note.
Filter Modulation	0 (0 Hz)	127 (5 kHz)	22	Sets the amount that the LFO modulates the filter cutoff.

bassline

category: synthesis

input: MIDI *cpu: light* *audio output: mono*

Name	*Min*	*Max*	*CC#*	*Description*
Filter Envelope Amount	0 (0%)	127 (100%)	23	Sets the amount that the Filter Envelope modulates the filter cutoff.
LFO Waveform	0 (saw)	2 (triangle)	24	Determines the waveform output by the LFO. Options include sawtooth, square and triangle waveforms.
LFO Speed	0 (.6 Hz)	127 (20 Hz)	25	Sets the rate of the LFO modulator.
Amplitude Envelope Attack	0 (0 ms)	127 (5000 ms)	26	Sets the amplitude envelope attack rate. When a note is played on the synthesizer, the amplitude envelope is triggered, and the attack time determines how long it takes the amplitude to move from zero to its maximum value.
Amplitude Envelope Decay	0 (0 ms)	127 (4000 ms)	27	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the amplitude to move to its sustain level.
Amplitude Envelope Sustain	0 (0%)	127 (100%)	28	Sets the amplitude envelope sustain level. The amplitude will remain at this level until the note is released.
Amplitude Envelope Release	0 (0 ms)	127 (6000 ms)	29	Sets the amplitude envelope release rate. This is the amount of time it takes for the amplitude to return to zero after a note has been released.
Filter Envelope Attack	0 (0 ms)	127 (5000 ms)	46	Sets the filter envelope attack rate. When a note is played on the synthesizer, the filter envelope is triggered, and the attack time determines how long it takes the modulation output to move from zero to its maximum value.
Filter Envelope Decay	0 (0 ms)	127 (4000 ms)	47	Sets the filter envelope decay rate. After the envelope has reached its maximum value (based on the Attack parameter), the decay time determines how long it takes for the modulation output to move to its sustain level.
Filter Envelope Sustain	0 (0%)	127 (100%)	48	Sets the filter envelope sustain level. The modulation output will remain at this level until the note is released.
Filter Envelope Release	0 (0 ms)	127 (6000 ms)	49	Sets the filter envelope release rate. This is the amount of time it takes for the modulation output to move to zero after the note has been released.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.

Insights

- Although designed as a bass synthesizer, bassline can also acts as a credible lead synthesizer – especially with the addition of some overdrive and delay effects. Good examples of lead synth presets are Majestic Walk and Rubber Band Man.
- For a more "analog" sound, use at least a little glide, and apply some filter envelope modulation. The sound that most people refer to as "phat-ness" often comes from creative use of filter and amplitude envelopes. Presets like Moog-a-licious, Big Foot and Shaky Tail use these ideas to add authenticity to their sound.

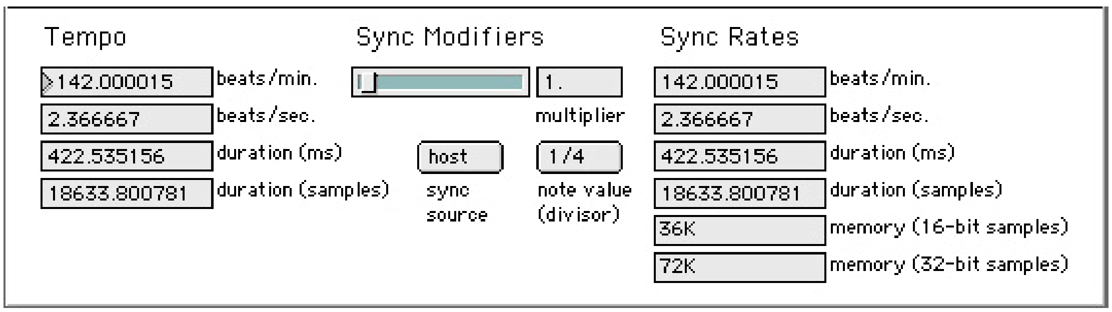
beatN

accepts sync | *category: visual display*

audio input: none | *cpu: light* | *audio output: none*

What It Does

beatN neither accepts nor produces audio output. It is a handy calculator which accepts either host sync information from a host application or values you enter and calculates useful tempo and sync-based values which are useful in recording sessions. In addition to beats/min. and beats/sec. calculations, *beatN* also calculates beat durations in milliseconds and durations at 44.1 kHz. sampling rates.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
beats/min	1	300	BPM	beats/min (or Tempo) are displayed and can be set in the number box. When the sync source is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the calculated time. When in Host and Plug modes, the tempo is an indicator that cannot be changed--it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range and beatN will calculate everything based on the input value. This parameter is disabled when the sync source is set to Free. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
multiplier	0	10		Sets a multiple of the note value unit currently selected that determines the calculated time at the current tempo. For example, if the value of this parameter were 2 and the units were set to 1/4 (quarter note), the resulting delay time would be a half-note at the current tempo. This parameter is disabled when the sync source is set to Free.

Name	*Min*	*Max*	*Units*	*Description*
sync source	Free	UDT		sync source is a pop-up menu that lets you set one of four modes of sync: • Free mode lets you set the delay time independent of the host sequencer. • Host mode synchronizes the delay time to the host tempo. • Plug mode synchronizes the delay time to the beat output of PluggoSync • UDT (User-Defined Tempo) mode is equivalent to Free mode but allows you to set the calculations in terms of tempo and note unit values. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
note value (divisor)	1	64t.		Sets the base note duration value used in determining the calculated values in relation to the current tempo. This note value is multiplied by the multiplier to obtain the total beat number used to calculate the displayed values. This parameter is disabled when the sync source is set to Free.

big ben bell

category: synthesis

input: MIDI | *cpu: light* | *audio output: mono*

What It Does

big ben bell is an FM-based bell synthesizer, with five preset harmonic settings, a stereo spread synthesizer and a simple attack/decay envelope. It provides wonderfully complex tones with relatively few controls.

Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Harmonic	0 (preset 1)	4 (preset 5)	14	Sets the FM modulator to one of five “magic” preset values that provide bell-like tones.
Stereo	0 (0%)	127 (100%)	15	Alters the left-channel FM modulation value to provide a stereo “widening” effect.
Attack	0 (2 ms)	127 (10,002 ms)	16	Sets the amplitude envelope attack rate. When a note is played on the synthesizer, the amplitude envelope is triggered, and the attack time determines how long it takes the amplitude to move from zero to its maximum value.
Decay	0 (100 ms)	127 (10,100 ms)	17	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes the amplitude to move back to zero.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.
Volume Level	0.0 (0%)	1.0 (100%)		Attenuates the output of the synthesizer to prevent overload.

Insights

- Use the stereo spread control with caution – since it alters the FM modulator, it will also seem to alter the tuning. Conversely, sequenced modulation of the stereo spread control can provide some head-twisting psycho-acoustic effects. Preset Church Bells (MW) is a good example of Stereo Spread used tastefully.

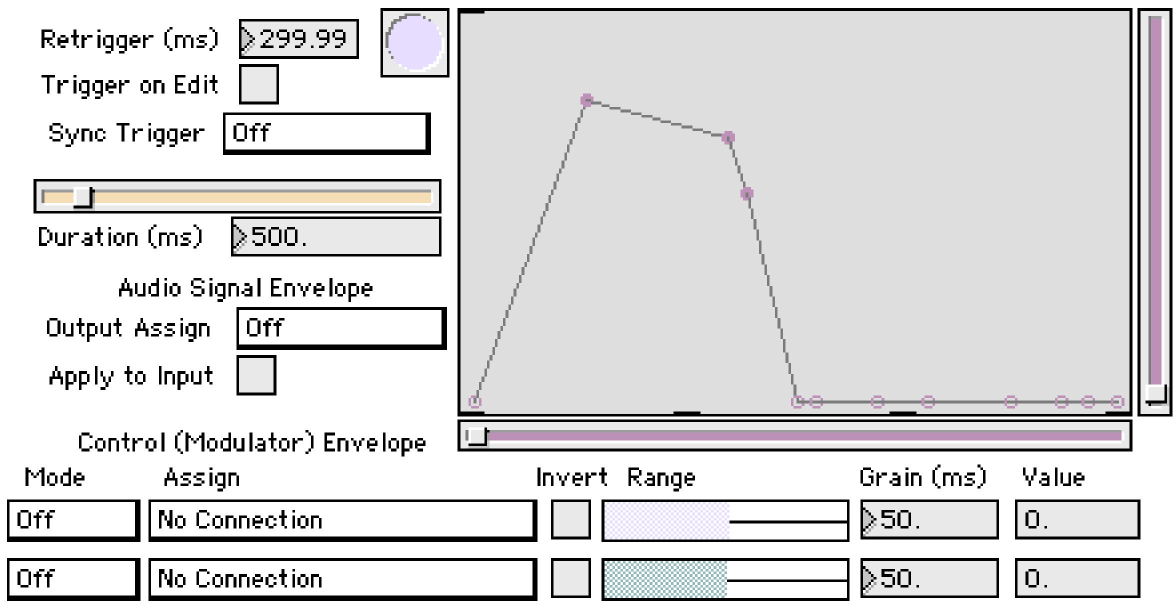
Breakpoints

accepts sync *categories: volume, modulator*

audio input: mono *cpu: light* *audio output: mono, pluggoBus, none*

What It Does

Breakpoints generates an audio-rate envelope that you can draw on the screen using 12 breakpoints. The envelope can be applied to an audio signal or sent as a modulation data to be applied to other plug-in parameters. The triggering of the envelope can be tied to the beat of the music using *PluggoSync*, or you can assign another modulator plug-in to trigger the envelope.



Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Env Duration	100	5000	ms	Sets the total duration of the envelope. This is the amount of time represented by the X axis of the graph, not the amount of time for which the envelope has a non-zero value. The Env Duration is also the interval for triggering when the Sync Trigger is set to Internal.
Trigger	0	—		Any non-zero value (or clicking on the blue button) will trigger the envelope, provided the envelope was not previously triggered within the amount of time set by the Retrigger Interval parameter.
Retrigger Interval	0	500	ms	Sets the amount of time after triggering before the envelope will trigger again based on a non-zero value in its Trigger parameter or a trigger signal coming from a Sync Trigger source.
Apply Env to Input	Off	On		If enabled, the audio-rate envelope is applied to the plug-in's input.

Name	*Min*	*Max*	*Units*	*Description*
Output Assign	Off	pluggoBus 4L		Sets the destination for the audio-rate envelope. If the value is Off, the audio-rate envelope is not sent anywhere. If the value is Out, the audio-rate envelope is sent to the plug-in's output. Note that this will cause a change in the level meter.
Sync Trigger	Off	Internal		Sets an automatic trigger source for the envelope. If the value is Off, no automatic triggering occurs. If the value is pluggoSync 1 - pluggoSync 4, triggering occurs when a sync signal is present on of these "sync busses." If the value is Host, the envelope retriggers at the rate set by the Unit and Mult parameters, which specify a beat value with respect to the current tempo The Unit and Mult parameters become visible in the Breakpoints interface when Host is chosen from the Sync Trigger menu. In addition, the Duration of the envelope is always the same as the retrigger interval. Note: Host mode is only available in VST 2.0 and MAS applications that support it. If the value is Internal, the envelope triggers at the interval specified by the Env Duration parameter.
Unit	1	64t.		This parameter is only visible and enabled when the Sync Trigger is set to Host. It sets a base note duration value that determines the effective retriggering interval relative to the host tempo. The note duration value is multiplied by the Mult parameter. For example, a value of 1/4 (quarter note) when Mult is set to 2 gives an effective panning interval of a half note at the current tempo.
Mult	0	1000		This parameter is only visible and enabled when the Sync Trigger is set to Host. It sets a multiplication factor on the Unit parameter to produce the effective retriggering interval relative to the host tempo. For example, a value of 2 when the Unit parameter is set to a quarter note gives an effective retrigger interval of a half note at the current tempo.

Parameters for Each Modulation Destination

Name	*Min*	*Max*	*Units*	*Description*
Mode	Off	Scale		When the mode is Set (the most common), the modulator's value directly sets the assigned parameter. When the mode is Offset, the modulator's value is added to the current value of the parameter (before modulation). When the mode is Scale, the modulator's value multiplies the current value of the parameter. When the mode is Off, no changes to the parameter occur.
Assign				Choose a plug-in and a parameter to modulate from this menu. The parameters listed will depend on the plug-ins that are currently inserted.

Breakpoints

accepts sync *categories: volume, modulator*

audio input: mono *cpu: light* *audio output: mono, pluggoBus, none*

Name	*Min*	*Max*	*Units*	*Description*
Invert	Off	On		When checked, the sampled value is inverted. In Audio2Control this means that negative sample values are made positive.
Range Low Value	0	1		Use the range bar to scale the modulator's output data. The low value in the range is the minimum output value sent to the assigned parameter.
Range High Value	0	1		Use the range bar to scale the modulator's output data. The high value in the range is the maximum output value sent to the assigned parameter.
Grain	20	300		Sets the interval at which the audio-rate envelope is sampled to produce the control-rate modulator envelope.

Interface Elements

- The **Trigger on Edit** box, if checked, plays the envelope each time you change it with the mouse.
- The purple sliders at the bottom and left of the envelope display the progress of the envelope as well as its current value.

Insights

- When applied to its audio input, *Breakpoints* could be used for noise gating effects on drum tracks if it is retriggered in sync with the track. This is a snap when you use the Host value of the Sync Trigger.

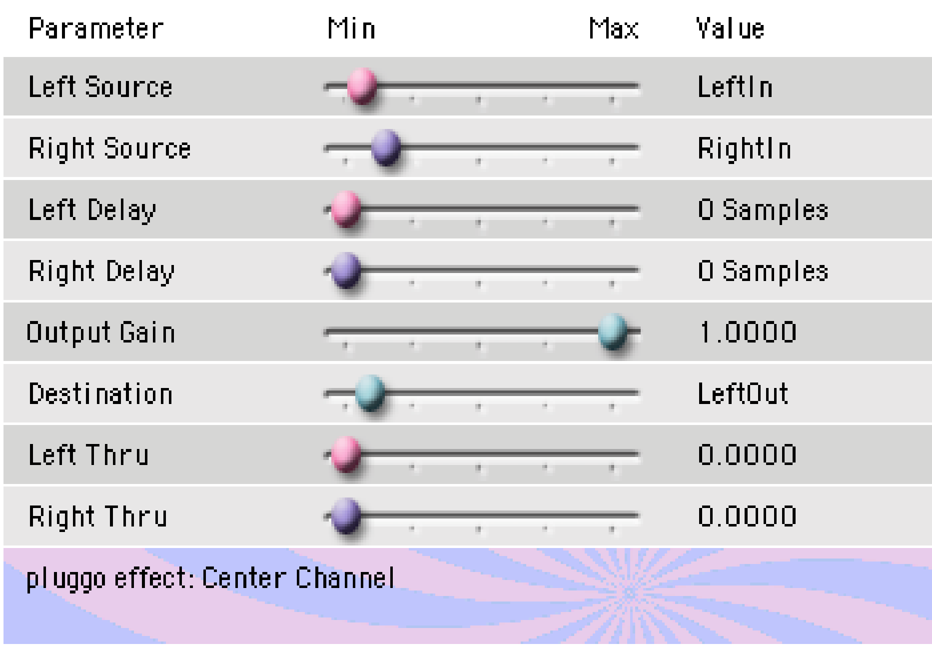
category: volume

Center Channel

audio input: stereo | *cpu: light* | *audio output: mono*

What It Does

Center Channel subtracts the one of its input channels from another. A useful utility for eliminating what is in common to both channels in a stereo recording (vocals, bass lines, etc.). Channels can be delayed relative to each other in case one channel has been delayed in the mixer. Slight delays produce interesting effects in and of themselves.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Left Source	Off	pluggo Bus 4R		Sets the input source for the left side of the subtraction. If Off, the signal is zero and the effect would be to phase invert the right side. LeftIn is the left input signal to the plug-in. RightIn is the right input signal to the plug-in. The pluggoBus selections patch the mono output of one of the signal routing busses to the plug-in.
Right Source	Off	pluggo Bus 4R		Sets the input source for the right side of the subtraction. If Off, the signal is zero and the effect would be to leave the left side unchanged. LeftIn is the left input signal to the plug-in. RightIn is the right input signal to the plug-in. The pluggoBus selections patch the mono output of one of the signal routing busses to the plug-in.

Center Channel

category: volume

audio input: stereo | *cpu: light* | *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Left Delay	0	2048	samples	Sets the delay of the left input. This may be necessary when using a pluggoBus as a Left Source in order to line up the two inputs—in this case, the delay value will be equal to the size of a processing vector (typically 512, 1024, or 2048 samples). Alternatively, you can introduce small delays (a sample or so) and get some interesting effects.
Right Delay	0	2048	samples	Sets the delay of the right input. This may be necessary when using a pluggoBus as a Right Source in order to line up the two inputs—in this case, the delay value will be equal to the size of a processing vector (typically 512, 1024, or 2048 samples). Alternatively, you can introduce small delays (a sample or so) and get some interesting effects.
Output Gain	0	1		Reduces the gain of the plug-in's output.
Destination	Left	pluggo Bus 4R		Sets the destination for the mono output of the plug-in.
Left Thru	0	1		Sets the gain on the unprocessed left channel input.
Right Thru	0	1		Sets the gain on the unprocessed right channel input.

Insights

- So-called "vocal eliminators" use this right-from-left subtraction technique to remove vocals from stereo recordings. This technique depends on the vocal being mixed equally in the left and right channels. Sometimes you can still hear the reverb tail of the vocal, as this is often panned to the left or right.
- The *No Bass* preset demonstrates how the subtraction of a signal from a delayed copy of itself (or something similar) acts as a high-pass filter. Indeed, the simplest digital high-pass filter takes differences between successive samples.
- The *Airy Ambience + Dry* preset demonstrates the effect when the two channels are subtracted with a larger delay—a distinctive phasing effect that sounds as if "air" has been pumped into the recording. Here, the processed signal is added back to the dry signal.

category: reverb **Chamberverb**

audio input: mono *cpu: medium* *audio output: stereo*

What It Does

Chamberverb is a reverberator that is made up of a network of allpass filters. The network—as well as the parameter values in the Original effect preset—are taken from Hal Chamberlin's book *Musical Applications of Microprocessors*. The Chamberverb Info view in the plug-in provides a diagram of the allpass network. Several additions have been made to the original design, including a low-pass filter on the output, a variable delay on two of the allpass stages, and a reasonably useless parameter called Punch that adds low end, something you usually don't want in reverb.

Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*

Chamberverb

category: reverb

audio input: mono | *cpu: medium* | *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Direct Gain	0	1		Sets the gain on the output of the direct signal input to the reverberator. This does not control the level at which the input is fed to the reverberator.
Reverb Gain	0	1		Sets the gain on the reverberator output. Parameters 1 and 2 together may be thought of as setting a “wet/dry mix.”
AP1 Delay	1	200	ms	Sets the delay time of the first allpass filter.
AP1 Feedback	0	0.90		Sets the feedback level of the first allpass filter.
AP2 Delay	1	200	ms	Sets the delay time of the second allpass filter.
AP2 Feedback	0	0.90		Sets the feedback level of the second allpass filter.
AP3 Delay	1	200	ms	Sets the delay time of the third allpass filter.
AP3 Feedback	0	0.90		Sets the feedback level of the third allpass filter.
AP4/5 Base Delay	1	200	ms	Sets the delay time of the fourth (left) and fifth (right) allpass filters. The delay time of the filter on the left is offset from that of the right by 0.15 ms to create a slight difference in the stereo image.
AP4/5 Feedback	0	0.90		Sets the feedback level of the fourth (left) and fifth (right) allpass filters.
AP 4/5 Mod Freq	0	5	Hz	Sets the delay time modulation frequency for the fourth and fifth allpass filters
AP 4/5 Mod Depth	0	10		Sets the amount of delay time modulation for the fourth and fifth allpass filters.
AP 6/7 Delay	1	200	ms	Sets the delay time of the sixth (left) and seventh (right) allpass filters. The delay time of the filter on the left is offset from that of the right by 0.14 ms to create a slight difference in the stereo image.
AP 6/7 Feedback	0	0.90		Sets the feedback level of the sixth (left) and seventh (right) allpass filters.
Filter Cutoff	100	20000	Hz	Sets the cutoff frequency of the lowpass filter at the output stage of the reverberator.
Punch	Off	On		If the value is On, the phase inversion of the feedback loop in the second allpass filter is eliminated. This tends to boost lower frequencies.

Insights

- The use of the variable delay in the fourth and fifth allpass filters creates a chorus effect within the reverb. Much of the time this is a very subtle effect, but then, reverb is typically about subtlety. You can hear the chorus in the *Windy Chorus* preset.
- All of the various delay time and feedback settings in this plug-in are quite a bit more subtle than merely changing the filter cutoff and the direct and reverb gain. If you’re looking for a different effect, you might try these first.

- Here is what Chamberlin had to say about this design: "The subtle but significant differences between channels insures that the reverberation will be perceived as coming from all directions, while the original signal retains its normal directivity." From *Musical Applications of Microprocessors*, copyright 1980 Hayden Book Company.

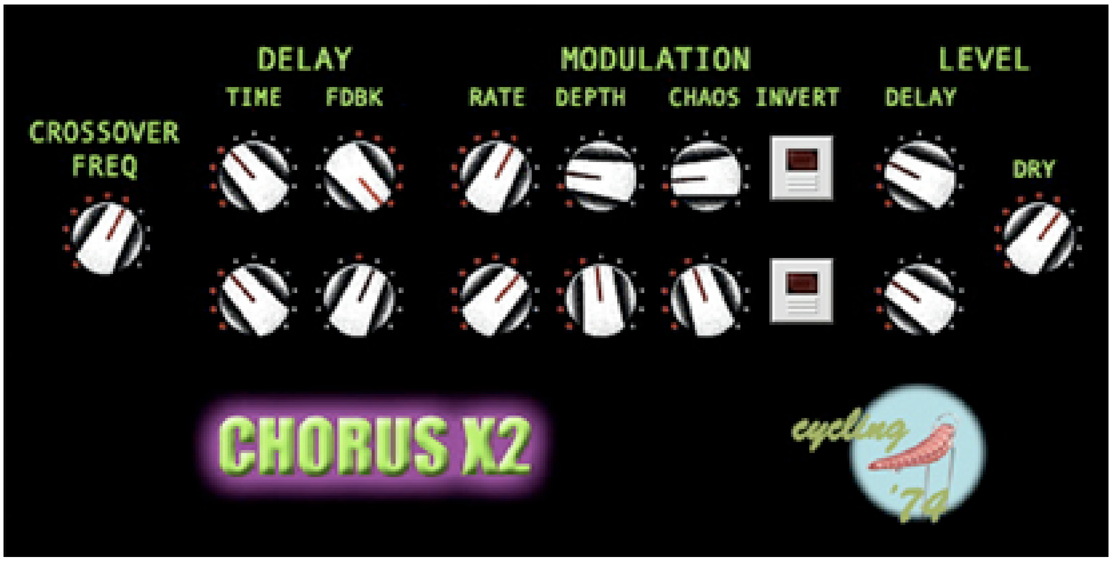
Chorus X2

category: delay

audio input: stereo *cpu: light* *audio output: stereo*

What It Does

ChorusX2 is a plug-in effect for creating chorusing and short delay effects. The input signal is divided into high- and low-frequency bands, with each frequency band processed separately by a pair of modulated delays.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Crossover Frequency	0	5000	Hz.	Sets the dividing frequency. Any audio content to the input signal above the dividing frequency will be processed by the upper stage of ChorusX2, and the remainder will be processed by the lower stage chorus unit.
Delay Time	0	500	msec.	Sets the delay time for the effect.
Fdbk	-99	99	Percent	Sets the amount of feedback injected into the delay line.
Modulation rate	0	10	Hz	Sets the rate at which the delay time is modulated.
Modulation depth	0	10	msec.	Sets the amount of delay time modulation.
Chaos	1	10	-	Sets the amount of randomness in the modulation stage.
Invert	0ff	On	-	Inverts the right-channel signal relative to the left-channel signal. The delayed signals (one for the upper band, one for the lower band) are sent to both output channels. If the Invert button is pressed, the right channel is inverted (multiplied by –1). This gives a nice stereo spread because the delayed signals are inverted with respect to each other, and also because they cause different cancellations with the dry signal.
Delay	1	10	-	Sets the gain on the output of each chorus
Dry	1	10	-	Sets the gain on the input fed directly to the output.

Insights

- The feedback knob is bi-directional: rotate it counter-clockwise for negative feedback, clockwise for positive feedback, and leave it at 12:00 for no feedback.
- This effect sounds best if its output is heard in stereo. The outputs of the two delays for each frequency band are sent exclusively to the right and left channels, creating a wide, moving stereo effect.

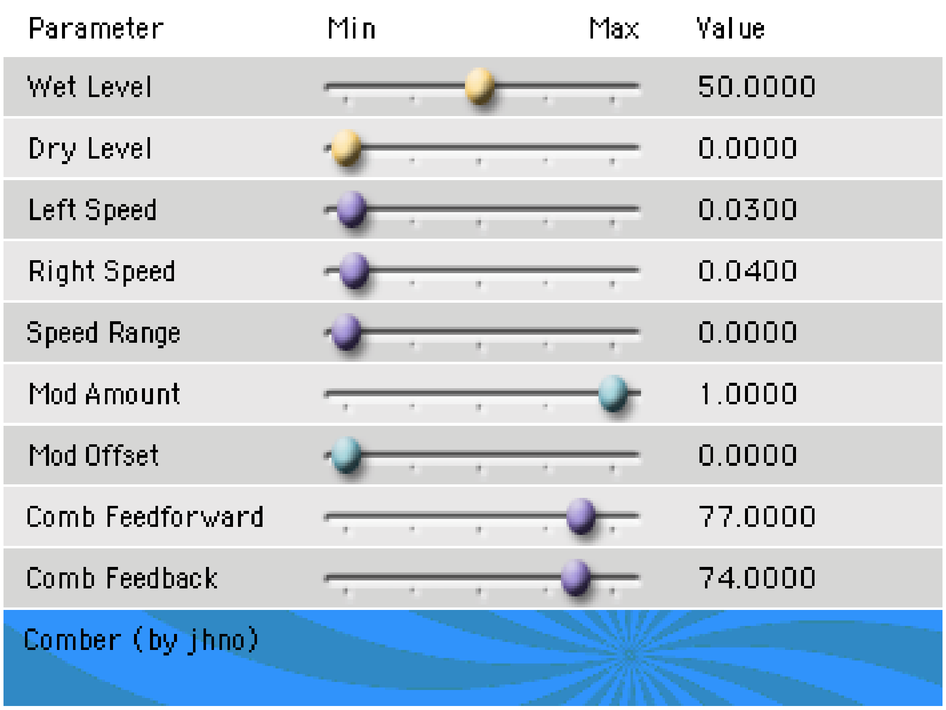
Comber

category: delay

audio input: stereo *cpu: light* *audio output: stereo*

What It Does

Comber is a stereo effect with two modulated comb filters. A comb filter is a series of delays, usually very short, that create phase cancellation and other effects. Comber is capable of a wide variety of sounds, due to its Feedback/Feedforward parameters, and a sine wave modulator that can range from sub-audio to audio frequencies.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the gain on the output of the comb filters.
Dry Level	0	1		Sets the gain on the input fed directly to the output.
Left Speed	0	1		Sets the modulation frequency of the left comb filter's delay time.
Right Speed	0	1		Sets the modulation frequency of the right comb filter's delay time.
Speed Range	0	1		Sets the maximum frequency for modulation of comb filter delay times. By keeping this low, you can precisely set LFO-speed modulation. By turning it up, you can coax the speeds into audio range.
Mod Amount	0	1		Sets the amount of modulation of comb filter delay times.

Name	*Min*	*Max*	*Units*	*Description*
Mod Offset	0	1		Sets an offset to be added to all of the comb filter delay time modulation, letting you set the center point from which they are modulated.
Comb Feedfoward	0	1		Sets the feedforward level of the comb filters.
Comb Feedback	0	1		Sets the feedback level of the comb filters.

Insights

- The *Space Delay* preset shows how you can use longer delay times to create trippy echo and ambience effects. Play with the Mod Offset parameter to hear what happens in different ranges of delay times.
- The term *feedforward* refers to a process in which past samples of the input are added to the current input, while *feedback* refers to a process in which past samples of the *output* are added to the current input. The two types of filtering have different characteristic sounds, with feedback having a more obvious coloring of the input signal.

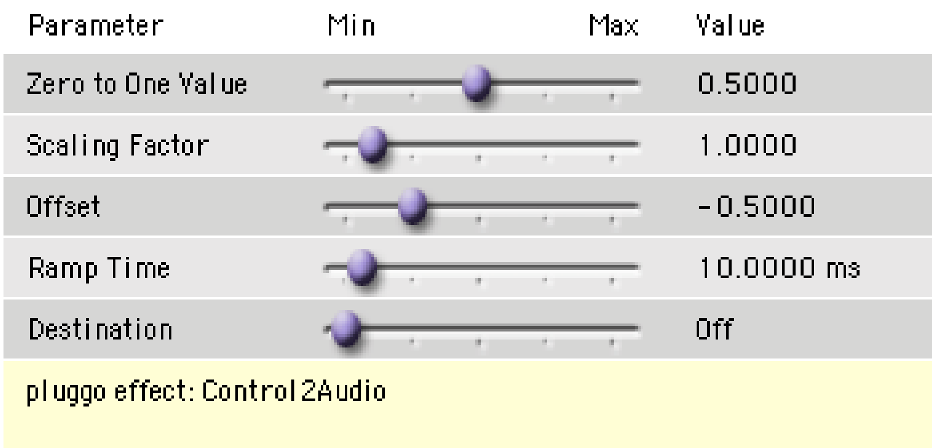
Control2Audio

category: signal generator

audio input: none | *cpu: light* | *audio output: mono, pluggoBus*

What It Does

Control2Audio generates an audio signal from a control input. You probably won't be able to hear this signal, it's the digital equivalent to a "control voltage" in an analog synthesizer. *Control2Audio* is designed to be controlled by another Modulator plug-in. It allows you to record the output of a Modulator in an audio track.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Zero to One Value	0	1		This value, after being scaled and offset by the parameters listed below, is converted to a sample value and sent out the selected audio destination. Changes in the Zero to One Value could be generated by moving the slider or by selecting this parameter as a destination parameter for a Modulator plug-in.
Scaling Factor	0	10		Sets the multiplier on the Zero to One Value before it is converted to an audio signal.
Offset	-1	1		Adds a constant offset to the scaled Zero to One Value before it is converted to an audio signal.
Ramp Time	0	150	ms	At a setting of 0, the audio output signal will be a "stairstep" changing immediately from one value to the next. At higher settings, more and more "lag" is introduced into the signal as it progresses from one value to the next.
Destination	Off	pluggo Bus 4R		Sets the destination for the audio output of the plug-in.

Insights

- The original *raison d'être* for *Control2Audio* was that it might be more efficient to record and playback tracks of control information than to generate them on the fly or to use a sequencer's automation facilities.

- Storing control information as audio will also allow you to use all of the existing audio editing and effecting functions to do interesting (or mysterious) things to the control data.

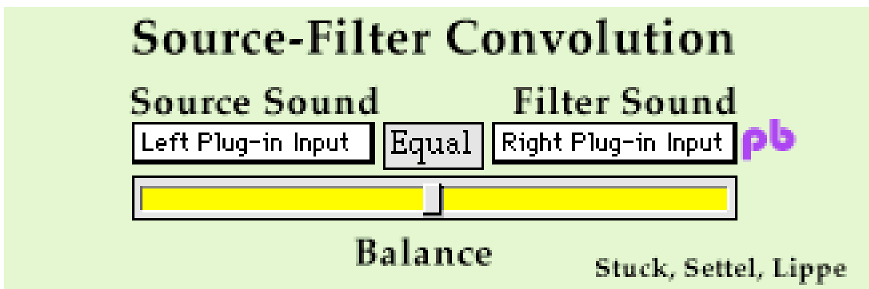
Convolver

category: spectral domain

audio input: mono, stereo, pluggoBus *cpu: heavy* *audio output: mono*

What It Does

Convolver is a real-time cross-synthesis implementation is based on a frequency-domain signal processing technique. Each of the two input signals, the Source Sound and the Filter Sound, are converted to a corresponding frequency-domain signal using the Fast Fourier Transform (FFT). The Source Sound is then convolved (complex multiplied) with the amplitude spectrum of the Filter Sound. The resulting output signal will retain some, or most of the original frequency content of the Source Sound, while its energy (or spectral envelope) will be largely determined by the energy in the Filter Sound. The output depends on the spectral intersection of both signals, so don't try crossing whale songs with piccolos.



Visible Parameters

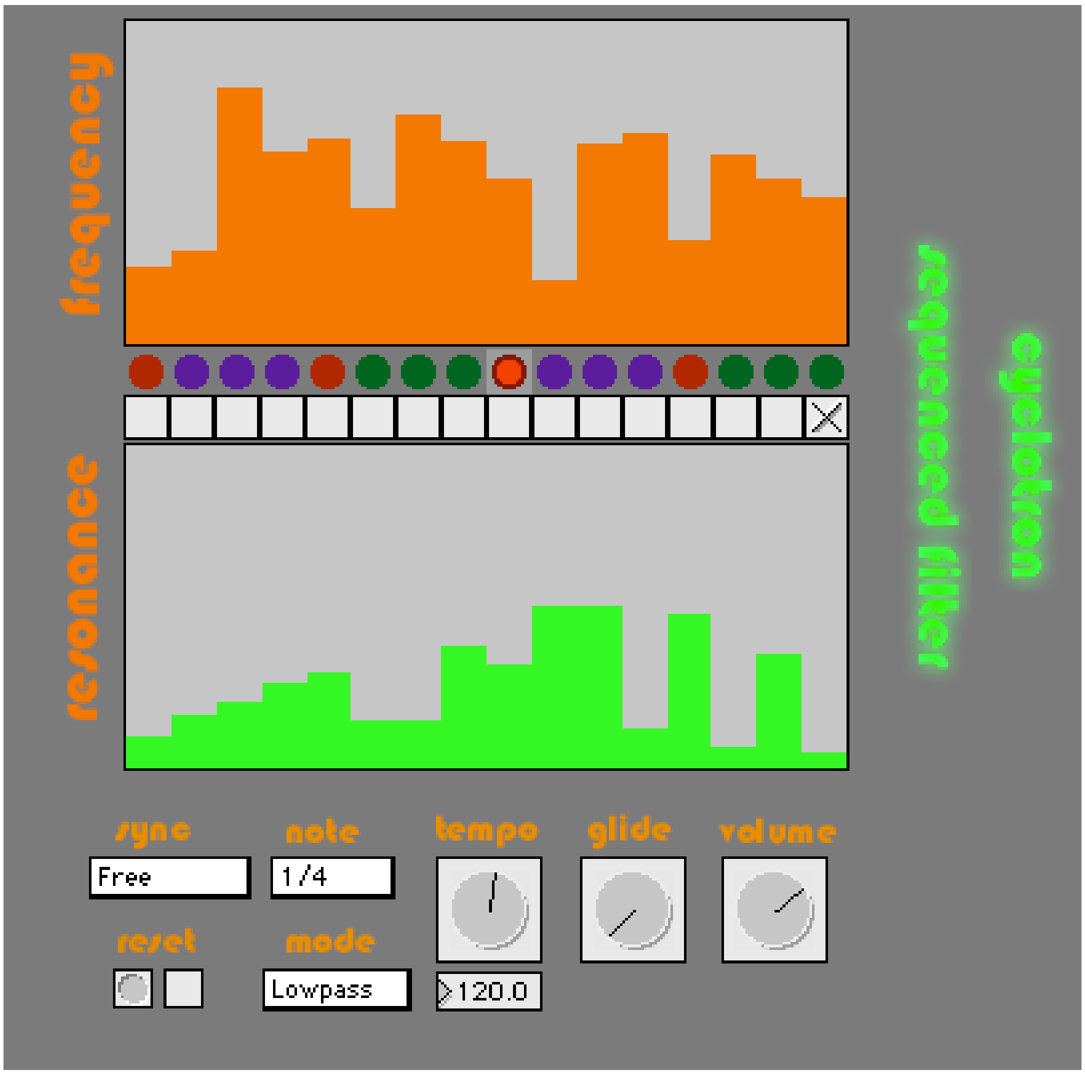
Name	*Min*	*Max*	*Units*	*Description*
Source Sound	Left	pluggo Bus 4R		Selects the source signal for the convolution.
Filter Sound	Left	pluggo Bus 4R		Selects the filter signal for the convolution.
Balance	Source	Filter		This parameter sets a three-way balance between the input Source Sound, the Convolution output, and the input Filter Sound.

Insights

- You can use the same signal as the Source and Filter Sound. It's a charming effect resembling what you might imagine things sound like under water (they don't sound like this underwater the last time we checked).
- In order to achieve better spectral intersection of two sounds, you can "pre-process" one of the inputs. For instance, try the *Frequency Shift* plug-in to move the spectrum of one of the sounds so it intersects with the spectrum of the other.
- This plug-in is particularly nice with a drum track as a Source Sound and signal with a rich spectrum (i.e., something noisy) as a Filter Sound.

What It Does

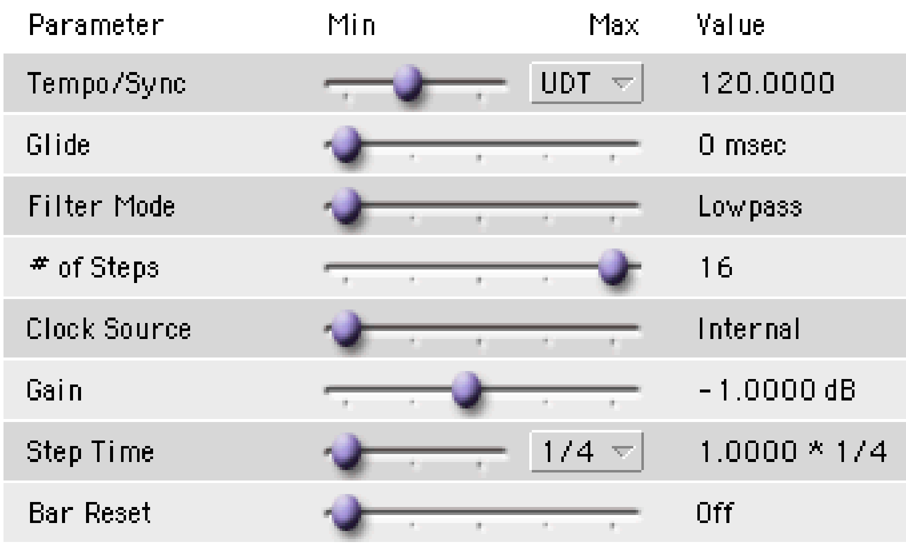
Cyclotron is a sequenced filter. A step sequence controls the filter frequency and resonance of either a lowpass or bandpass filter.



Cyclotron

accepts sync | *category: filter*

audio input: mono | *cpu: light* | *audio output: mono*



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Tempo	10	240	beats per minute	Tempo is displayed as a slider in the Tempo/Sync parameter. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the rate of panning. When in Host and Plug modes, the tempo is an indicator that cannot be changed--it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range to set the speed of the filter sequencer. The Step Time parameter also affects the speed of the sequencer.
Sync	Free	UDT		Sync Mode is a pop-up menu in the Tempo/Sync parameter. There are four modes of sync available: • Free and UDT (User-Defined Tempo) modes let you set the tempo of the filter sequencer yourself. • Host mode synchronizes the filter sequencer to the host tempo. • Plug mode synchronizes the filter sequencer to the beat output of PluggoSync. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Glide	0	500	ms	Sets the amount of time it takes the frequency and resonance parameters to change values. Non-zero values create a gliding effect.
Filter Mode	Low pass	Band-pass		Switches the filter between lowpass and bandpass response characteristics.

Name	*Min*	*Max*	*Units*	*Description*
# of Steps	1	16		Number of steps in the sequence. You can also set this by clicking on the check boxes between the sequencer multisliders.
Gain	-12	12	dB	Sets the output gain of the filter.
Step Time	1	1/64t		Sets the metrical duration of one step in the sequence. For example, if the Tempo setting is 120 bpm, and Step Time is 1/4, the sequencer will play four steps in every measure.
Bar Reset	Off	On		Determines whether the sequencer resets to the first step when the beginning of a bar occurs. This applies only to Host and Plug sync modes where beat and bar information is available.

Insights

- Use either Host or Plug sync mode to synchronize the sequencers to the tempo of your music. For example, you can sync *Cyclotron* to match the tempo of a bass line, and add accents on certain beats, by raising the filter frequency on only those beats.
- Use the Glide parameter to smooth out the transitions between steps, to create LFO-based filter-sweeping effects.

deep bass

category: synthesis

input: MIDI *cpu: light/medium* *audio output: mono*

What It Does

deep bass is a very simple analog-style monosynth with integrated cutoff and pitch sequencers. Both sequencers are tightly synced to the host clock, and the pitch sequencer alters the pitch received via MIDI (as opposed to simply playing its own pitch). Most user controls can be modified using MIDI continuous controllers.

Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Waveform Type	0 (saw)	1 (square)	21	Selects the waveform of the oscillators.

category: synthesis

deep bass

input: MIDI *cpu: light/medium* *audio output: mono*

Name	*Min*	*Max*	*CC#*	*Description*
Waveform Mix	0 (0%)	127 (100%)	14	The deep bass oscillator is a unique 4-oscillator design, with a primary oscillator (at the normal pitch), an octave-down sub-oscillator, and two detune oscillators (at +1 and –1 cents respectively). The Waveform Mix control determine the level of oscillators 2-4 mixed with the primary oscillator for output.
Glide	0 (2 ms)	127 (256 ms)	15	Sets the amount of time for the pitch to reach a new note value.
Filter Type	0 (12 dB)	1 (24 dB)	16	Determines the filter type, with either 12 dB (2-pole) or 24 dB (4-pole) filter types available.
Filter Cutoff	0 (0 Hz)	127 (10 kHz)	17	Sets the cutoff frequency of the filter. Note: This value is only used when the filter sequencer is not active.
Filter Resonance	0 (0.0)	127 (1.0)	18	Sets the resonance value (sometimes called Q or pre-emphasis) of the filter. Resonance values range from 0.0 (no resonance) to 1.0 (self-oscillation).
Filter Envelope Modulation	0 (0%)	127 (100%)	19	Sets the amount that the Filter Envelope modulates the filter cutoff.
Filter Envelope Attack	0 (0 ms)	127 (4000 ms)	22	Sets the filter envelope attack rate. When a note is played on the synthesizer, the filter envelope is triggered, and the attack time determines how long it takes the modulation output to move from zero to its maximum value.
Filter Envelope Decay	0 (0 ms)	127 (4000 ms)	23	Sets the filter envelope decay rate. After the envelope has reached its maximum value (based on the Attack parameter), the decay time determines how long it takes for the modulation output to move back to zero.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.
Sequence Switches	0 (off)	1 (on)	24-25	Turns the individual sequencers on and off.
Sync Rates	0 (32nd's)	2 (8th's)	26-27	Selects the note division represented by each sequencer step. Values are 32nd notes, 16th notes or 8th notes.
Filter Sequence Sliders	0 (0 Hz)	127 (10 kHz)		When a step becomes active, it forces the filter cutoff to move to the slider value. This value overrides the “filter cutoff” knob value. When the sequencer is active, the active step is indicated by a red LED on the slider cap. The filter sequence sliders are not accessible via MIDI, nor by external modulators.

deep bass

category: synthesis

input: MIDI | *cpu: light/medium* | *audio output: mono*

Name	*Min*	*Max*	*CC#*	*Description*
Pitch Sequence Sliders	-12 semitones	+12 semitones		When a step becomes active, it alters the current pitch by the amount of the slider value. The pitch sequence sliders are not accessible via MIDI, nor by external modulators.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.

Insights

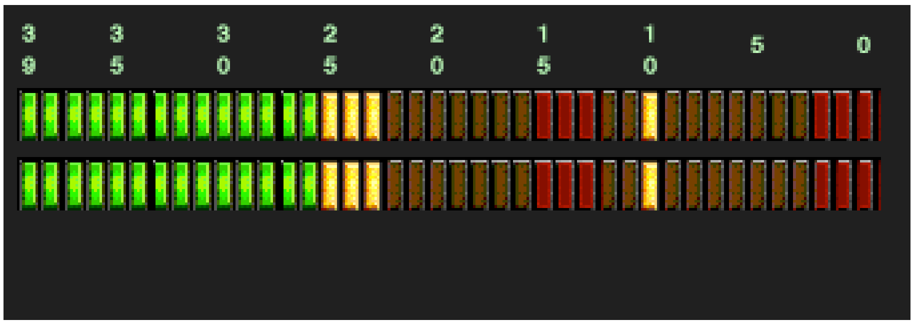
- The deep bass sequencers are tightly synced with your host sequencer's clock. Rewinding or moving the "current time" indicator will cause the sequencer to jump to the current position. In some cases, the sequencer position may seem to be off by one step – this is because some sequencer position themselves "just before" the selected position. For example, if you select the beginning of the fourth measure (4:1:0), the sequencer may actually position the cursor at one tick before that position (3:4:479, for example). This allows for events occurring on the beat to properly sound. Don't worry – the sequencer will catch up just fine…
- Since the pitch sequencer alters the current MIDI note (rather than replacing it), you can use the pitch sequencer to add interest to a static bass sequence. Most of the pitch sequences in the presets were created to liven up a single held note.

D-Meter

audio input: stereo | cpu: light | audio output: stereo

What It Does

D-Meter is a mastering meter indicates average and peak values on one-dB-per-LED scale. Insert into your master mix to optimize both levels at the top of the two yellow sections for maximum loudness. The average indicator (row of LEDs shown on at left) reacts more slowly than a VU meter. The peak indicator (single LED on at right) attempts to work like an AES-EBU digital meter. The exaggerated difference in levels between the two allows each to be displayed more clearly. *D-Meter* was inspired by Dorrough hardware meters.



Insights

- *D-Meter* echoes its input to its output, so it is best used as a stereo insert effect.

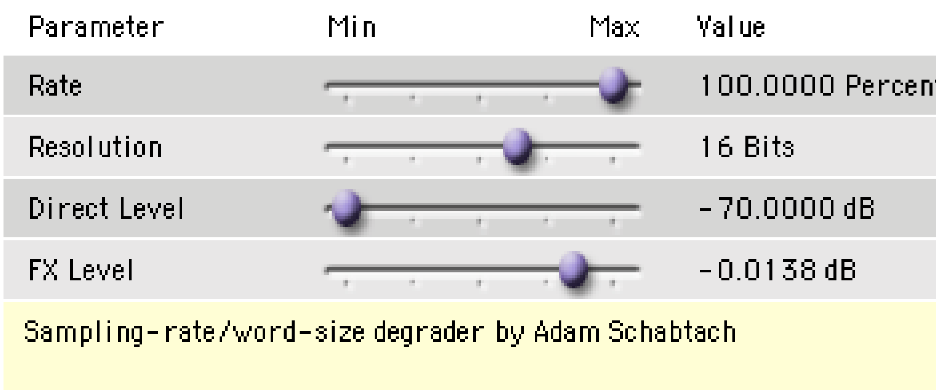
Degrader

category: distortion

audio input: stereo *cpu: light* *audio output: stereo*

What It Does

Degrader reduces the effective sampling rate and bit depth of its input. Since both the sampling rate and bit depth can't be changed by a plug-in, *Degrader* doesn't actually reduce either of these things—it just sounds as if it does.



Visible Parameters

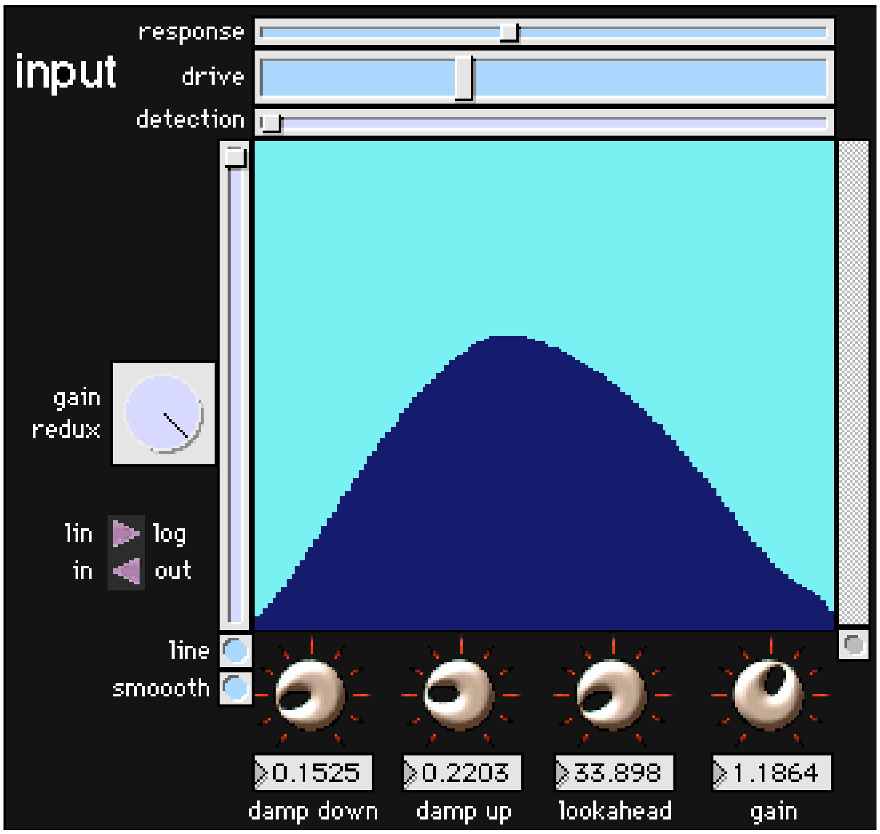
Name	*Min*	*Max*	*Units*	*Description*
Rate	0	100	percent	Sets the sampling rate of the output with respect to the current sampling rate.
Resolution	1	24	Bits	Sets the sample word size. Smaller values will sound both more distorted and louder.
Direct Level	-70	+12	dB	Sets the gain on the unprocessed input signal.
FX Level	-70	+12	dB	Sets the output gain on the effect.

Insights

- *Degrader* can be used to demonstrate why more bits and a higher sampling rate are better. Or worse, depending on your aesthetic.
- Try using harmonically simple signals as input, and setting the sampling rate near or below their fundamental frequency. (Don't bother pulling out your calculator—just mess with the slider.) Interesting sideband aliasing will result.
- Lowering the Resolution to a small number of bits increases the signal's loudness. Why? The signal starts to resemble a full-scale square wave, which has greater average energy than the original signal. Be cautions of your speakers and ears.
- If you really want to lower the audio quality of your source material, try *Degrader* cascaded with *Phone Filter*.
- If you're asking yourself, "why would I want to reduce the sampling rate and/or bit depth of a signal?" you might find that this plug-in isn't of much use to you. But try it anyway. You may find that this plug-in will save you hundreds or thousands of dollars by convincing you that you don't have to upgrade your audio converters to higher quality units quite yet.

What It Does

Dynamical is a versatile gain controller, capable of compressing, expanding, gating, and more. The interface allows you to draw a dynamics response curve and adjust how it is used to process the input signal.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Gain	0	2		Adjusts the overall output level.
Response	0	1		Changes the speed of amplitude detection, from slow and sloppy to fast and tight.
Drive	0	1		Adjusts the level of the input signal.
Damp Down	0	1	“	Adjusts the speed at which *Dynamical* turns down the volume of the input signal. With a compression setting, this is the “attack.”
Damp Up	0	1		Damp Up adjusts the speed at which *Dynamical* turns up the volume of the input signal. With a compression setting, this is the “release.”

Dynamical

category: dynamics

audio input: mono | *cpu: medium* | *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Linear/Log	0	1		Selects the response mode. In general, the minimum Linear setting will be more extreme and edgy, while the maximum Log setting will be more smooth and musical. This is because the perceived loudness of a sound is a logarithmic function of its amplitude
In/Out	In	Out		A bypass switch for comparing the processed and unprocessed audio.
Lookahead	0	250	ms	Determines how far ahead the amplitude detection works. By adding a little bit of lookahead, *Dynamical* can anticipate sudden amplitude peaks.

Interface Elements

- The Response Curve is the graph of input to output volume, as described above. Note that you can draw shapes here as bizarre as you like, not just the usual nuts-and-bolts dynamics.
- The Smooth button lets you smooth out the response curve. Repeatedly hit this and all the edges will grow rounder. This is very useful to take the random edges and caffeine-induced twitches out of a hand-drawn curve.
- The Line button to instantly create a linear response curve—a diagonal line that will result in no change in the amplitude. This is handy as a starting point for creating a new curve.
- The Gain Reduction knob shows you how much *Dynamical* is turning the output volume up or down.

Insights

- It will help you to know the basic principle of a dynamics processor. *Dynamical* analyzes the amplitude level of the input signal, which is displayed by the Detection indicator. It then uses the Response Curve graph to determine whether it should turn the volume of the signal up or down. The graph represents the ratio of the input to the output signal as the amplitude gets louder: input volume is on the x-axis, and output volume is on the y-axis. Therefore, a straight line diagonally from bottom-left to top-right means that the amplitude at every point should be the same, and *Dynamical* does not adjust the volume at all. If the line becomes more horizontal, it is compressing the sound, because the output amplitude is specified as less than the input amplitude, and *Dynamical* will turn the signal down. If the line becomes more vertical, dynamic expansion occurs, which means that the output is being "turned up" faster than the input - so you get an exaggeration of amplitude peaks. Check out some of the presets to get an idea of how different response curves produce different effects to the dynamics of the sound.
- For best results when comparing using the In/Out control, adjust the Gain control so that the processed sound is about the same loudness as the original. If one of them is louder, your perception of them will be biased—probably toward whichever is louder. This is a psychoacoustic effect, so if you want to listen closely to the different character of two similar sounds, make sure they are at the same loudness.
- With compression settings, you usually have to add some Gain to restore the overall loudness of the signal-since *Dynamical* is turning the volume down at higher input levels. As noted above, set the Gain so that the perceived loudness is the same as the input signal if you want to compare them.
- Lookahead response is built in to analog compressors; here you can adjust it, even to unusual extremes. In order for Lookahead to work, it must delay the output signal, so you may need to adjust the audio delays in your host environment if synchronization with other signals is important.

- Expander settings (The *Drum Loop Weed-Out* preset is a good example) can be very useful to remove some of the low-level 'clutter ' in a sound, so you hear only the peaks. Tweak the curve shape and Damp parameters to find a gain response that works for the source material.

easy sampler

category: synthesis

input: MIDI *cpu: medium* *audio output: mono*

What It Does

easy sampler is a simple but effective one-sample playback device. A sample can be selected from your computer, and assigned loop, tuning and root note information. Additionally, an ADSR amplitude envelope allows for level-based effects.

Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Play Button			N/A	Press the Play button to sound the sample at a pitch of E3 (the E above middle C – MIDI note 64). Hitting this button is the equivalent of playing E3 at a velocity of 127 for 1/2 second.
Sample Selector			N/A	This field contains the file name of the sample in use. You can change the sample by clicking on the sample field (which will display an Open File dialog), or by dragging a file onto the field. While only the file name is display, the entire file path is stored with a saved preset.
Loop Toggle	0 (off)	1 (on)	18	If the loop toggle is "on", the file will loop as long as a MIDI note-off message is not received. If the sample file contains defined loop points, they will be used; otherwise, the file will loop from beginning to end.
Root Note	0 (C-2)	127 (G8)	19	Sets the tuning adjustment made by the sample playback engine based on MIDI note values. This value represents the MIDI note value at which no transposition will occur. Note values greater than the root note will be transposed upward, while values lower than the root note will transpose down.

Name	*Min*	*Max*	*CC#*	*Description*
Tuning	0 (- 1 semitone)	127 (+ 1 semitone)	10	Sets the fine tuning of the sample.
Amplitude Envelope Attack	0 (0 ms)	127 (5000 ms)	14	Sets the amplitude envelope attack rate. When a note is played on the synthesizer, the amplitude envelope is triggered, and the attack time determines how long it takes the amplitude to move from zero to its maximum value.
Amplitude Envelope Decay	0 (0 ms)	127 (4000 ms)	15	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the amplitude to move to its sustain level.
Amplitude Envelope Sustain	0 (0%)	127 (100%)	16	Sets the amplitude envelope sustain level. The amplitude will remain at this level until the note is released.
Amplitude Envelope Release	0 (0 ms)	127 (6000 ms)	17	Sets the amplitude envelope release rate. This is the amount of time it takes for the amplitude to return to zero after a note has been released.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

easy sampler

category: synthesis

input: MIDI | *cpu: medium* | *audio output: mono*

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.
Polyphony	1	32	voices	Adjusts the number of available sample playback voices. Reducing this number may reduce CPU use.

Insights

- Although very simple, *easy sampler* takes us back to the original digital samplers – which were often digital delays with some simple storage and tuning options. While somewhat primitive, a single-sample device makes it easy to create wild pitch and modulation variations.
- *easy sampler* is perfect for quick loop playback. Using a loop doesn't require preparation with a looping tool – you just drop it in, and hit a MIDI note to play it back. If there are timing issues, you can adjust the tuning until it matches the remainder of your track.

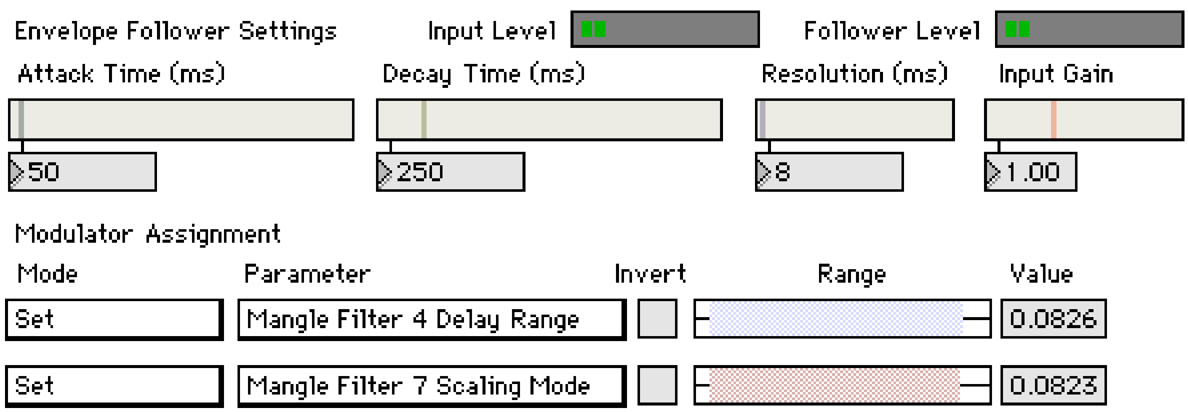
category: modulator

Env Follower

audio input: mono *cpu: light* *audio output: mono*

What It Does

Env Follower tracks the amplitude of its audio input and generates a control signal that can be used to modulate the parameters of other plug-ins.



Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Attack Time	0	2000	ms	Sets the amount of time used to determine when increases in signal amplitude are reflected in the output control signal. Larger values mean *Env Follower* will be slower to respond to the onset of a note or other event in the input signal.
Decay Time	0	2000	ms	Sets the amount of time in used to determine when decreases in signal amplitude are reflected in the output control signal. Larger values mean *Env Follower* will be slower to let go of a note or other event in the input signal.
Resolution	1	500	ms	Sets the rate of output of the envelope follower. Larger values mean less frequent updating of the current amplitude envelope value.
Input Gain	0	3		Sets the gain on the input to the envelope follower.

Env Follower

category: modulator

audio input: mono | *cpu: light* | *audio output: mono*

Parameters for Each Modulation Destination

Name	*Min*	*Max*	*Units*	*Description*
Mode	Off	Scale		When the mode is Set (the most common), the modulator's value directly sets the assigned parameter. When the mode is Offset, the modulator's value is added to the current value of the parameter (before modulation). When the mode is Scale, the modulator's value multiplies the current value of the parameter. When the mode is Off, no changes to the parameter occur.
Assign				Choose a plug-in and a parameter to modulate from this menu. The parameters listed will depend on the plug-ins that are currently inserted.
Invert	Off	On		When checked, the output value is inverted from the signal's envelope amplitude level.
Range Low Value	0	1		Use the range bar to scale the modulator's output data. The low value in the range is the minimum output value sent to the assigned parameter.
Range High Value	0	1		Use the range bar to scale the modulator's output data. The high value in the range is the maximum output value sent to the assigned parameter.

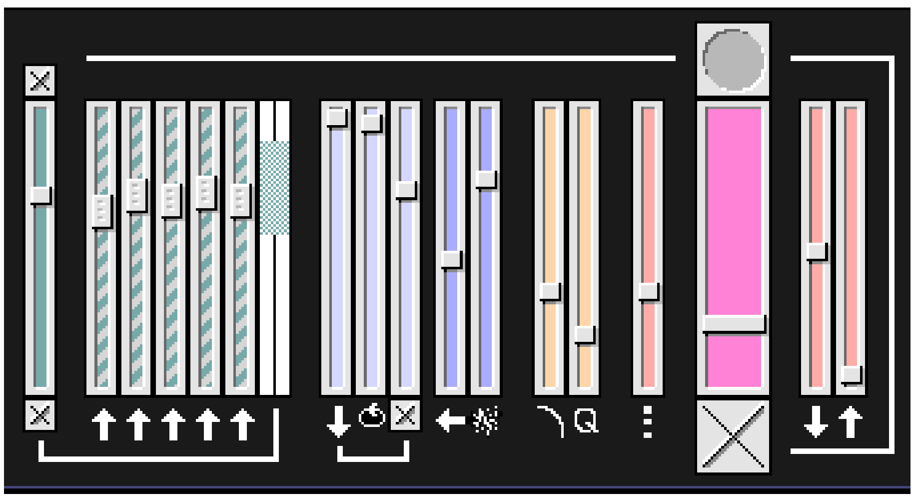
Insights

- In comparison with *Audio2Control*, the *Env Follower* is more useful if you're interested in using the overall amplitude of an audio signal for control purposes. *Audio2Control* translates sampled values from the audio signal into control data, which is not the same thing as an amplitude envelope.

What It Does

Feedback Network consists of five Feedback Units. Each one has an independent bandpass filter and delay line. The output from each Feedback Unit can be routed to any of the other five, at an independent volume. You can't set all these parameters directly; all you do is hit a big Randomize button, and they will be shuffled for you. If you want a different sound, just hit the button again—or set it to Auto-Randomize.

Feedback Network features an Auto-Feedback mode that will gradually turn up the volume of all the Feedback Units until the volume level reaches a threshold, at which point it will turn them down a bit, and keep riding the volume controls to maintain constant feedback. At any point you can adjust the volume parameters manually to provoke the system into activity.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the output gain of the Feedback Network.
Dry Level	0	1		Sets the output gain of the original input signal.
Frequency Scaling	0	127		Scale the cutoff frequencies of the bandpass filters in all Feedback Units.
Global Q	0	127		Sets the resonance characteristic of the bandpass filters in all Feedback Units.
Delay Scaling	0	127		Scale the delay times in all Feedback Units.
Feedback Output	0	127		Sets the overall output of the Feedback Units.
Feedback Gain	0	127		Scales the internal feedback levels of the Feedback Units.
Feedback Sensitivity	0	127		Sets the feedback volume at which the auto-feedback mechanism will start turning down the gain.

Feedback Network

category: distortion

audio input: stereo | *cpu: medium* | *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Feedback Clipping	0	127		Determines the amount of clipping that takes place at the feedback output.
System Randomize Trigger	0	1		Triggers randomization of the cutoff frequencies, delay times, routing, and volume of the Feedback Units.
Auto-Randomize	Off	On		When Auto-Randomize is enabled, the system randomization will be triggered according to the Auto-Randomize Rate.
Auto-Randomize Rate	0	127		Determines how often the system is randomized, when Auto-Randomize is turned on.
Auto-Feedback	Off	On		When Auto-Feedback is enabled, the feedback volume will be adjusted to maintain constant feedback, as described above.
Auto-Feedback Rate	0	127		Sets the speed of the feedback volume detector, when Auto-Feedback is enabled.
Randomize In Levels	Off	On		When Randomize In Levels is enabled, the five Feedback Unit input level controls will be continuously randomized.
Randomize In Levels Rate	0	127		Determines the speed with which the Feedback Unit input level controls are randomized when Randomize In Levels is enabled.
In Levels Min	0	127		Sets the lower limit of the Feedback Unit input levels when Randomize In Levels is enabled.
In Levels Max	0	127		Sets the upper limit of the Feedback Unit input levels when Randomize In Levels is enabled.
Input 1 Level	-76	+18	dB	Determines the level of the input signal sent to Feedback Unit 1.
Input 2 Level	-76	+18	dB	Determines the level of the input signal sent to Feedback Unit 2.
Input 3 Level	-76	+18	dB	Determines the level of the input signal sent to Feedback Unit 3.
Input 4 Level	-76	+18	dB	Determines the level of the input signal sent to Feedback Unit 4.
Input 5 Level	-76	+18	dB	Determines the level of the input signal sent to Feedback Unit 5.

Insights

- Given the variety and complexity of sounds that *Feedback Network* can produce, you might want to record its output (either within your sequencer, or to an external tape deck) and just let it run wild for a while. Afterward you can pick the best bits.
- If you turn off the Auto-Randomize functions, you can manually randomize the network and tweak the parameters (especially Delay Scale) until you get a sound that works well with the source material. At this point, again, you probably want to record the output of *Feedback Network*, since you might have trouble creating the same exact sound again, even with all the parameters saved in your song.

What It Does

filtered drums is an eight-channel sampled drum module. In addition to numerous channel controls, there is a routing system allowing individual channels to be sent through an overdrive and resonant filter circuit.

Visible Channel Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Filter Routing	0 (off)	1 (on)	110-117	If the filter routing button is activated for a drum channel, that output will be routed through the overdrive/filter circuit.
Tuning	0 (.07x)	127 (1.9x)	102-109	Sets the tuning for a drum channel's sample playback.
Attack	0 (5 ms)	127 (2005 ms)	54-61	Sets the amplitude envelope attack rate. When a note is played for a drum channel, the amplitude envelope is triggered. The attack time determines how long it takes the amplitude to move from zero to its full value.

filtered drums

category: synthesis

input: MIDI | *cpu: light* | *audio output: mono/stereo*

Name	*Min*	*Max*	*CC#*	*Description*
Decay	0 (50 ms)	127 (6050 ms)	46-53	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the amplitude to return to zero.
Pan Position	0 (Left)	127 (Right)	24-31	Adjusts the pan setting of each of the drum channels. A value of 64 (a vertical knob orientation) represents panning to the center. Note: Since the Overdrive/Filter output channels are monophonic, the pan control will not affect a channel routed through the filter.
Volume	0 (0%)	127 (100%)	14-21	Adjusts the individual channel levels output.
Load Button			N/A	Press the Load Button to choose a new sample for a given drum channel. An Open File dialog will be displayed, allowing a new sample to be selected.
Trigger Button			N/A	Click on the Trigger Button for a drum sample to audition that channel's settings.
Name Display			N/A	The name display will show the first six characters of each drum channel's sample.

Visible Global Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Channel 3/4 Exclusive	0 (off)	1 (on)	3	If engaged, the samples on channels 3 and 4 will "cut off" the decay of the other channel. This is useful for hi-hat samples (where open hats have to be damped by closed hat hits).
Drive	0 (0%)	127 (100%)	70	Determines the amount of overdrive (soft distortion) applied to signals routed to through filter subsystem.
Filter Cutoff	0 (0 Hz)	127 (10 kHz)	71	Sets the cutoff frequency of the filter.
Filter Resonance	0 (0.0)	127 (1.0)	72	Sets the resonance value (sometimes called Q or pre-emphasis) of the filter. Resonance values range from 0.0 (no resonance) to 1.0 (self-oscillation).
Filter Type	0 (low-pass)	2 (band-pass)	73	Determines the filter type used by the filter subsystem. This setting determines the function of the filter cutoff setting. Values are low-pass, hi-pass and band-pass.
Filter Channel Volume	0 (0%)	127 (100%)	74	Adjusts the mixed-in level of the filter subsystem signal.

Name	*Min*	*Max*	*CC#*	*Description*
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

Insights

- The filter subsystem allows a broad variety of overdrive and filtering effects to be applied to selected drum channels.
- The filter system is monophonic, so any drum channels routed through the filter and overdrive will lose their pan position, and be routed dead-center.
- Using a drum channels amplitude envelope allows you to "tighten" the drum sound. Preset *Woodcock Drums* is an example of shortened decays creating a very tight drum kit.
- The MIDI notes used to fire the drums sounds are:

  Channel 1: 36 (generally kick)
  Channel 2: 38 (generally snare)
  Channel 3: 42 (generally closed hi-hat)
  Channel 4: 46 (generally open hi-hat)
  Channel 5: 40 (generally 2^{nd} snare or percussion)
  Channel 6: 41 (generally low tom)
  Channel 7: 43 (generally mid tom)
  Channel 8: 45 (generally high tom)

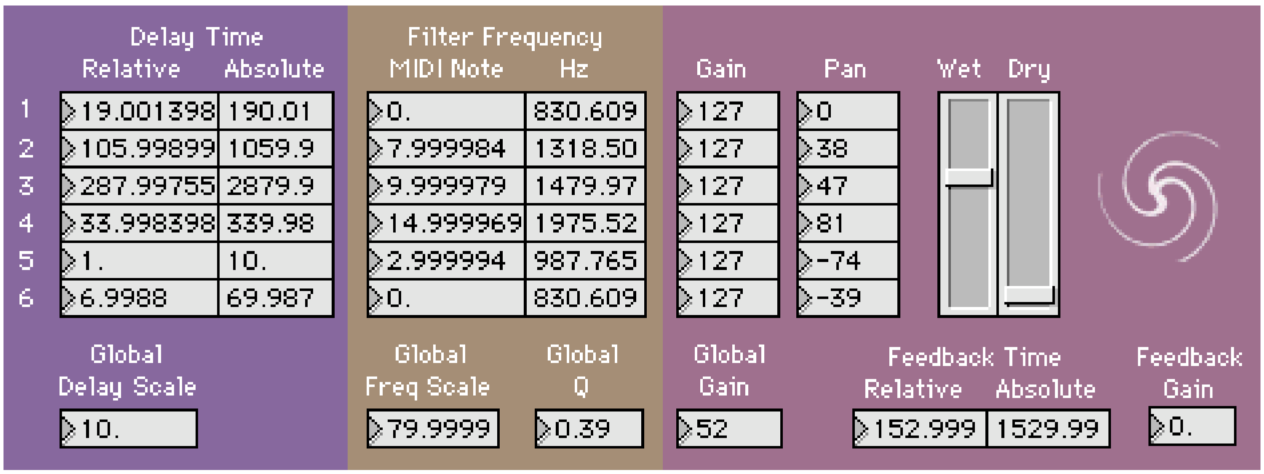
FilterTaps

category: filter/delay

audio input: mono *cpu: medium* *audio output: stereo*

What It Does

FilterTaps gives you a six-tap delay line, with an independent bandpass filter on each tap, as well as control over gain and pan positions. There is also a seventh feedback tap and global scaling factors on all tap parameters. *FilterTaps* is capable of subtle spatialization effects as well as rhythmic sounds that you can tweak to synergize with the input signal.



Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Mix	0	1		Sets the output gain on the processed signal.
Dry Mix	0	1		Sets the output gain on the unprocessed signal.
Global Delay Scale	0	10		A multiplication factor for the time of each tap's delay. The maximum delay time is 5000 ms, and the resulting multiplied value for each tap's delay time that exceeds the maximum value will be clipped.
Global Freq Scale	0	100		A multiplication factor for the base center frequency of each tap's bandpass filter.
Global Q	0	1		Sets the resonance for all bandpass filters.
Global Gain	-300	300		Sets the gain for all the bandpass filters, which can help compensate for the changes in amplitude that different Global Q settings can cause
Feedback Time	1	5000	ms	Sets the delay time of the feedback tap. This time is multiplied by the Global Delay parameter.
Feedback Gain	0	1		Sets the level of the signal in the feedback tap that is fed back into the input of the delay line.

Parameters for Each Delay

Name	*Min*	*Max*	*Units*	*Description*
Band Delay	1	5000	ms	Sets the base delay time for each tap, in milliseconds. This is multiplied by the Global Delay factor to determine the actual delay time, which is indicated just to the right of the base value. The maximum delay time is 5000 ms.
Band Pitch	0	127	note number	Sets the base center frequency of the bandpass filter for each tap, specified as a MIDI Note Number. The frequency of that note is then multiplied by the Global Frequency factor to determine the actual center frequency, which is indicated in Hz just to the right of the base value
Band Gain	0	127		Sets the volume of each tap.
Band Pan/Phase	-127	127		Lets you change the pan position of each tap. A value of 0 pans the output hard left, 127 is hard right, and 64 is directly centered. In addition, you can specify negative values which invert the phase of the signal. Especially with short and closely-spaced delays, phase-inverted signals can create striking stereo effects.

Insights

- As with any multi-tap delay, you can create interesting rhythms by setting the delay times to different values that sync up with the tempo of the source material. Setting the filters to different frequencies, especially with the Q parameter turned up, can accentuate these rhythms. (As an example, consider the preset called *Surfing Indonesia*.) Then, use a Modulator plug-in to modulate the Global Frequency parameter.
- The filter frequencies are represented as MIDI note numbers so that you can easily set them to standard equal-tempered notes. When you scale them with the Global Frequency parameter, it is the MIDI note numbers that are affected, and therefore the filters maintain their relative tuning. In other words, the filter frequencies move on a logarithmic scale, instead of linear.

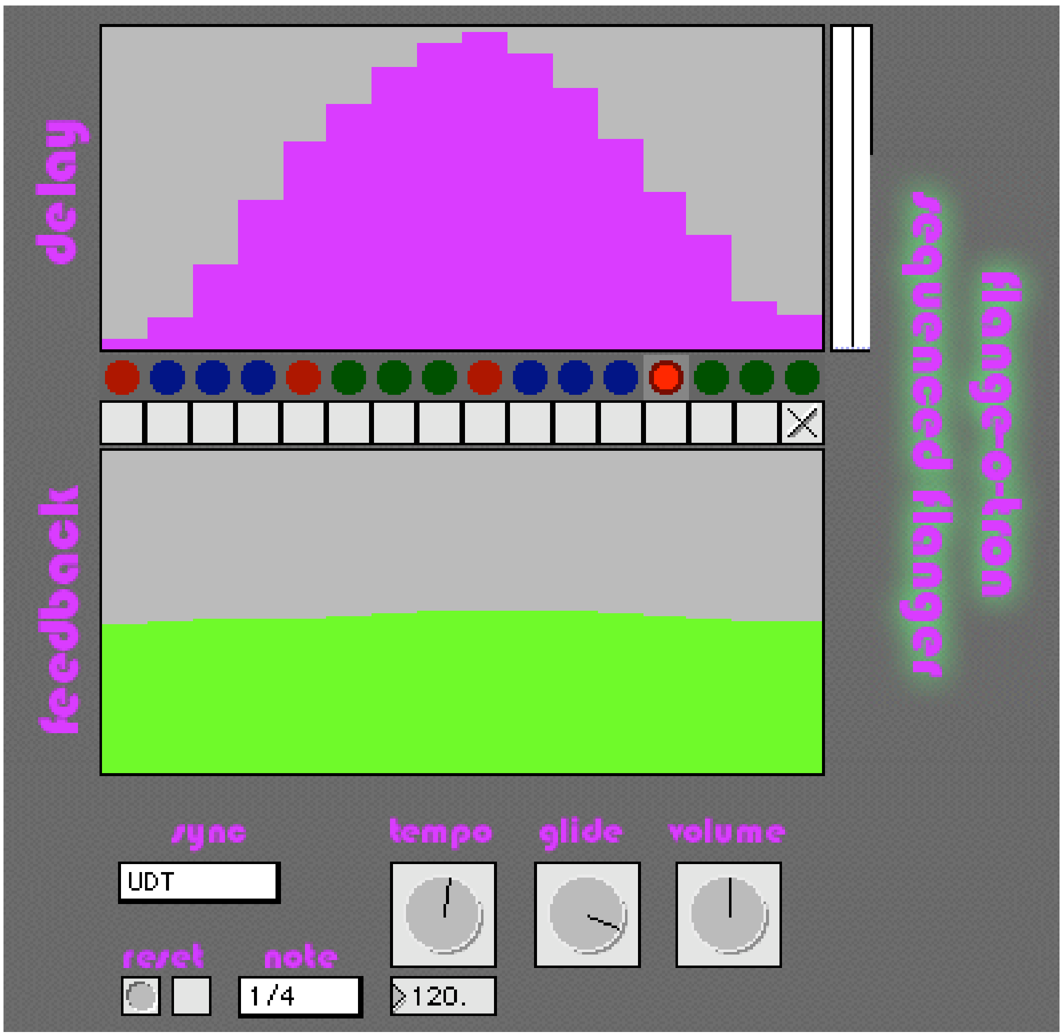
Flange-o-tron

accepts sync | *category: delay*

audio input: mono | *cpu: light* | *audio output: mono*

What It Does

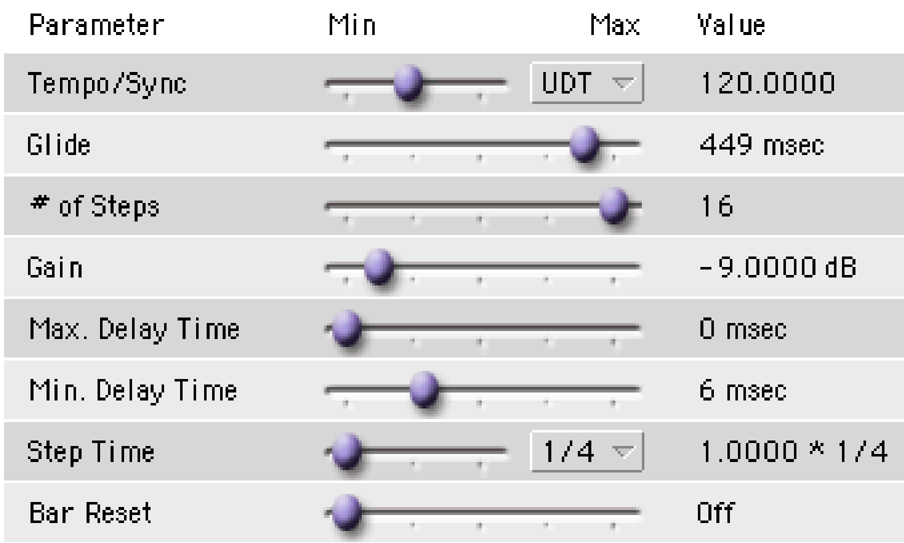
Flange-o-tron is a flanger driven by two step sequencers. Conventional hardware flangers usually have a low-frequency oscillator to vary the flanger's delay time (creating its characteristic sweeping effect), and a knob for varying the amount of feedback (intensifying the flange effect). *Flange-o-tron* has step sequencers for both, allowing you to create flanging effects that would be very difficult to create with traditional hardware flangers.



category: delay *accepts sync:*

Flange-o-tron

audio input: mono *cpu: light* *audio output: mono*



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Tempo	10	240	beats per minute	Tempo is displayed as a slider in the Tempo/Sync parameter. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the rate of panning. When in Host and Plug modes, the tempo is an indicator that cannot be changed--it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range to set the speed of the flanger sequencer. The Step Time parameter also affects the speed of the sequencer.
Sync	Free	UDT		Sync Mode is a pop-up menu in the Tempo/Sync parameter. There are four modes of sync available: • Free and UDT (User-Defined Tempo) modes let you set the tempo of the flanger sequencer yourself. • Host mode synchronizes the flanger sequencer to the host tempo. • Plug mode synchronizes the flanger sequencer to the beat output of PluggoSync Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Glide	0	500	ms	Sets the amount of time it takes the delay and feedback parameters to change values. Non-zero values create a gliding effect.
# of Steps	1	16		Number of steps in the sequence. You can also set this by clicking on the check boxes between the sequencer multisliders
Gain	-12	12	dB	Sets the output gain of the flanger.

Flange-o-tron

accepts sync | *category: delay*

audio input: mono | *cpu: light* | *audio output: mono*

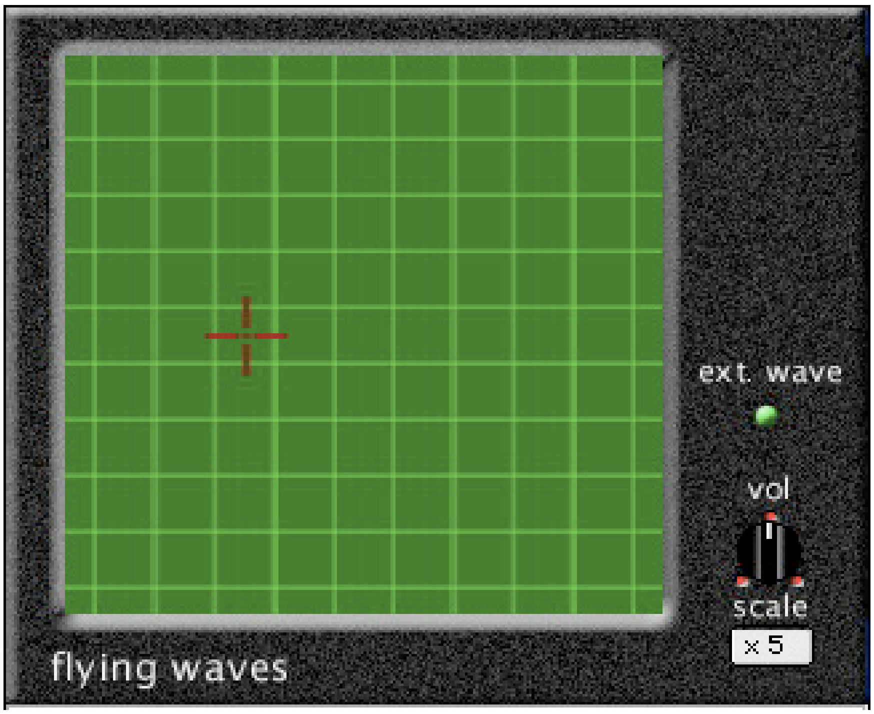
Name	*Min*	*Max*	*Units*	*Description*
Note Value	1/1 (whole note)	1/32		Sets the metrical duration of one step in the sequence. For example, if the Tempo setting is 120 bpm, and Note Value is 1/4, the sequencer will play four steps in every measure. This control has no effect if the Clock Source is a PluggoSync source.
Gain	-12	12	dB	Sets the output gain of the flanger.
Min. Delay Time	0	20	ms	Sets the minimum delay time of the flanger's delay line.
Max. Delay Time	0	20	ms	Sets the maximum delay time of the flanger's delay line.
Step Time	1	1/64t		Sets the metrical duration of one step in the sequence. For example, if the Tempo setting is 120 bpm, and Step Time is 1/4, the sequencer will play four steps in every measure.
Bar Reset	Off	On		Determines whether the sequencer resets to the first step when the beginning of a bar occurs. This applies only to Host and Plug sync modes where beat and bar information is available.

Insights

- The *Basic Flanger* preset simulates the traditional up-and-down flanging effect by using the Glide parameter to smooth the stepped delay times. The *Up* and *Down* presets are similar, but they change the delay time slowly in one direction and rapidly in the other.
- The *Kinda Queasy* preset illustrates how flanging can create large timbral changes if the feedback level is high.
- Set the Sync mode parameter to Host or Plug to synchronize the sequencers to the tempo of your music. For example, you can sync *Flange-o-tron* to match the tempo of a drum loop, and add dramatic flanging effects only on certain beats, by raising the Feedback level on only those beats.

What It Does

flying waves is the virtual Thermion that you've always dreamed about – swooping howls, but without having to get out of the chair! In addition to volume and pitch, you have control of the pitch scaling, and can use the Thermion interface to drive any sampled waveform.



Interface Elements

- The Play Area: Using your mouse, drag the crosshair cursor around the gridded play area. Vertical movement will change the pitch (subject to scaling by the Scale setting), while horizontal movement will alter the output level. These movement can also be generated using MIDI Controls - Controller #6 (Data Entry) will change the pitch, while Controller #1 (Mod Wheel) will control the output level.

flying waves

category: synthesis

input: MIDI | *cpu: light* | *audio output: mono*

Visible Parameters

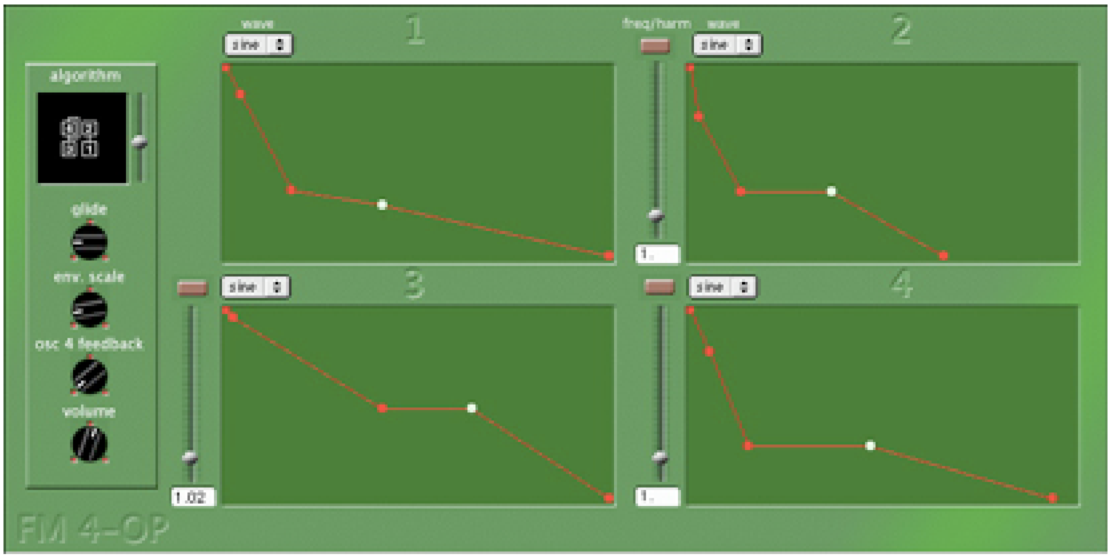
Name	*Min*	*Max*	*Description*
Pitch/Volume Grid			This is the main user interface to the Thermion emulation. The control grid is manipulated by dragging the mouse over its surface – the crosshair cursor will follow the mouse movement. Vertical movement will vary the pitch (based on the Scale setting), while horizontal movement will alter the level. These movements can also be done using MIDI Controllers, with CC#1 (mod wheel) controlling the volume and CC#6 (data slider) controlling the pitch. These controllers were chosen because the are present on most inexpensive MIDI keyboards.
External Waveform Selector			By pressing the button labeled "ext. wave", you can select any waveform as the basis for audio output. Pressing the External Waveform button, a File Open dialog appears, allowing you to select any audio file on your system.
Volume	0 (0%)	127 (100%)	Sets the global level of the output signal.
Scale	0 (x10)	4 (x1)	Sets the pitch range provided by vertical movement. The valid ranges are 10x, 5x, 4x, 2x and 1x, where the range in measured in octaves (i.e., 10 is 10 octaves, 2x is two octaves).

Insights

- Using MIDI controllers, rather than the mouse, will allow you to make more realistic Thermion melodies. A bit of pitch and level vibrato will help you Rock More…
- Adam Schabtach, master of the Way That Is Pluggo, recommends a combination of *flying waves* with a side order (insert effect) of the *Resonation* effect for instant B-Movie madness. Of course, since Adam developed *Resonation*, you should adjust this recipe to taste.

What It Does

fm 4-op is an experimenter's kit for FM programming. Unlike dealing with those old two line displays, you are now able to play with FM as the gods intended – using graphic envelopes!



Interface Elements

- The four-segment envelope editor provides on-screen manipulation of the amplitude output of each operator. The "white dot" represents the "sustain" level. The envelopes are not accessible with MIDI Controllers or external modifiers.

Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Algorithm	0	7	14	The standard 4-operator FM system contains eight algorithms – output routings that provide the mechanism for sound design. This selector allows you to select the appropriate routing for the sound you will make.
Glide	0 (2 ms)	127 (256 ms)	23	Sets the amount of time to move from an existing pitch to a new one.
Envelope Scale	0 (0 ms)	127 (25400 ms)	27	Sets the duration represented by the envelope editing displays. Since the envelope can be shorter than the full length of the display, the Envelope Scale setting provides a relative adjustment to envelope duration.

fm 4-op

category: synthesis

input: MIDI | *cpu: medium/heavy* | *audio output: mono*

Name	*Min*	*Max*	*CC#*	*Description*
Oscillator 4 Feedback	0 (0%)	127 (100%)	22	Sets the amount of feedback used by the fourth operator. Operator feedback is used to create a very complex waveform out of simpler waves.
Volume	0 (0%)	127 (100%)	7	Sets the global level of the output signal.
(each operator)				
Waveform	0 (sine)	4 (pulse)	15-18 (1-4)	Determines the waveform used for each operator.
Frequency / Harmonic Toggle	off (harm)	on (freq)	19-21 (2-4)	Sets the function of the frequency slider on operators 2-4. When off (unlit), the slider will set the operator's tuning to a frequency relative to the incoming MIDI pitch. When set on (lit), the slider will set the operator pitch to a fixed frequency.
Frequency / Harmonic Slider	0 harm=0x freq=0Hz	500 harm=10x freq=10kHz	24-26 (2-4)	Sets the tuning of operators 2-4. When in harmonic mode, the values range from 0 (no harmonic) to the 10th harmonic (pitch x 10). When in frequency mode, the values range from 0 Hz to 10 kHz.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.

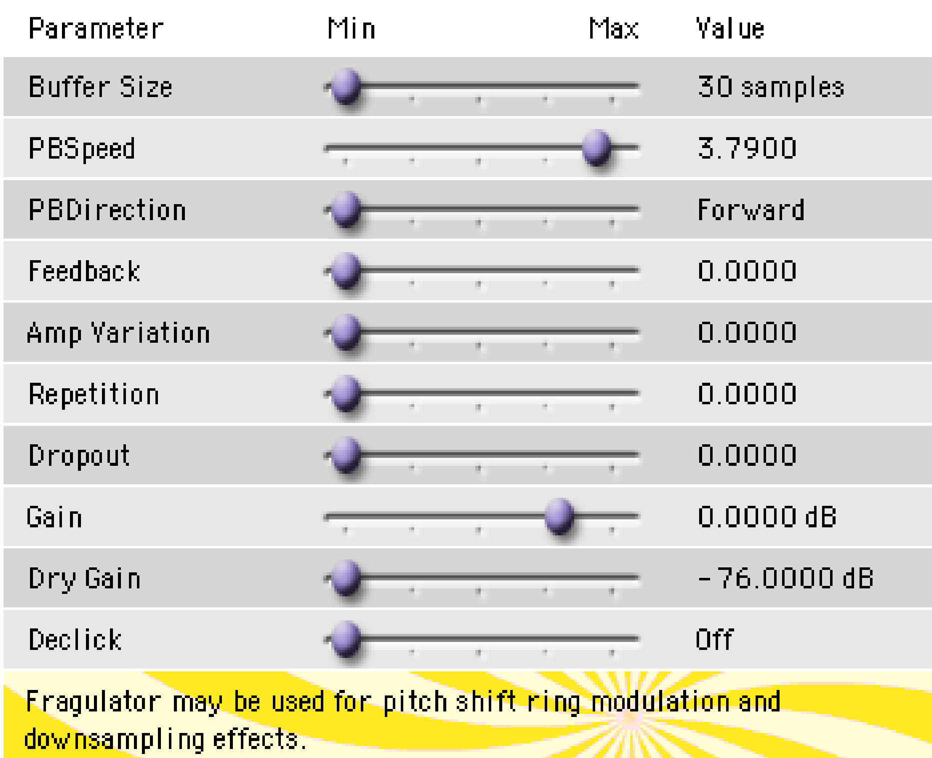
Insights

- *fm 4-op* makes FM programming fun – and educational, too! FM programming is one of the most difficult to master, but the visualization tools available in this synthesizer (along with the example presets) can help you come to grips with this tough task.
- FM synthesizers can sound a little "thin" without the application of effects. Cycling '74 official effect recommendations include *Tapped Delay*, *Space Echo* and *Very Long Delay*.

category: distortion		**Fragulator**
audio input: mono	*cpu: light*	*audio output: mono*

What It Does

Fragulator chops the input signal into pieces (fragments) specified by a Buffer Size, and then loops each fragment either forwards or backwards at varying speeds. Once a number of samples equal to the Buffer Size has been played, a new buffer begins playing back. The result is a pitch-shifting sampler that introduces digital distortion effects, so it's not for people looking for a vintage sound.



Parameters

Name	*Min*	*Max*	*Units*	*Description*
Buffer Size	6	1024	samples	Sets the size of the buffer to be recorded and played back, as well as the number of samples until the playback switches to the next buffer.
PB Speed	0	2		Sets the speed as a multiple of the original. At any speed other than the original, you'll hear digital artifacts such as downsampling or modulation.
PB Direction	fwd	rev		Sets the direction of playback speed. Forward plays the sample from beginning to end, and Reverse plays the sample from the end back to the beginning.

Fragulator

category: distortion

audio input: mono *cpu: light* *audio output: mono*

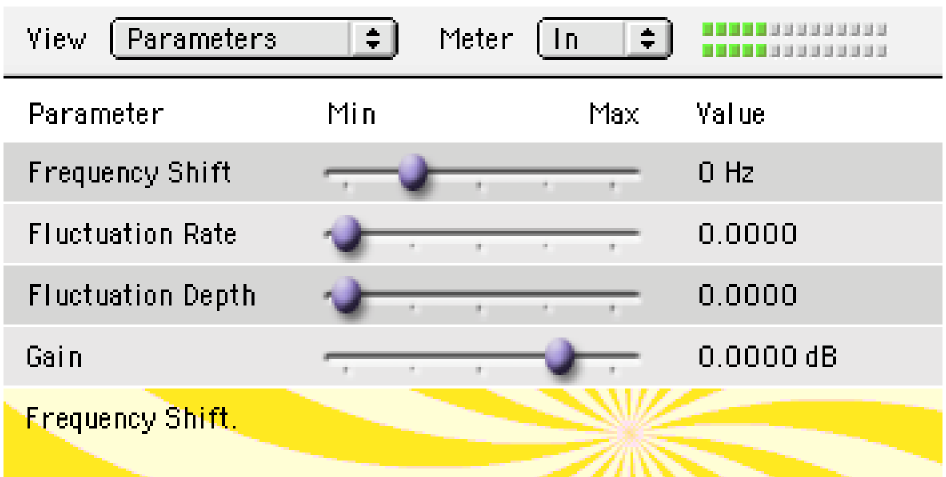
Name	*Min*	*Max*	*Units*	*Description*
Feedback	0	0.99		Sets the amount of the effect's output that is fed back to the input before recording a buffer of samples for playback. The delay on the feedback is the same number of samples as the Buffer Size.
Amp Variation	0	1		Sets the amount of random modulation of the amplitude, changed every Buffer Size samples.
Repetition	0	1		Sets the probability of reusing the previous playback buffer rather than grabbing a new one from the input signal.
Dropout	0	1		Sets the probability that silence will be played back rather than the playback buffer.
Gain	-76	+18	dB	Sets the gain on the processed signal.
Dry Gain	-76	+18	dB	Sets the gain on the unprocessed input signal.
Declick	Off	On		When On, an enveloped muting out of the output occurs when the buffer size parameter is changed.

Insights

- With smaller buffer sizes, the effect is similar to bit-depth reduction (see *Degrader*); the *Metal Filings* preset demonstrates this.
- With larger buffer sizes, you can produce pitch shift and ring modulation effects.
- The *RoboDuck* preset uses a high feedback level to achieve its transformation of the input into a cuddly alien waterfowl.

What It Does

Frequency Shift performs frequency (not pitch) shifting—in other words, all components of the spectrum of the input signal are shifted by a fixed amount, turning harmonic spectra into inharmonic ones. The effect also features a Fluctuations control that introduces random jitter into the shifting process.



Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Frequency Shift	-1000	3000	Hz	Amount by which the frequencies of the input are shifted.
Fluctuation Rate	0	1		Sets the rate at which random changes in the frequency shift occur.
Fluctuation Depth	0	4		Sets the amount of at which random changes in the frequency shift occur.
Gain	-76	+18	dB	Sets the gain on the output signal.

Insights

- We encourage the use of *Frequency Shift* on vocals.
- At small (but non-zero) values of Fluctuation Rate and Fluctuation Depth, you can achieve subtle warble effects. Take that record player into the repair shop!

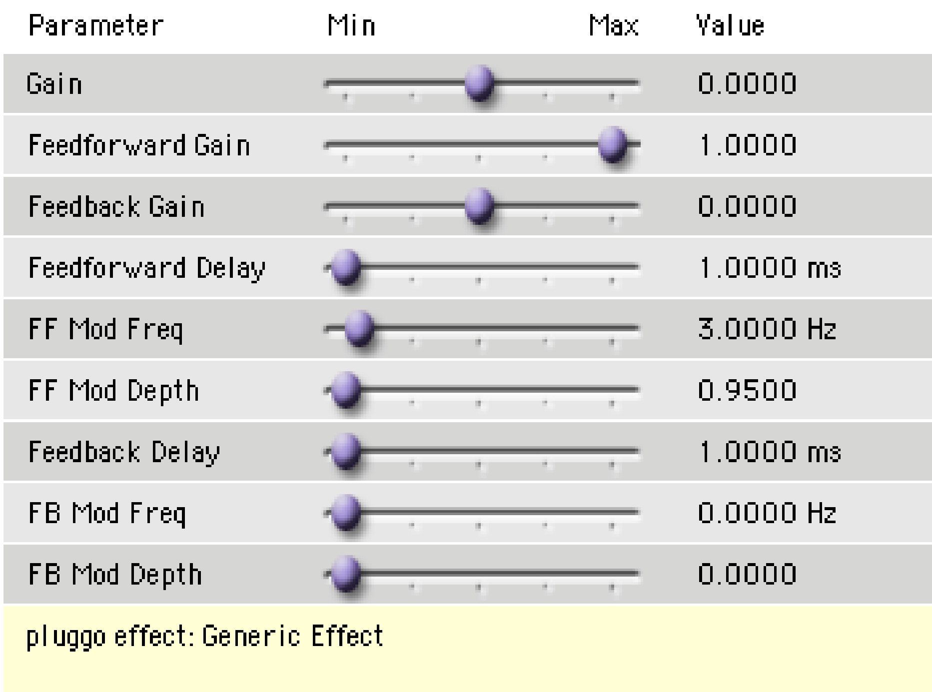
Generic Effect

category: delay

audio input: mono | *cpu: light* | *audio output: mono*

What It Does

Generic Effect is a modulated comb filter that can be configured for vibrato, chorus, flange, and other "industry standard" effects. The difference between these effects is merely a matter of modulating the comb filter differently. Beyond its standard presets, *Generic Effect* can be configured for effects that defy categorization.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Gain	-1	1		Reduction in the gain of the undelayed input signal. Negative values invert the input's phase.
Feedforward Gain	-1	1		The level of the delayed input signal. Negative values invert the delayed signal's phase.
Feedback Gain	-1	1		The level of the delayed output of the comb filter. Negative values invert the signal's phase.
Feedforward Delay	0	100	ms	Base delay time before modulation of the feedforward delay. Values below the amount of modulation may clip the delay at 0 ms producing unwanted clicks.
FF Mod Freq	0	50	Hz	Frequency of modulation of the feedforward delay time.
FF Mod Depth	0	100		Amount of modulation of the feedforward delay time.

Name	*Min*	*Max*	*Units*	*Description*
Feedback Delay	0	100	ms	Base delay time before modulation of the feedback delay. Values below the amount of modulation may clip the delay at 0 ms producing unwanted clicks.
FB Mod Freq	0	50	Hz	Frequency of modulation of the feedback delay time.
FB Mod Depth	0	100		Amount of modulation of the feedback delay time.

Insights

- The data for the first five presets is taken from "Effect Design: Part 2" by Jon Datorro, published in the *Journal of the Audio Engineering Society.* Datorro observes that modulating the feedback delay time is not typically as useful in conventional effects as modulating the feedforward delay.
- Feedback delay modulation can produce your more exotic effects, such as in the *Mind of its Own* preset, which sweeps the feedback delay very slowly through a large range of values—and correspondingly, a large sonic territory.

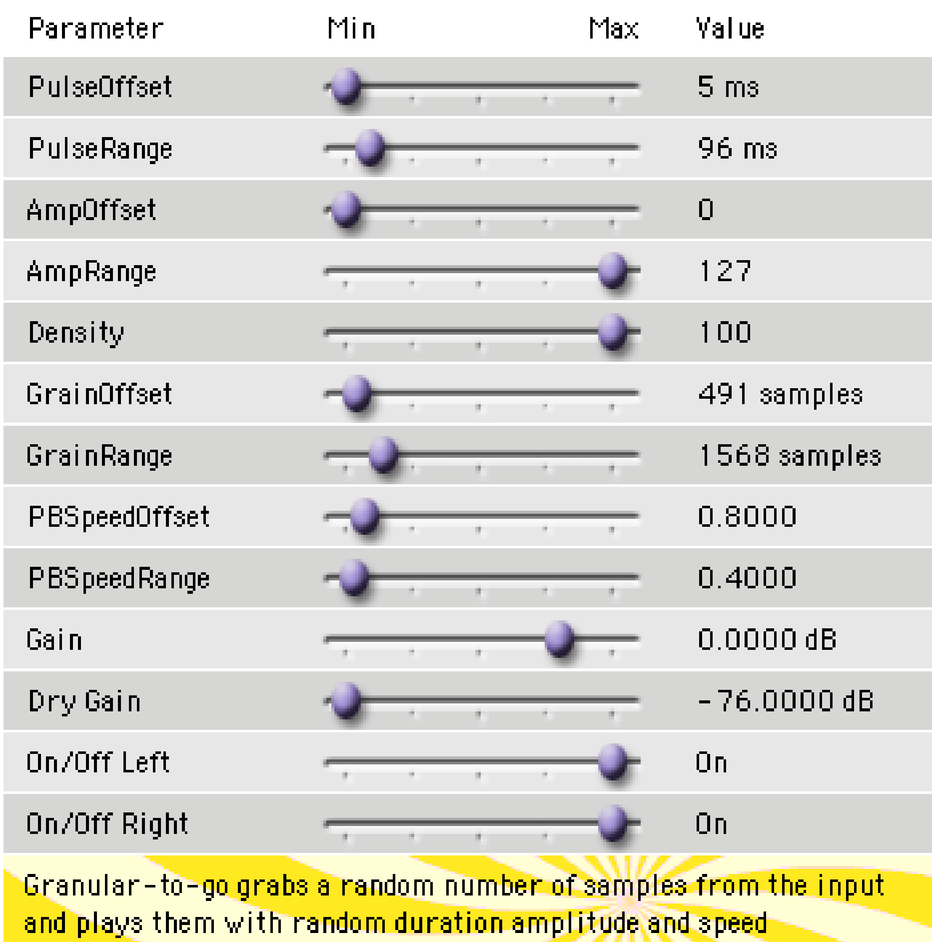
Granular-to-Go

category: granular

audio input: mono *cpu: medium* *audio output: stereo*

What It Does

Granular synthesis plays back short excerpts —called *grains*—from a source. The length of the grains and the variety of ways they are played back can create a variety of timbres, from stuttering vocalizations to purely abstract timbres. In the case of the granular synthesizers described here, the source material is the input to the plug-in. *Granular-to-Go* is a granular synthesizer that allows you to randomize—or not, depending on your mood—the basic parameters involved in generating a granular playback event. Don't forget to try the other Pluggo Signature Series Granular Synthesizers, *Wheat* and *Rye*.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
PulseOffset	5	1000	ms	Minimum duration of a grain playback event, or Pulse. The actual playback duration for each event is determined by a random process that generates durations between the PulseOfffset and the PulseOffset plus the PulseRange.
PulseRange	0	1000	ms	Sets the range within which random grain playback durations are generated for each event.
AmpOffset	0	127	-	Minimum volume scaling of a grain playback event. The actual volume scaling for each event is determined by a random process that generates durations between the AmpOffset and the AmpOffset plus the AmpRange.
AmpRange	0	127	-	Sets the range within which random volume scalings are generated for each event.
Density	0	100	percent	Sets the percent chance that a Pulse is heard. At 0 percent, the output of the plug-in is complete silence, while at 100 percent, all of the generated events are heard.
GrainOffset	1	11025	samples	Minimum duration of the source material for a grain playback event. If the source duration is shorter than the total grain playback event duration, the source sample will be looped. The actual source duration for each event is determined by a random process that generates durations between the GrainOfffset and the GrainOffset plus the GrainRange.
GrainRange	0	11025	samples	Sets the range within which random source durations are generated for each event.
PBSpeedOffset	0	10	-	Sets the minimum rate at which new grain playback events are generated. The actual playback interval between each event is determined by a random process that generates intervals between the PBSpeedOffset and the PBSpeedOffset plus the PBSpeedRange.
PBSpeedRange	0	10	-	Sets the range within which random playback intervals between events are generated for each event.
Gain	-76	+18	dB	Sets the level of the processed signal.
Dry Gain	-76	+18	dB	Sets the level of the unprocessed input signal.
On/Off Left	Off	On		Turns the left output on and off.
On/Off Right	Off	On		Turns the right output on and off.

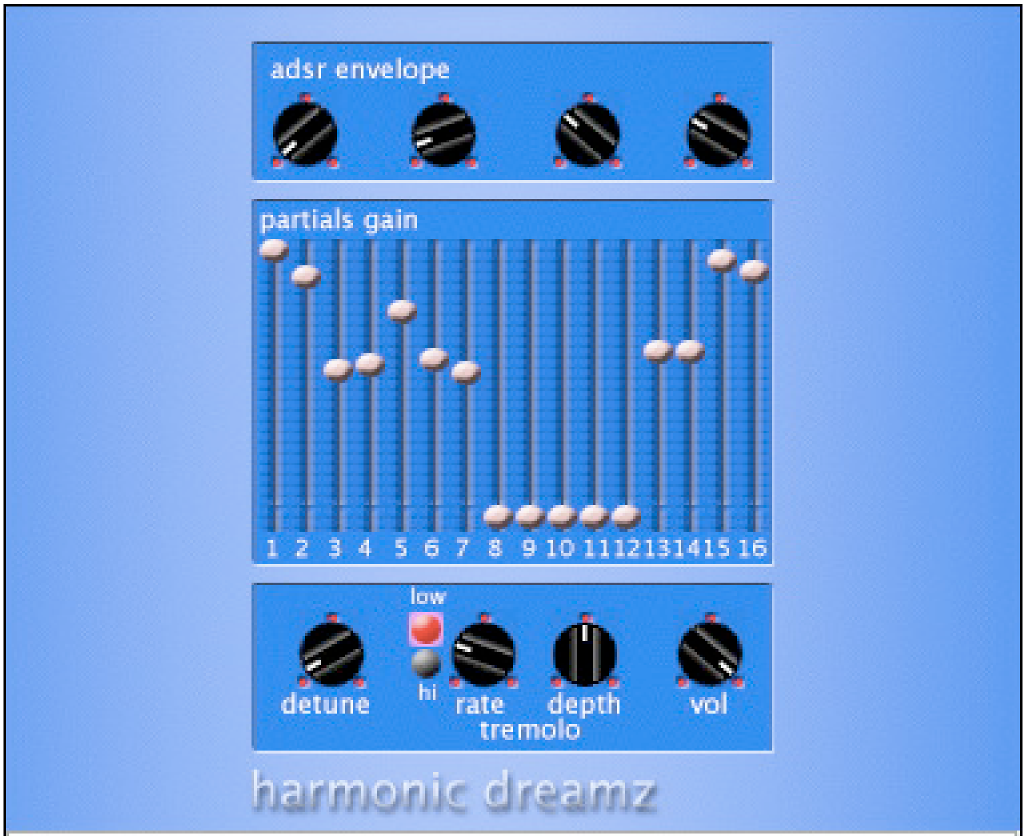
Insights

- You can think of the following analogies for the terms used to name the parameters. The Pulse is similar to a note event. The Amp affects its envelope. The Grain is similar to a looped sample used to generate the note. And the PBSpeed is the rate at which the notes are played.

- When all of the range parameters (PulseRange, AmpRange, GrainRange, and PBSpeedRange) are set to zero, you can produce predictable pitch shift, ring modulation, and rhythmic effects. Using non-zero range values randomizes these parameters, creating a "cloud of sound" effect—the dynamics generally occur too fast and too unpredictably for our ears to follow, and the result the auditory equivalent of a visual blur.
- The GrainOffset and GrainRange parameters affect the recognizability of the source material. When these values are small, the effect is playing back tones whose timbre is set by waveforms in the source material. As the Grain parameters are increased, long excerpts from the input material are used, and you will begin to recognize the material.

What It Does

harmonic dreamz is an additive synth that exposes the first 16 partials of the primary tone. This allows for wide ranging sound possibilities with harmonic tonality. This synthesizer also includes a random detuner, a tremolo system (with both LFO and audio-rate oscillation) and an amplitude envelope for sound shaping.



Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Envelope Attack	0 (0 ms)	127 (5000 ms)	14	Sets the amplitude envelope attack rate. When a note is played for a drum channel, the amplitude envelope is triggered. The attack time determines how long it takes the amplitude to move from zero to its full value.

harmonic dreamz

category: synthesis

input: MIDI | *cpu: medium* | *audio output: mono*

Name	*Min*	*Max*	*CC#*	*Description*
Envelope Decay	0 (0 ms)	127 (4000 ms)	15	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the amplitude to move to its sustain level.
Envelope Sustain	0 (0%)	127 (100%)	16	Sets the amplitude envelope sustain level. The amplitude will remain at this level until the note is released.
Envelope Release	0 (0 ms)	127 (6000 ms)	17	Sets the amplitude envelope release rate. This is the amount of time it takes for the amplitude to return to zero after a note has been released.
Partials Gain Sliders	0 (0%)	127 (100%)	102-117	Sets the level of each of the first 16 partials (harmonics) of the main tone.
Detune Depth	0 (0 Hz)	127 (+/- 50 Hz)	18	The Detune control adds a random tuning factor to a randomly selected partial. Every 1/10th second, one of the partials will be detuned by up to 50Hz.
Tremolo Range Selector	0 (Low)	1 (Hi)	21	Sets the rate range of the Tremolo system. Low range will provide rates from 0.01 - 20 Hz, while Hi range is 0.01 - 10kHz.
Tremolo Rate Control	0 (0.01 Hz)	127 (low:20 Hz hi: 10 Khz)	20	The Tremolo system provides amplitude variations using a sinewave oscillator. The range selector determines the available rates - low provides standard LFO rates, while Hi extends into the AM (ring modulation-like) effects.
Volume	0 (0%)	127 (100%)	7	Sets the global level of the output signal.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.

Insights

- Working with high or odd-numbered partials will provide a rougher, somewhat inharmonic sound. It's great for creating sounds that are tonal, but somewhat odd. Presets *24-hr Plumbing* and *Paradox* are examples of this type of programming.
- Traditional additive synthesis will often use modulation controls on each partial to create evolving tones. While harmonic dreams does not have partial modulators, you could utilize the Pluggo modifiers (such as

LFO and Envelope Follower) pointing at individual partials (labeled part_1 to part 16) for more complex sound design.

Harmonic Filter

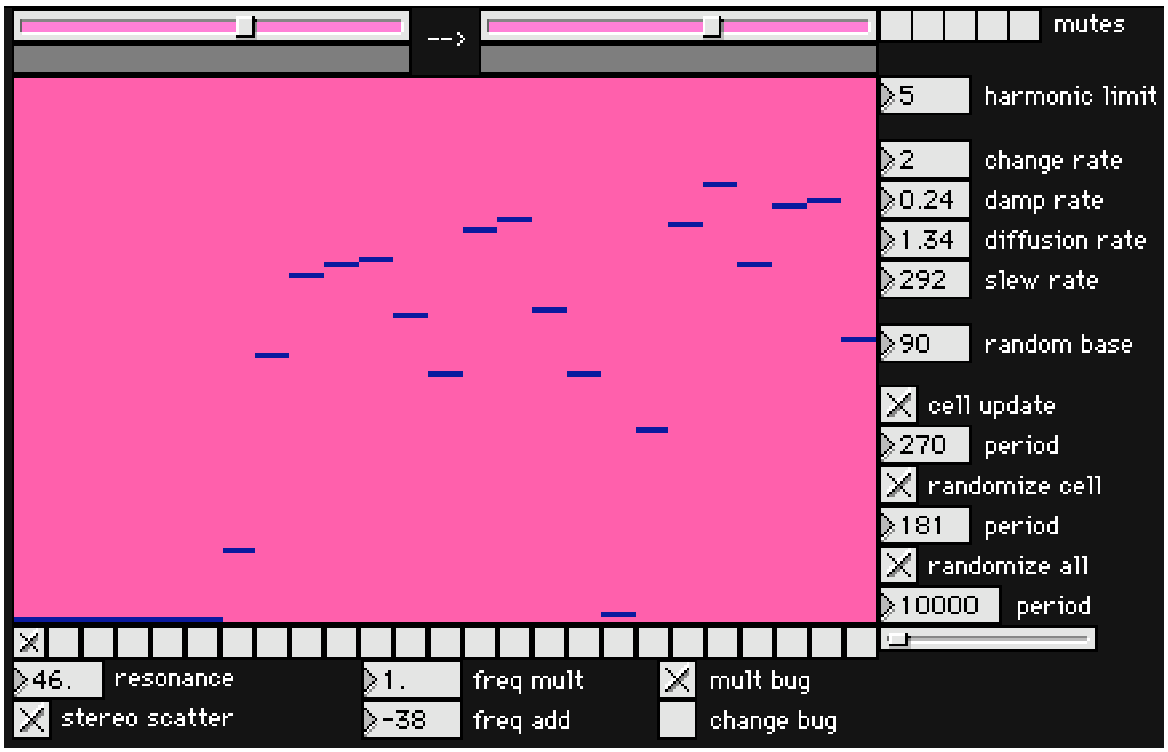
category: filter

audio input: mono *cpu: medium - heavy (adjustable)* *audio output: stereo*

What It Does

Harmonic Filter consists of 25 band-pass filters, and a cellular automata (CA) algorithm to control them. The CA causes the frequency of each filter to move toward a harmonic of the frequency of its neighbor. This results in complex filters that can be tuned to pure harmonics. With variable speed and randomization settings, *Harmonic Filter* can produce undulating, perpetually moving sonic effects.

Harmonic Filter is based on the original Max/MSP application, *CellSound*, by jhno/Delicate Ear.



Interface Elements

- Multislider: The vertical position of each slider represents the cutoff frequency of a band-pass filter. There are 25 of them, arranged from left to right. Unless Stereo Scatter is enabled, they will be positioned in the stereo field according to their position on the screen. When the filter frequencies are being changed by the cellular automata, or randomization parameters, their positions on the screen will change. You can also set them with the mouse, either adjusting one slider, or sweeping through entire groups.
- Anchors: Below each of the 25 filter sliders is a checkbox. If you enable it, the frequency for that filter will be fixed. This means the cellular automata, mult, offset, and randomization parameters will not affect it. This is useful for tuning filters to the pitch of the input material.
- Mutes: The five check boxes at the upper right allow you to turn off the filters in groups of five. This changes the sound, but it also conserves CPU resources.

audio input: mono *cpu: medium - heavy (adjustable)* *audio output: stereo*

Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Input Gain	0	199		Adjusts the level of the incoming audio signal.
Output Gain	0	199		Adjusts the level of the filtered signal output.
Resonance	0	200		Sets the resonance characteristic for all the band-pass filters.
Stereo Scatter	0	1		With this turned on, the 25 filters will be placed randomly in the stereo field. Otherwise, they will be placed from left to right, according to their position on the screen.
Frequency Multiplier	0	8		Scales the cutoff frequencies of the band-pass filters. This has the effect of global pitch transposition, and can be useful to tune a filter configuration to the key of the input. Set this to 1 to turn it off.
Frequency Adder	-200	200		Offsets the cutoff frequencies of the band-pass filters each update period. This causes them to migrate upward, for positive values, and downward, for negative values. Set this to 0 to turn it off.
Mult Bug	Off	On		Causes some mysterious and subtle behavior related to the Frequency Multiplier parameter and drawing new filter frequencies.
Change Bug	Off	On		Causes some yet more mysterious and subtle behavior. If you find out what it does, please send email to ear@sirius.com.
Randomize All Period	5	60000	ms	Sets the length of time between randomization of all filter frequencies (if the Randomize All Switch is enabled).
Randomize All Switch	0	1		Enables randomization of all filter frequencies, at a rate determined by the Randomize All Period parameter.
Randomize Cell Period	5	600000	ms	Sets the length of time between randomization of single cells (if the Randomize Cell Switch is enabled).
Randomize Cell Switch	0	1		Enables randomization of single cells, at a rate determined by the Randomize Cell Period parameter.
Cell Update Period	5	60000	ms	Sets the length of time between iterations of the cellular automata algorithm (if Cell Update is enabled).
Cell Update Switch	0	1		Enables the cellular automata algorithm, which will adjust the frequencies of the band-pass filters at a rate determined by the Cell Update Period parameter.
Base Frequency	0	200		Sets the center frequency for the Randomize All function.
Slew Rate	0	500		Sets the rate at which internal frequencies can change.
Diffusion Rate	0	5		Adjusts the diffusion of output volumes in the cell network.
Damp Rate	0	5		Adjusts the decay rate of output volumes in the cell network.
Change Rate	1	74		Sets the speed with which cell frequencies will move to new values.

Harmonic Filter

category: filter

audio input: mono | *cpu: medium - heavy (adjustable)* | *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Harmonic Limit	1	23		Sets the number of harmonics for the cellular automata to use. For example, if this is set to 5 (a good choice), each filter frequency will move to the nearest of the first five harmonics of its neighbor. If a cell's neighbor is at 440 Hz, this means the cell will move toward 440 Hz, 880 Hz, 1760 Hz, 3520 Hz, or 7040 Hz, whichever is closest.

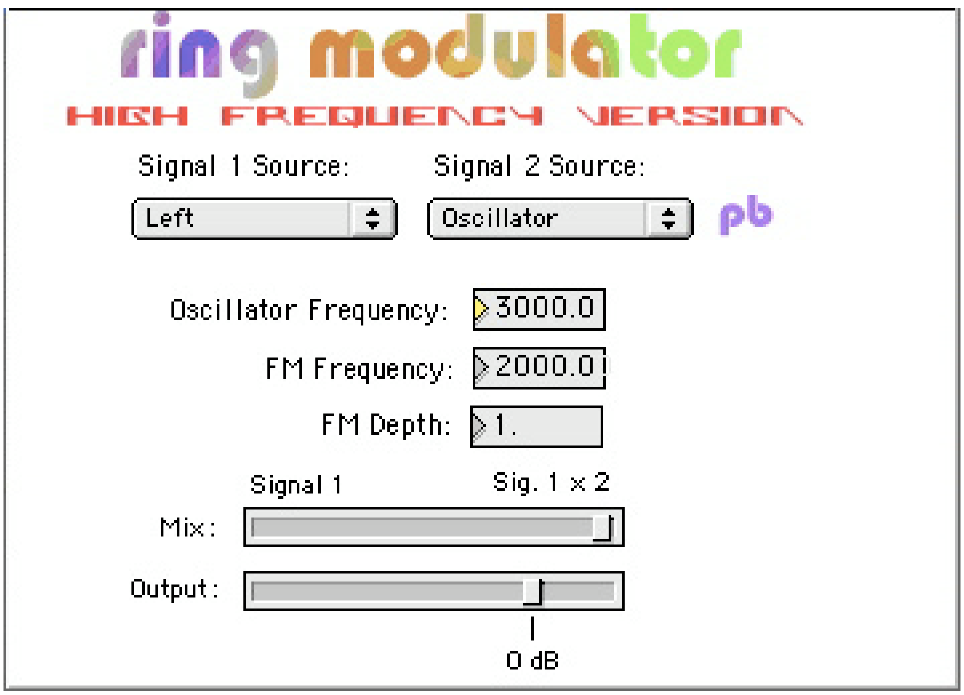
Insights

- If you are trying to figure out *Harmonic Filter*, start playing with the most important parameters: Cell Update Period, Slew Rate, Change Rate, and Harmonic Limit. The Frequency Adder and Multiplier parameters are also pretty critical. Sometimes, depending on the settings, it might seem that certain parameters have little or no effect.
- Set the frequency of a few of the filters to match the pitch of the input material, then anchor them into place and let the other filters move around. The anchors should keep the system in key, and the moving filters will reinforce different harmonics and resonances.

What It Does

HF Ring Mod bears a striking resemblance to its more staid and proper cousin *Ring Modulator*. It shares a similar lineage, too – it multiplies two signals together to produce a resulting signal that contains only frequency components equal to the sum(s) and difference(s) of the frequencies present in the input signals.

HF Ring Mod differs from *Ring Modulator* in that its internal oscillators have a wider range of frequencies. The sum frequencies created by a high modulation frequency can easily exceed the limits of hearing. The difference frequencies can fall below zero, where they are reflected back into the positive range (e.g., a difference frequency of –1974Hz becomes an audible signal of 1974Hz). The internal oscillator's modulation oscillator also has a higher range than *Ring Modulator*, so the modulating signal can have a wider range of frequency components. Typical hardware ring modulators provide an internal oscillator as signal, and hence have only one input. Like *Ring Modulator*, the *HF Ring Mod* object's signals may be any of the following: either input channel for this plug-in, an internal sine-wave oscillator, or any of the PluggoBus signals.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Signal 1 Source	Off	Pluggo Bus 4R		Selects an input signal. The signals designated "left" and "right" are the inputs to this plug-in. They will be the same if the plug-in is used in a mono context.
Signal 2 Source	Off	Pluggo Bus 4R		Selects an input signal. The signals designated "left" and "right" are the inputs to this plug-in. They will be the same if the plug-in is used in a mono context.
Oscillator Frequency	0	20000	Hz	Sets the frequency of the internal sine-wave oscillator.

HF Ring Mod

category: distortion

audio input: mono, pluggoBus | *cpu: light* | *audio output: mono*

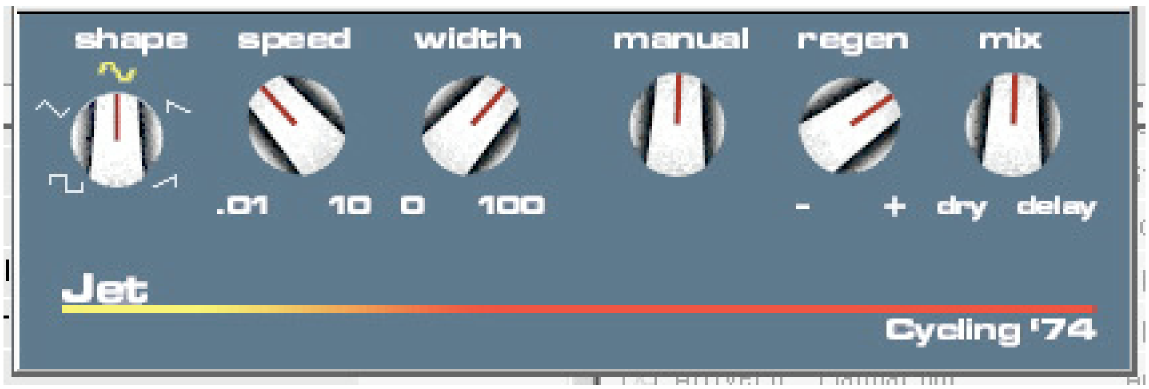
Name	*Min*	*Max*	*Units*	*Description*
FM Frequency	0	10000	Hz	Sets the frequency of a second oscillator which modulates the frequency of the first oscillator.
FM Depth	0	10000	Hz	Sets the amount that the second oscillator changes the frequency of the first oscillator.
Mix	0	100	%	Blends the output of the ring modulator with the Signal 1 input. Setting this slider to the far left sends only Signal 1 to the output. Setting it to the far right sends only the output of the ring modulator.
Output	-70	18	dB	Sets the overall output level of the effect.

Insights

- *HF Ring Mod* was created at the request of one of our loyal customers, who liked the original *Ring Modulator* but desired a wider range of modulation frequencies. To satisfy his request, and to avoid disturbing those customers who were already using the *Ring Modulator* plug-in in their projects, we created this high-frequency version..

What It Does

Jet is a fond homage to the stomp-box and rackmount flangers of yesteryear. It includes a faithful recreation of the frequency response of the analog delay chip found in one famous flanger, and a bi-directional knob for adding both positive and negative regeneration (or feedback) to intensify the flange effect. While most hardware flangers use a triangular or sinusoidal modulation oscillator to produce the familiar up-and-down flanging effect, *Jet* offers five different modulation wave shapes.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Shape				Sets the waveform of the low-frequency oscillator that varies the dalay time. You can select square, triangle, sine, or ramp waveforms.
Speed	.01	10	Hz.	Sets the speed of the low-frequency oscillator that varies the delay time.
Width	0	100	percent	Sets the amount the delay time is modulated by the low-frequency oscillator.
Manual	0.5	5	msec.	Sets the initial duration of the delay line.
Regen	-99	99	percent	Sets the amount of feedback for the delay line.
Mix	Dry	Delay		Sets the mix of the unprocessed to the processed signal output.

Insights

- *Jet* performs particularly well when its inputs are modulated by host-sync-based modulators like LFO.
- If the Width parameter is greater than zero, the delay time will be varied above and below the value set with the Manual knob, producing the "swooshing" flanger effect. If the Width parameter is zero, *Jet* acts like a comb filter; rotating the Manual knob will change the filtering effect.

- To create classic flanging effects, set the Mix knob to its center position (12:00 o' clock) to mix the raw and delayed signals in equal amounts

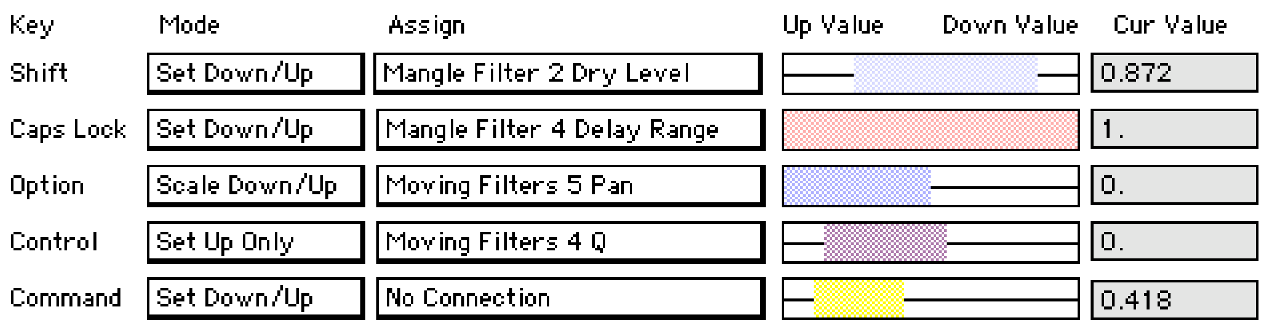
category: modulator

Key Triggers

audio input: none *cpu: light* *audio output: none*

What It Does

Key Triggers turns the state of the shift, caps lock, option, control, and command keys on your computer's keyboard into control signals you can use to modulate parameters of other plug-ins. Since pressing and releasing these keys by themselves is not detected as user input to the host application, they're safe to use for this purpose. You can apply the computer keyboard to triggering events (for example, with the *Breakpoints* plug-in) or for toggling between two values of an effect parameter.



Parameters for Each Key and Modulation Destination

Name	*Min*	*Max*	*Units*	*Description*
Mode	Off	Set Up Only		When the mode is Set Down/Up, the modulator's value directly sets the assigned parameter, with the Down value corresponding to the Range High Value and the Up value corresponding to the Range Low Value. When the mode is Offset Down/Up, the modulator's value is added to the current value of the parameter (before modulation). When the mode is Scale Down/Up, the modulator's value multiplies the current value of the parameter. When the mode is Set Down Only, only pressing the key down will trigger a change in the parameter value. When the mode is Set Up Only, only releasing a key will trigger a change in the parameter value. When the mode is Off, no changes to the parameter occur.
Assign				Choose a plug-in and a parameter to modulate from this menu. The parameters listed will depend on the plug-ins that are currently inserted.
Range Low Value	0	1		Use the range bar to scale the modulator's output data. The low value in the range is the minimum output value sent to the assigned parameter. You can verify the low value by releasing the corresponding key and watching the Cur Value display immediately to the left of the range bar.

Key Triggers

category: modulator

audio input: none | *cpu: light* | *audio output: none*

Name	*Min*	*Max*	*Units*	*Description*
Range High Value	0	1		Use the range bar to scale the modulator's output data. The high value in the range is the maximum output value sent to the assigned parameter. You can verify the high value by pressing the corresponding key and watching the Cur Value display immediately to the left of the range bar.

Insights

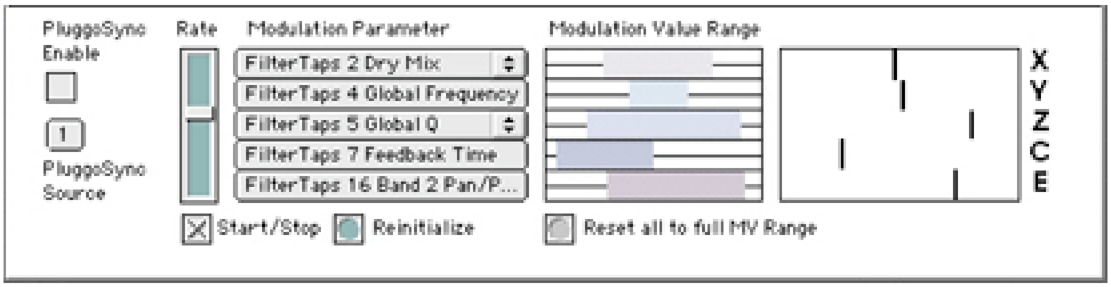
- To use with the *Breakpoints* plug-in, insert *Breakpoints* in the desired location in your mixer, then choose Breakpoints 8 Trigger from the *Key Triggers* Assign menu for the key you want to use as the trigger key. Then choose Set Down/Up from the *Key Triggers* Mode menu. With the *Breakpoints* edit window open, press the appropriate key and you should see the trigger button "fire" and the slider beneath the picture of the envelope begin to move to the right.

category: modulator **KnaveStories**

audio input: none *cpu: light* *audio output: none*

What It Does

KnaveStories uses the Navier-Stokes equation to generate modulation sources that are seemingly random, but also have some sense of correlation or coupling.



Parameters for Each Key and Modulation Destination

Name	*Min*	*Max*	*Units*	*Description*
PluggoSync Enable	Off	On		Enables the use of the *PluggoSync* plug-in to control the rate of parameter production if selected.
PluggoSync Source	1	4		Selects one of the four *PluggoSync* output sources
Modulation Parameter				Choose a plug-in and a parameter to modulate from this menu. The parameters listed will depend on the plug-ins that are currently inserted.
Range Low Value	0	1		Use the range bar to scale the modulator's output data. The low value in the range is the minimum output value sent to the assigned parameter.
Range High Value	0	1		Use the range bar to scale the modulator's output data. The high value in the range is the maximum output value sent to the assigned parameter.
Start/Stop	Off	On		Turns the equation generator on and off.
Reinitialize	0	1		Clicking on the Reinitialize button resets the Navier-Stokes equation to its initial settings.
Reset all to full MV Range				Clicking on the reset switch resets the low and high values for the Modulation Value ranges to their full range.

Insights

- The Navier-Stokes equation is calculated using one of Richard Dudas' lovely Chaos objects for Max/MSP. See ftp://ftp.ircam.fr/pub/forumnet/max/FAT/chance/ChaosCollectionFAT.sea.bin for a free downloadable version of these objects.
- If you have ever had the urge to use Max/MSP to create your own plug-ins, you might find Gregory Taylor's personal testament **Plug-In Confidential: How I Built My First Plug-in** to be amusing light reading. The article, along with a copy of the Max/MSP patch used to create this plug-in, can be downloaded from http://www.cycling74.com/download/KnaveStoriesKit.sit

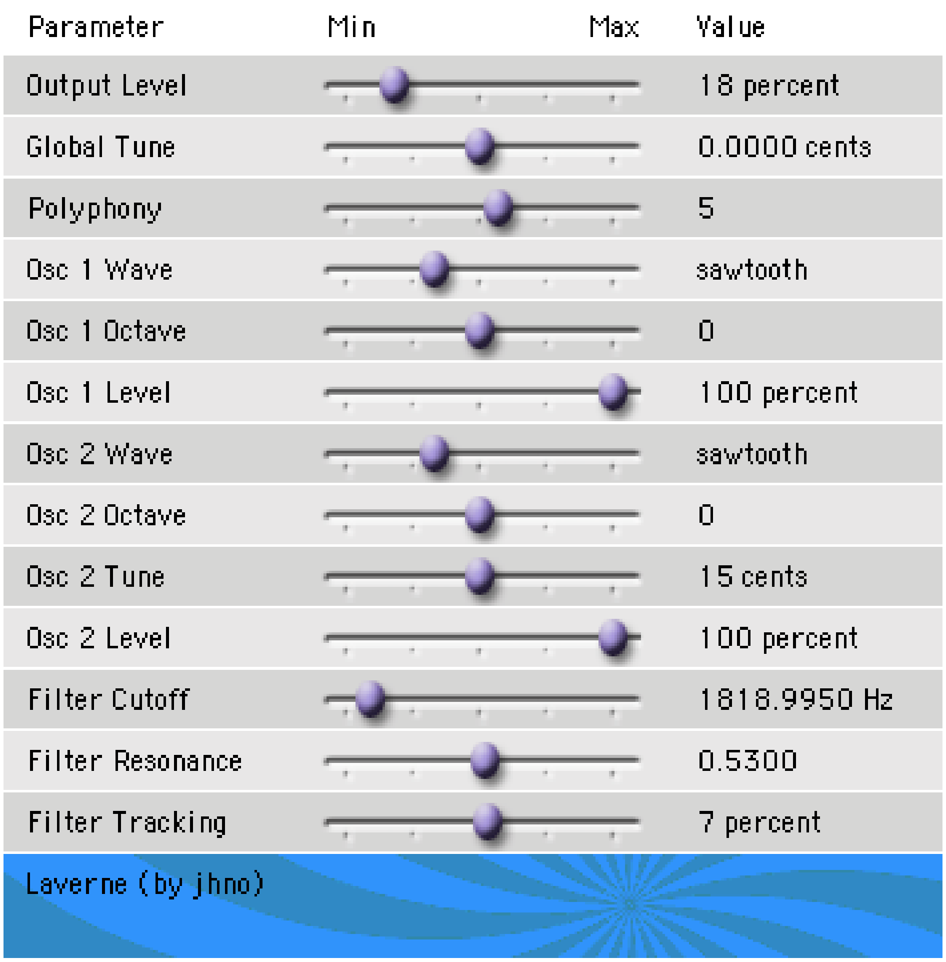
Laverne

category: synthesis

input: MIDI | *cpu: light* | *audio output: mono*

What It Does

Laverne is a simple, working-class subtractive synthesizer that works with VST 2.0 and MAS. It is intended as a demonstration of the MIDI capabilities of the plug-in environment, and an example for Max/MSP users who want to make their own synth plug-ins. Plus, it can make a few dope sounds. *Laverne* will be grayed out in the VST plug-in effects menu. You must use it as a VST Instrument in Cubase or on an Audio Instrument channel in Logic. In Digital Performer and Pro Tools, you can use it as a regular insert effect.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Output Gain	0	100	percent	Adjusts the global level of the output signal.
Global Tune	-100	100	Cents	Adjusts the overall pitch of the synthesizer in cents (a cent is 1/100th of a half step).
Polyphony	1	8	Voices	Sets the number of voices that can be played simultaneously by the synthesizer. Lower Polyphony settings use less CPU power.

Laverne

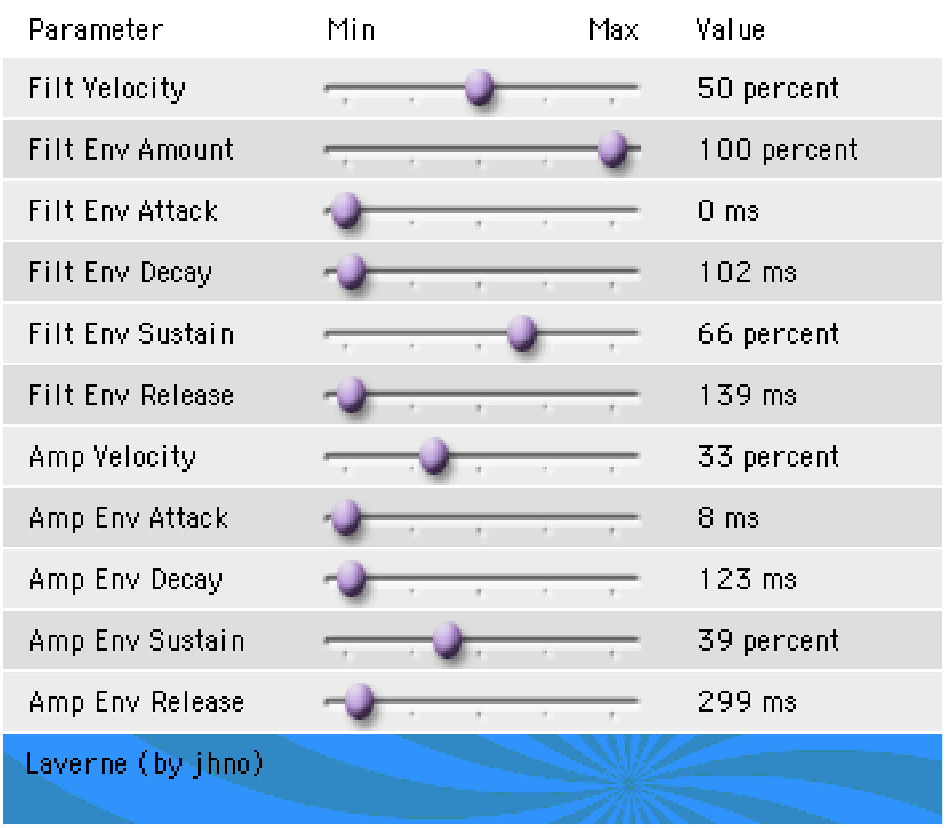
input: MIDI *cpu: light* *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Osc 1 Wave	Sine	Square		Selects a sine, sawtooth, triangle, or square waveform for Oscillator 1.
Osc 1 Octave	-5	5		Offsets the pitch of Oscillator 1 in octaves.
Osc 1 Level	0	100	percent	Sets the output level for Oscillator 1.
Osc 2 Wave	Sine	Square		Selects a sine, sawtooth, triangle, or square waveform for Oscillator 2.
Osc 2 Octave	-5	5		Offsets the pitch of Oscillator 2 in octaves.
Osc 2 Tune	-1200	1200	Cents	Sets the relative tuning in cents for Oscillator 2 in relation to Oscillator 1.
Osc 2 Level	0	100	percent	Sets the output level for Oscillator 2.
Filter Cutoff	10.0	10000.	Hz	Sets the base cutoff frequency for a 12dB/octave lowpass filter.
Filter Resonance	0	1		Sets the resonance or Q factor of the lowpass filter. Q is defined as the center frequency divided by the bandwidth.
Filter Tracking	-100	100		Determines how the filter cutoff frequency changes according to the MIDI note being played. Positive Filter Tracking values will cause higher notes to increase the filter's cutoff frequency; negative values will cause the cutoff frequency to decrease with higher notes. A Filter Tracking of 0 means that the MIDI note number will have no effect on the filter cutoff frequency.

Laverne

category: synthesis

input: MIDI | *cpu: light* | *audio output: mono*



Filter and Amplitude Envelope Parameters

Name	*Min*	*Max*	*Units*	*Description*
Filt Velocity	0	100	Percent	Determines the amount of MIDI note velocity applied to the filter cutoff frequency. At 0 percent, velocity will have no effect on the filter.
Filt Env Amount	0	100	Percent	Determines the amount of that the filter envelope affects filter cutoff frequency. At 0 percent, the filter envelope will have no effect on the cutoff frequency.
Filt Env Attack	0	100	Percent	Sets the attack time for the filter envelope, in milliseconds. When a note is played on the synthesizer, the filter envelope is triggered, and the attack time determines how long it takes the envelope to move from zero to its maximum effect, as determined by the Filt Env Amount parameter.
Filt Env Decay	0	100	Percent	Sets the decay time for the filter envelope, in milliseconds. After the envelope has reached its maximum effect (according to the Attack parameter), the decay time determines how long it takes for the envelope to move to its sustain level.

Name	Min	Max	Units	Description
Filt Env Sustain	0	100	Percent	Sets the sustain level for the filter envelope, specified as a percentage of its maximum effect. The envelope will remain at this level until the note is released.
Filt Env Release	0	100	Percent	Sets the release time for the filter envelope, in milliseconds. This is the amount of time it takes for the envelope to move back to zero after the note has been released.
Amp Velocity	0	100	Percent	Determines the amount of MIDI note velocity applied to the synthesizer's output amplitude. At 0 percent, velocity will have no effect on amplitude.
Amp Env Attack	0	100	Percent	Sets the attack time for the amplitude envelope, in milliseconds. When a note is played on the synthesizer, the amplitude envelope is triggered, and the attack time determines how long it takes the amplitude to move from zero to its maximum value.
Amp Env Decay	0	100	Percent	Sets the decay time for the amplitude envelope, in milliseconds. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the amplitude to move to its sustain level.
Amp Env Sustain	0	100	Percent	Sets the sustain level for the amplitude envelope, specified as a percentage of its maximum effect. The amplitude will remain at this level until the note is released.
Amp Env Release	0	100	Percent	Sets the release time for the amplitude envelope, in milliseconds. This is the amount of time it takes for the amplitude to move back to zero after the note has been released.

Insights

- Like its effects processing brethren, *Laverne's* parameters are available for manipulation using a Modulator plug-in, greatly expanding the possibilities for modulation in this synthesizer.

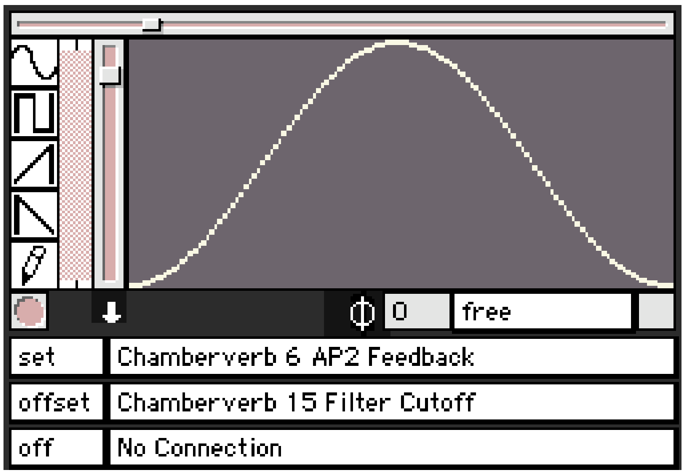
LFO

accepts sync | category: modulator

audio input: none | cpu: medium | audio output: none

What It Does

LFO (which stands for low-frequency oscillator) sends a repeating waveform of control information to modulate parameters of another plug-in.



LFO Module Parameters

Name	*Min*	*Max*	*Units*	*Description*
LFO Waveform	Sine	Draw		Sets the current shape of the LFO waveform. By clicking on the buttons to on the left side of the LFO module, you can select a Sine, Square, Saw Up, Saw Down, or User wave shape. When the User wave shape is selected, you can draw in the LFO Waveform display to set the LFO shape.
LFO Min	0	127		Sets the minimum value that the LFO will produce (when the LFO Waveform display shows its lowest value).
LFO Max	0	127		Sets the maximum value that the LFO will produce (when the LFO Waveform display shows its highest value).
LFO Phase	0	360	degrees	Offsets the start position of the LFO, in other words, where it starts in its waveform when it is retriggered. A value of 0 means the LFO starts at the beginning, and a value of 360 means it starts at the end. This parameter can be useful to employ when LFO modules are synchronized

Name	*Min*	*Max*	*Units*	*Description*
LFO Sync Source	UDT	Plug		Sets the tempo and synchronization mode for LFO. The pop-up menu lets you choose from three sync modes: • UDT (User-Defined Tempo) mode lets you select the LFO tempo with the Tempo slider. • Host mode synchronizes the LFO Tempo with the host tempo. • Plug mode sets the LFO Tempo from the PluggoSync plug-in if it is loaded. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
LFO Tempo	1	300	BPM	Sets the LFO tempo in beats per minute. This parameter, along with the Note and Multiplier parameters below, determines the rate of the LFO cycle.
LFO Sync Multiplier	0.01	75		Sets the multiplication factor that determines the LFO speed, as described below.
LFO Sync Note Value	1	1/64t		Sets the base note duration for a single LFO cycle. This note duration is multiplied by the LFO Sync Multiplier to set the LFO speed, according to the current LFO Tempo. For example, if you select a Sync Unit of 1/4 (a quarter note), and a Sync Multiplier of 1, LFO will cycle once every quarter note. With an LFO Tempo of 120 BPM, this would be one cycle every 500 ms, or 2 Hz. If you change the Sync Multiplier to 2, LFO will cycle once every two quarter notes: twice as slow. On the other hand, a Multiplier of 0.25 will make it four times faster - one cycle every sixteenth note.

Parameters for Each Modulation Destination

Name	*Min*	*Max*	*Units*	*Description*
Mode	Off	Scale		When the mode is Set (the most common), the modulator's value directly sets the assigned parameter. When the mode is Offset, the modulator's value is added to the current value of the parameter (before modulation). When the mode is Scale, the modulator's value multiplies the current value of the parameter. When the mode is Off, no changes to the parameter occur.
Assign				Choose a plug-in and a parameter to modulate from this menu. The parameters listed will depend on the plug-ins that are currently inserted.

Other Parameters

The individual points that make up the waveform used in Draw mode are saved with each effect program.

Note: a parameter labeled "147. LFO Sync" will appear in the Assign menu of a Modulator plug-in when the *LFO* plug-in is loaded. Modulating this parameter will have no effect.

LFO

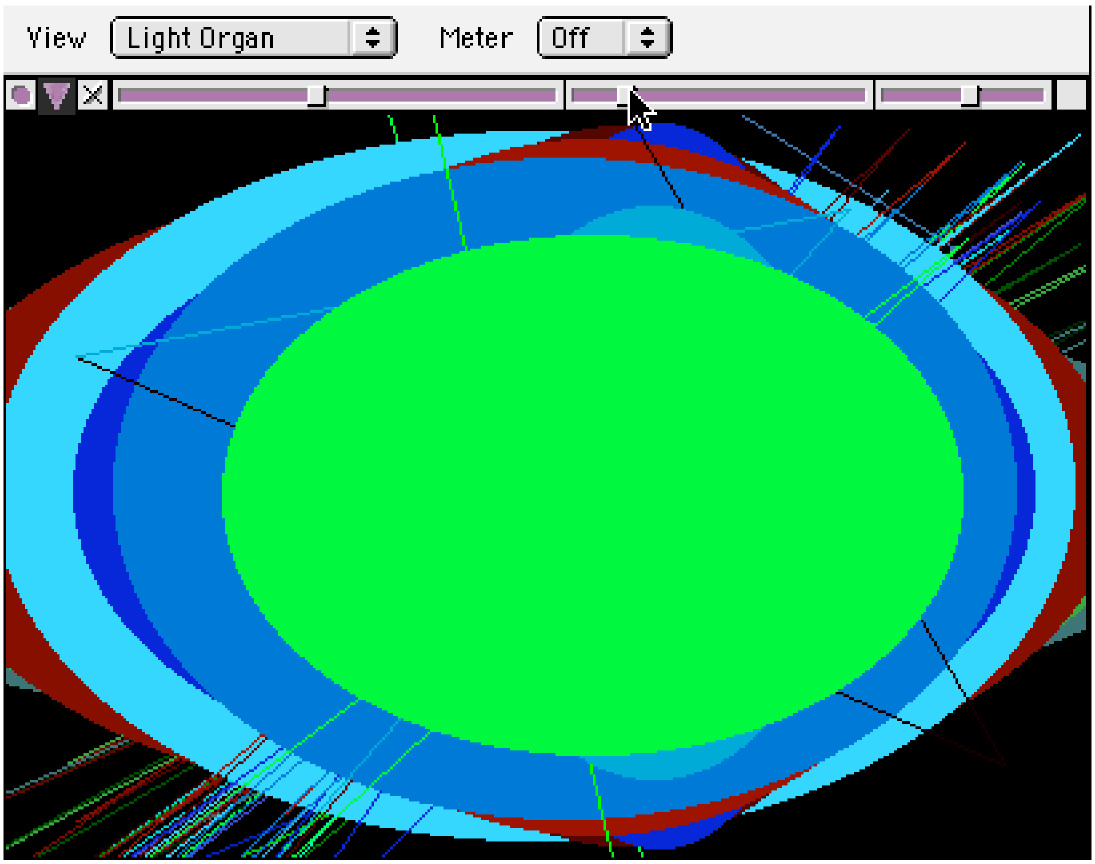
	accepts sync	*category: modulator*
audio input: none	*cpu: medium*	*audio output: none*

Insights

- Just as in a traditional synthesizer, you can't listen to the *LFO* directly. Indeed, this one doesn't even output an audio signal.
- You'll find LFO modules contained within the *Swish* and *Warble* effects.
- You can use *LFO* as an envelope by retriggering it (either manually or with another Modulator plug-in).
- The LFO Info view depicts an inchworm on a piece of broccoli.

What It Does

Light Organ is a visual toy that paints shapes and colors according to the sound you put into it. It does not process audio—the input signal is passed straight through to the output. To generate its display, *Light Organ* analyzes its audio input with three band-pass filters. Various aspects of the drawing algorithm are influenced by the signal levels present in each band.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Sensitivity	0	199		Adjusts the input signal volume for Light Organ to analyze.
Spread	0	100		Adjusts the bandwidth of the filters used for signal analysis. Higher values create narrower response, for which you might want to compensate by increasing the Sensitivity parameter.
Variation	0	1		Selects one of two different positions for the drawing to be centered.
Mode	0	2		Selects ovals, lines, or both ovals and lines.

Light Organ

category: visual display

audio input: stereo | *cpu: light* | *audio output: none*

Name	*Min*	*Max*	*Units*	*Description*
Fade	0	99		Sets how often the screen will redraw its background color - which determines how long the shapes will persist before they are erased.
Flashes	0	1		Enables occasional flashes of color in the screen background.

Insights

- *Light Organ* is very sensitive to the level of its input—so be sure to adjust the Sensitivity parameter. The Spread parameter also affects the analysis signal levels. For example, increasing the Spread parameter makes the bandpass filters narrower, and you might need to increase Sensitivity to compensate.
- The Fade and Flashes parameters can be unreliable. This is partially because the redraw rate varies with the CPU load. Best not to worry about this. Just enjoy the pretty colors.

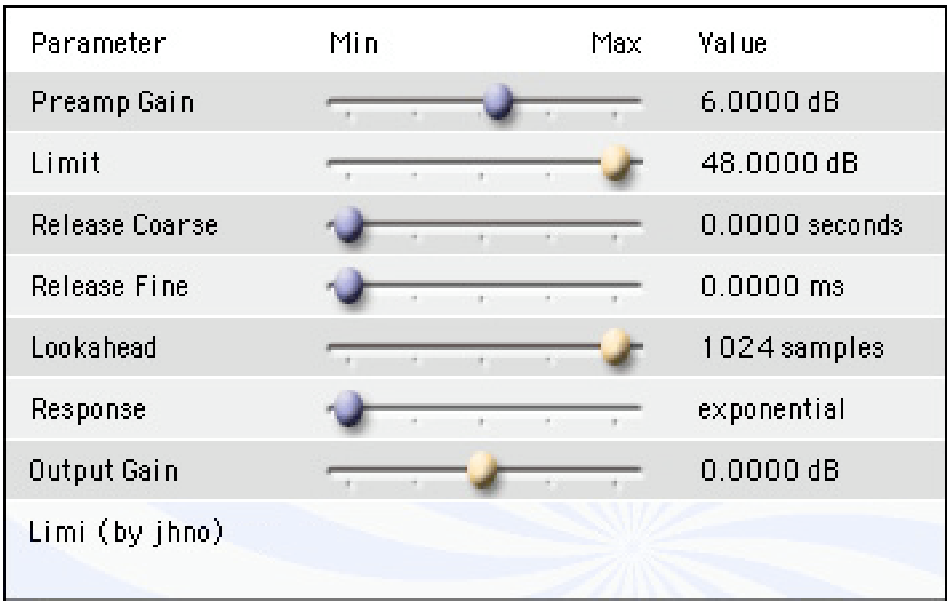
category: reverb/dynamics

Limi

audio input: stereo | *cpu: light* | *audio output: stereo*

What It Does

Limi is a general-purposes limiter plug-in that allows you be sure that your audio output will never exceed a maximum value you specify. You can use it in many ways; you can used it for clipping protection while you're recording, clipping and headroom control in live situations, as a compression/distortion effect, or for transparent level optimization during mastering. While *Limi* controls are simple, they operate over an extreme range, and *Limi* uses double-precision floating point math under the hood with bit-identical pass-through of latent stages for maximum fidelity.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Preamp Gain	-76	+18	dB	Sets the level of the unprocessed input signal.
Limit	-48	48	dB	Sets the maximum output amplitude of the input signal
Release Coarse	1	120	Seconds	Sets the coarse recovery time of the limiter in milliseconds
Release Fine	1	1000	msec	Used in conjunction with the Coarse Recovery Time parameter to fine-tune the recovery time of the limiter in milliseconds
Lookahead	1	1024	samples	Sets the number of samples to analyze before output. The output is delayed by the number of samples specified.
Response	Exponential	Linear		Sets the desired type of amplitude gain
Output Gain	-76	+18	dB	Sets the level of the limited output signal.

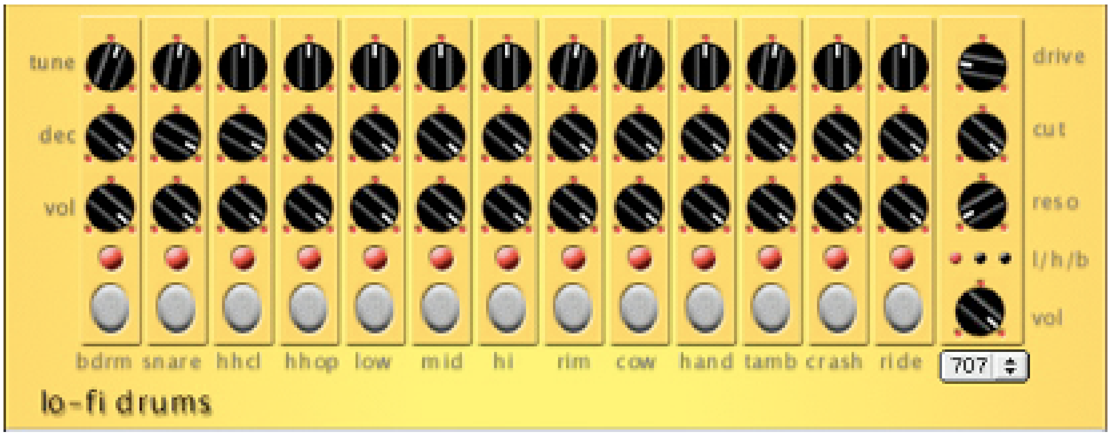
lo-fi drums

category: synthesis

input: MIDI | *cpu: light* | *audio output: mono/stereo*

What It Does

lo-fi drums is an easy-to-use drum module featuring gritty 8-bit samples. The samples are taken from some classic drum machines, but the recordings were purposely sample rate and bit depth reduced for some unique sounds.



Visible Channel Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Tuning	0 (.07x)	127 (1.9x)	40-52	Sets the tuning for a drum channel's sample playback.
Decay Rate	0 (5 ms)	127 (1505 ms)	27-39	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes the amplitude to return to zero.
Volume	0 (0%)	127 (100%)	14-26	Adjusts the individual channel levels output.
Load Button			N/A	Press the Load Button to choose a new sample for a given drum channel. An Open File dialog will be displayed, allowing a new sample to be selected.
Channel Trigger			N/A	Click on the Trigger Button for a drum sample to audition that channel's settings.

Visible Global Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Drive	0 (0%)	127 (100%)	53	Determines the amount of overdrive (soft distortion) applied to signals routed to through filter subsystem.

Name	*Min*	*Max*	*CC#*	*Description*
Filter Cutoff	0 (0 Hz)	127 (10 kHz)	54	Sets the cutoff frequency of the filter.
Filter Resonance	0 (0.0)	127 (1.0)	55	Sets the resonance value (sometimes called Q or pre-emphasis) of the filter. Resonance values range from 0.0 (no resonance) to 1.0 (self-oscillation).
Filter Type	0 (low-pass)	2 (band-pass)	56	Determines the filter type used by the filter subsystem. This setting determines the function of the filter cutoff setting. Values are low-pass, hi-pass and band-pass.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

Insights

- Every instrument is routed through the filter and overdrive systems. If you want to have only a few modified drum sounds, it's best to use two instances of lo-fi drums, one with overdrive/filter effects, and one without.
- The samples used by lo-fi drums were recorded at 22,050 kHz, at 8-bit depth. The aliasing that you may hear isn't a bug, it's a feature.
- The MIDI notes used to fire the drums sounds are:

  Channel 1: 36 - kick
  Channel 2: 38 - snare
  Channel 3: 42 - closed hat
  Channel 4: 46 – open hat
  Channel 5: 41 - low tom
  Channel 6: 43 – mid tom
  Channel 7: 45 – hi tom
  Channel 8: 37 – rim shot
  Channel 9: 56 - cow bell
  Channel 10: 39 – hand clap
  Channel 11: 54 – tambourine/percussion
  Channel 12: 49 – crash cymbal
  Channel 13: 51 – ride cymbal

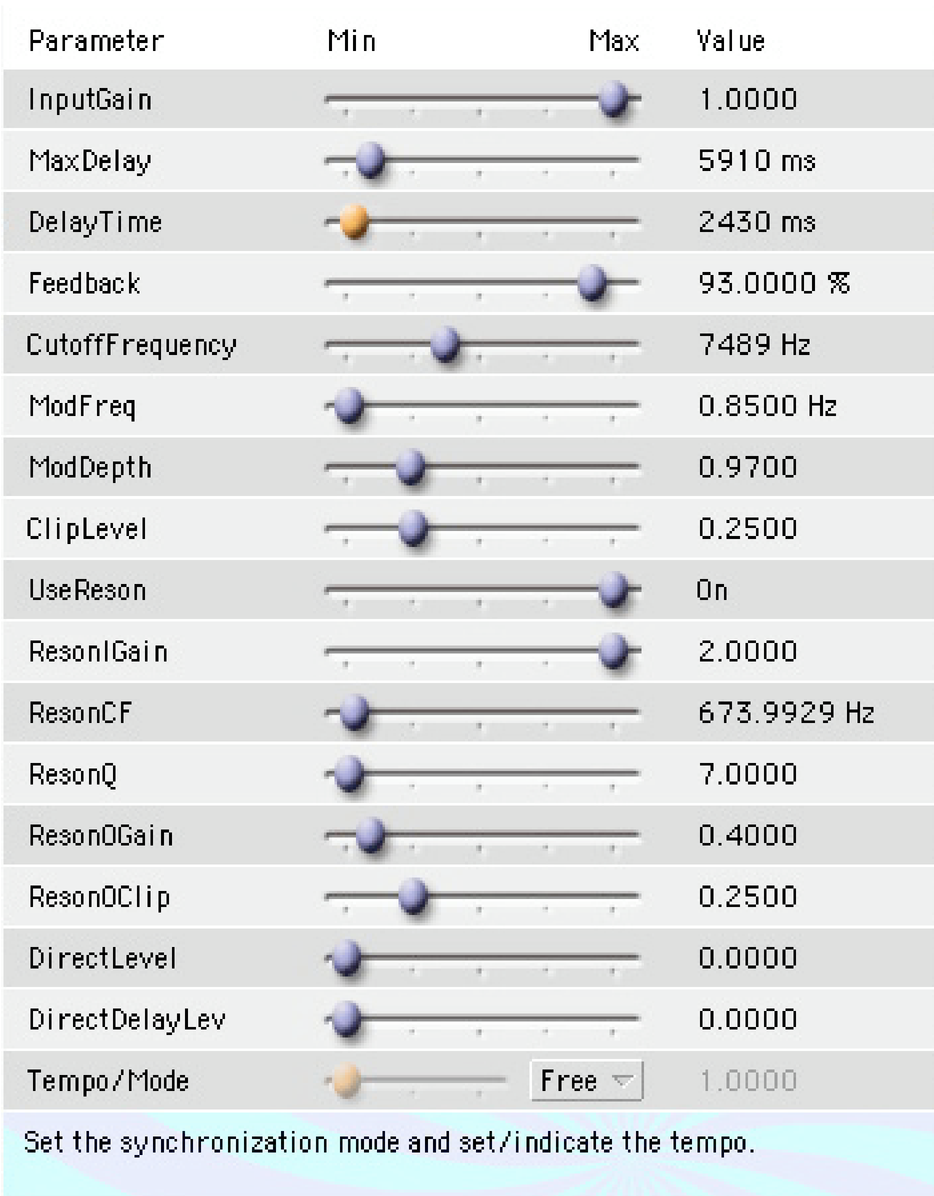
Long Stereo Delay

accepts sync | *category: delay*

audio input: stereo | *cpu: medium* | *audio output: stereo*

What It Does

Long Stereo Delay is very similar to the *Very Long Delay* plug-in. It has exactly the same parameters as *Very Long Delay* and includes the same vibrato, low-pass filter, and a resonant bandpass filter you can patch into the delay line, but uses *two* independent delay lines; when you change a parameter, both delay lines are affected. In addition, the maximum delay time is 60 seconds.).



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Input Gain	0	1		Sets the gain on the input to the delay.
MaxDelay	0	60000	ms	Sets the total available memory for the delay line. After changing this parameter, there is a short pause in the output as the newly allocated delay line is filled. Note that all the presets in *Long Stereo Delay* use a 3000 ms buffer, so there is not a pause when switching among them. If the delay line you attempt to allocate is too large for available memory, the effect will attempt to reallocate the delay line that existed before. In any event, you will not see an error message.
DelayTime	0	60000	ms	Sets the delay time. Note that the slider covers the full possible range, but only a subset that range will actually make sense if the MaxDelay parameter is set to a value less than the maximum possible delay time of 30 seconds. The delay time is set in milliseconds when the Sync Mode (shown in the Tempo/Sync parameter) is set to Free. In Host, Plug, and UDT modes, the delay time parameter is specified using note units and a multiplier.
Feedback	0	100	percent	Sets the percentage of the delayed signal that is fed back into the delay line.
CutoffFrequency	10	20000	Hz	Sets the cutoff frequency of a lowpass filter within the delay line's feedback loop.
ModFreq	0	50	Hz	Sets the frequency of delay time modulation.
ModDepth	0	4		Sets the amount of delay time modulation.
ClipLevel	0	1		Sets the amplitude above which values in the feedback delay line are clipped. Values less than 1 may produce distortion.
UseReson	Off	On		Sets whether the delay line includes a resonant bandpass filter.
ResonIGain	0	2		Input gain before the resonant bandpass filter. If either this value or the ResonOGain parameter is zero, there will be no feedback signal in the delay line.
ResonCF	10	18000	Hz	Sets the center frequency of the resonant bandpass filter.
ResonQ	0	500		Sets the resonance ("Q") of the bandpass filter. Q is defined as bandwidth divided by center frequency.
ResonOGain	0	4		Sets the output gain on the resonant bandpass filter.
ResonOClip	0	1		Sets the amplitude above which the output of the resonant bandpass filter is clipped before being fed back to the input of the delay line. Values less than 1 may produce distortion.
Direct Level	0	1		Sets the gain of the undelayed input signal.

Long Stereo Delay

accepts sync *category: delay*

audio input: stereo *cpu: medium* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
DirectDelayLev	0	1		Sets the gain of the output directly from the delay line. This provides an unprocessed but delayed version of the original signal to be included in the plug-in's output.
Delay Time Mult	0	10		Sets a multiple of the Delay Time Units currently selected that determines the delay time at the current tempo. For example, if the value of this parameter were 2 and the units were set to 1/4 (quarter note), the resulting delay time would be a half-note at the current tempo. This parameter is disabled when the Sync Mode is set to Free.
Delay Time Units	1	64t.		Sets the base note duration value used in determining the delay time in relation to the current tempo. This value is multiplied by the Delay Time to obtain the total beat value used to calculate the delay time. This parameter is disabled when the Sync Mode is set to Free.
Tempo	1	300	BPM	Tempo is displayed as a slider in the Tempo/Sync parameter. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the delay time. When in Host and Plug modes, the tempo is an indicator that cannot be changed--it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range and Very Long Delay will calculate the delay time based on the values of the Coarse Pan Freq Mult and Coarse Pan Freq Units parameters. This parameter is disabled when the Sync mode is set to Free. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Sync Mode	Free	UDT		Sync Mode is a pop-up menu in the Tempo/Sync parameter. There are four modes of sync available: • Free mode lets you set the delay time independent of the host sequencer. • Host mode synchronizes the delay time to the host tempo. • Plug mode synchronizes the delay time to the beat output of PluggoSync • UDT (User-Defined Tempo) mode is equivalent to Free mode but allows you to set the delay time in terms of tempo and note unit values. Note: Host mode is only available in VST 2.0 and MAS applications that support it.

Insights

- Both inputs and outputs are stereo; this means if you use it as a mono effect, only the left channel will output.

- The bandpass filter mode is useful for creating background ambience, distortion, and feedback effects that are semi-related to the input material.

M2M

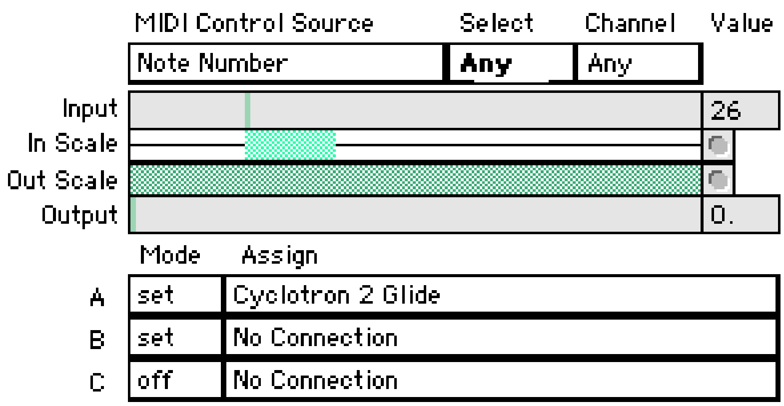
category: modulator

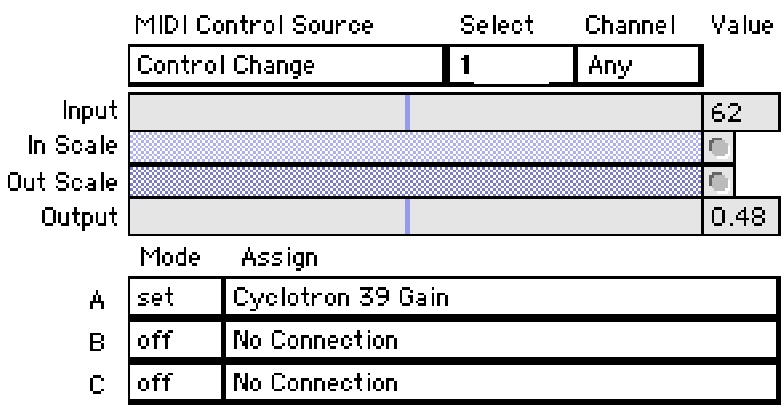
input: MIDI | *cpu: light* | *audio output: none*

What It Does

M2M stands for MIDI to Modulator. It converts incoming MIDI messages into modulation data to change the parameters of other plug-ins. Even though, like all Modulator plug-ins, it makes no sound, *M2M* is a VST instrument because in Cubase and Logic there is no way for an audio effect plug-in to receive MIDI. This means it will be grayed out in the VST plug-in effects menu. In Digital Performer and Pro Tools, *M2M* can be used as an insert effect on any audio channel.

M2M lets you use existing or special-purpose MIDI tracks as controllers for plug-ins. Just assign *M2M* as the output of your favorite MIDI track, then use the *M2M* edit window to choose another plug-in to modulate.





Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
MIDI Control Source	Note Number	Program change		Choose a type of MIDI message to use as a modulation source.
Select	Any	127		Filters incoming MIDI messages when the Control Source is set to Control Change, Velocity, or Poly Key Pressure. If the Control Source is set to Control Change, you can use Value Select to specify the controller number you want to use as a modulation source. All other controller numbers will be ignored. If you set Select to Any, all incoming control change messages will be used. In a similar manner, if the Control Source is set to Velocity or Poly Key Pressure, you can use Value Select to specify the MIDI note number that will cause modulation
Channel	Any	16		Filters incoming MIDI messages according to MIDI Channel. Choose the channel on which *M2M* will receive control information, or select Any to receive on all channels
Input	0	127		The top slider and number box reflect the value of the incoming MIDI message. You can also drag on the slider with the mouse to generate parameter modulation output for testing or amusement purposes.
In Scale	0	127		Sets the range of input values that represent the output range 0-1. Small input ranges allow you to magnify input data with a small amount of variance into a large output range. When the In Scale range bar is set to its full width, no scaling occurs.
Out Scale	0	1		The output values are constrained between the minimum and maximum values you set with the Out Scale range bar. When the Out Scale range bar is set to its full width, no modification occurs.
Output	0	1		Displays the value to be sent to the selected modulation destinations.

Parameters for Each Modulation Destination

Name	*Min*	*Max*	*Units*	*Description*
Mode	Off	Scale		When the mode is Set (the most common), the modulator's value directly sets the assigned parameter. When the mode is Offset, the modulator's value is added to the current value of the parameter (before modulation). When the mode is Scale, the modulator's value multiplies the current value of the parameter. When the mode is Off, no changes to the parameter occur.
Assign				Choose a plug-in and a parameter to modulate from this menu. The parameters listed will depend on the plug-ins that are currently inserted.

Interface Elements

- Clicking the small buttons to the right of the In Scale and Out Scale range bars will set them to their full range.
- You can preview the effect of incoming MIDI values by clicking and dragging on the top slider. The number box to the right of this slider displays the numeric "MIDI" value you're simulating, and scaled values are output to any assigned Modulation destinations.

Insights

- *M2M* can serve as an alternative to using plug-in automation generated by recording the movement of the plug-in's faders. You can experiment with different MIDI sources to determine what will produce the most interesting modulation functions.

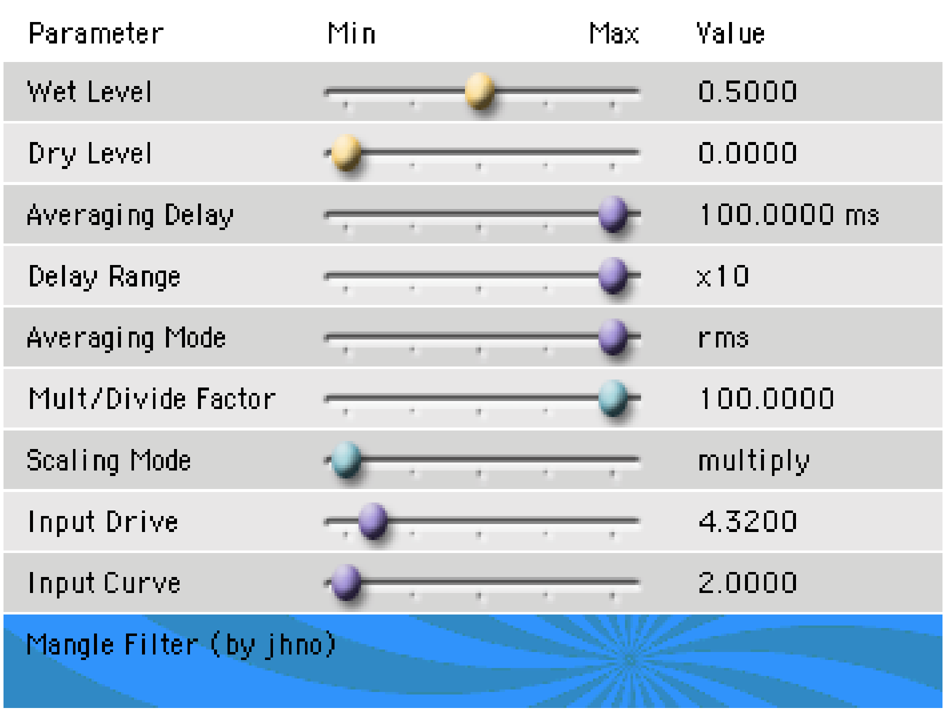
category: distortion **Mangle Filter**

audio input: mono *cpu: light* *audio output: mono*

Mangle Filter

What It Does

Mangle Filter affects the amplitude and delay time of its input using the average amplitude value of its input. In other words, as a signal gets louder the delay line it is running through gets longer. This can happen at a very fast rate, creating unusual and extreme filter and distortion sounds. *Mangle Filter* is similar to *Average Injector*, but contains arbitrary differences in its design that produce somewhat different results.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Reduction in the gain of the undelayed input signal. Negative values invert the input's phase.
Dry Level	0	1		The level of the delayed input signal. Negative values invert the delayed signal's phase.
Averaging Delay	0	100	ms	Sets the time interval over which the input signal is averaged. This value is multiplied by the Delay Range parameter to set the actual time over which the signal is averaged.
Delay Range	x1	x10		Multiplies the Averaging Delay by a scaling factor.
Averaging Mode	bipolar	rms		Sets the mode used to compute the average amplitude value of the signal. Note that setting the mode to rms consumes significantly more CPU power—and the difference between RMS and Linear averaging computation is usually subtle.

Mangle Filter

category: distortion

audio input: mono *cpu: light* *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Mult/Divide Factor	0	100		Sets a factor by which the average signal amplitude value is multiplied or divided (depending on the Scaling Mode).
Scaling Mode	mult	divide		Sets whether the average signal amplitude value is multiplied or divided before it modulates the delay time and amplitude of the input signal.
Input Drive	0.1	42		Gain on the input signal before it is averaged. However, this parameter does affect the gain on the signal that is delayed.
Input Curve	2	8		Applies an exponential function to the input signal, which changes the response of the averaging delay. The parameter sets the steepness of the exponential function.
Filter Cutoff	20	20000	Hz	Sets the cutoff frequency of a lowpass filter. Using the filter helps take some of the edge off of the mangled sound, if you start feeling agitated.

Insights

- *Mangle Filter* is extremely sensitive to input level. A setting that is having no effect on your sound might completely annihilate it if you turn the input up a little.
- *Mangle Filter* can work well with sounds that have well-defined peaks, such as drum tracks. If you tweak it correctly, it will pass most of the signal unchanged, and only the peaks will trigger mangling.

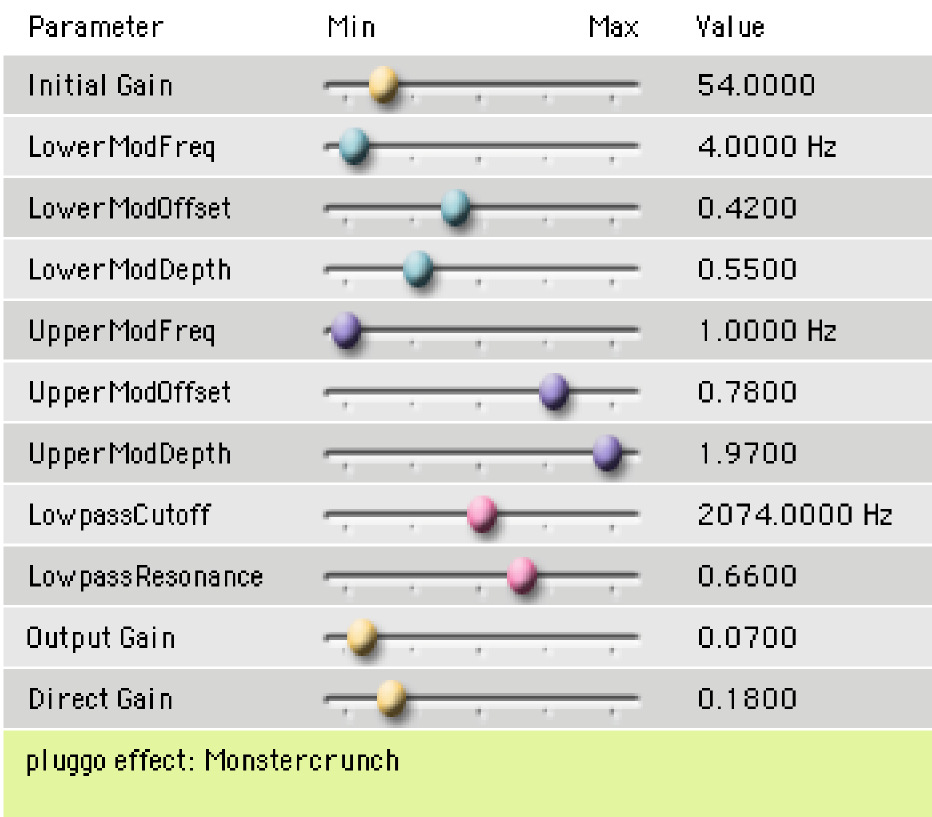
category: distortion

Monstercrunch

audio input: mono | *cpu: light* | *audio output: mono*

What It Does

Monstercrunch amplifies its input signal by a huge amount, clips it mercilessly, and then low-pass filters the daylights out of it. Isn't this sort of rough treatment the essence of distortion?



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Initial Gain	0	360		Sets the scaling factor by which the input is multiplied before it is clipped.
Lower Mod Freq	0	100	Hz	Sets the rate of modulation of the lower value of the signal clipper.
Lower Mod Depth	0	2		Sets the amount of modulation of the lower value of the signal clipper.
Upper Mod Freq	0	100	Hz	Sets the rate of modulation of the upper value of the signal clipper.
Upper Mod Depth	0	2		Sets the amount of modulation of the upper value of the signal clipper.
Lowpass Cutoff	0	4000	Hz	Sets the cutoff frequency of a filter that occurs after clipping.

Monstercrunch

category: distortion

audio input: mono *cpu: light* *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Lowpass Resonance	0	0.99		Sets the resonance of the lowpass filter that occurs after clipping.
Output Gain	0	1		Sets the gain on the output of the filter.
Direct Gain	0	1		Sets the gain on the uncrunched input signal.

Insights

- When creating an effect with *Monstercrunch*, you are mostly concerned with the oscillators that modulate the high and low signal clipping values. You can think of these modulators as pincers that are squeezing the signal; the effect of severe clipping is, of course, extreme distortion, but when both clipping values are being modulated independently, the level of distortion can change in complex (albeit repetitive) ways.
- The *nicely* preset demonstrates that when the signal is amplified by a large amount, the modulation of the clipping levels is effectively amplitude modulation. Move the Upper Mod Freq to 0 and you will hear more clearly what the modulation of the clipping levels was doing.

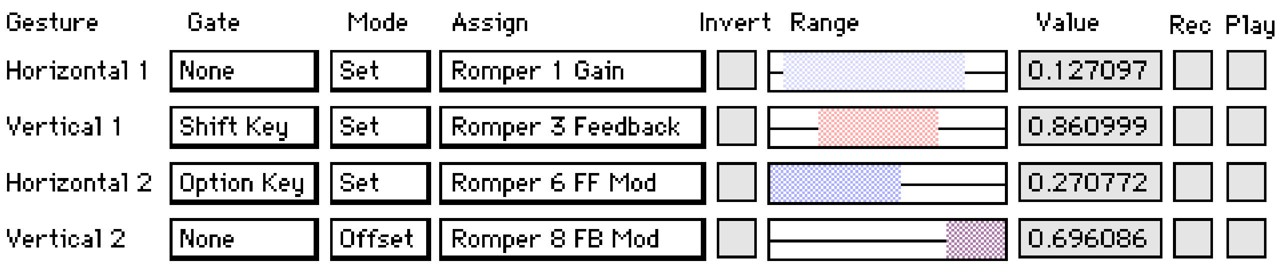
category: modulator

Mouse Mod

audio input: none *cpu: light* *audio output: none*

What It Does

Mouse Mod tracks the mouse cursor position on the screen and translates it into values you can use to modulate parameters of other plug-ins. Each modulation destination can be gated by a specific modifier key on the keyboard so that you can still operate your computer by moving the mouse and not make unwanted parameter value changes. By enabling the Record (Rec) control, you can use the host sequencer's automation facility to record your mouse gestures. Then, play the automation back into Mouse Mod, enabling the Play control, and the automation will be passed directly to the assigned modulation destination.



Parameters for Each Modulation Destination

Name	*Min*	*Max*	*Units*	*Description*
Value	0	1		Each destination has a parameter associated with it that can be used for recording and playback of a mouse gesture using the host sequencer's automation facility.
Gate	None	Command Key		If None is selected, changes in the mouse cursor position are always output to the modulation destination. Otherwise, only when the selected key (shift, caps lock, option, control or command) is held down will changes in the mouse cursor position cause an output of the modulator's value to the destination.
Mode	Off	Scale		When the mode is Set (the most common), the modulator's value directly sets the assigned parameter. When the mode is Offset, the modulator's value is added to the current value of the parameter (before modulation). When the mode is Scale, the modulator's value multiplies the current value of the parameter. When the mode is Off, no changes to the parameter occur.
Assign				Choose a plug-in and a parameter to modulate from this menu. The parameters listed will depend on the plug-ins that are currently inserted.
Invert	Off	On		When checked, the output value is inverted. This means, for example, that moving the cursor to the right edge of the screen produces a value at the low end of the range rather than the higher end of the range.
Range Low Value	0	1		Use the range bar to scale the modulator's output data. The low value in the range is the minimum output value sent to the assigned parameter.

Mouse Mod

category: modulator

audio input: none *cpu: light* *audio output: none*

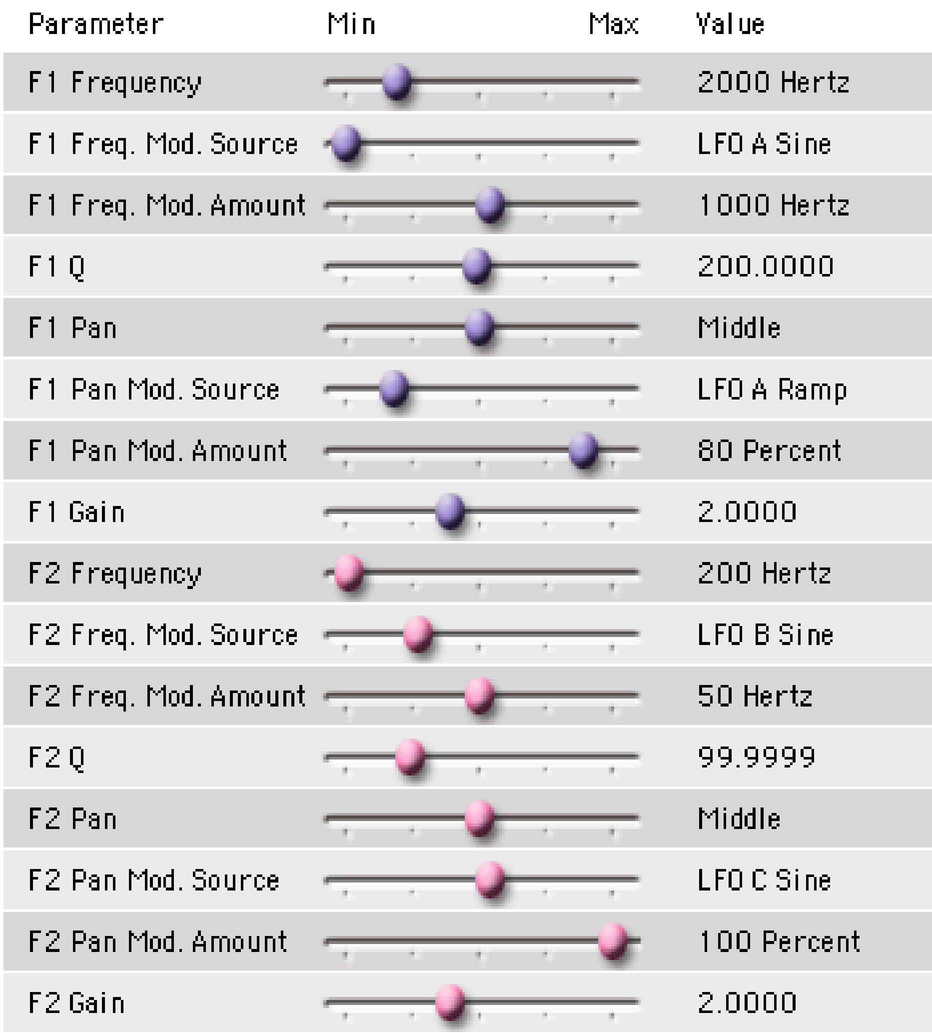
Name	*Min*	*Max*	*Units*	*Description*
Range High Value	0	1		Use the range bar to scale the modulator's output data. The high value in the range is the maximum output value sent to the assigned parameter.
Record	Off	On		When checked, the output value is written out to a parameter, allowing it to be recorded by the host's automation facility.
Play	Off	On		When checked, values for this destination come from changes to the corresponding parameter rather than from the mouse. This can be used to play back previously recorded automation data coming from a host. When Play is enabled, the mouse gesture and the associated controls (such as Invert and Range) are disabled for the specified modulation destination.

category: filter **Moving Filters**

audio input: mono/stereo *cpu: light* *audio output: stereo*

What It Does

Moving Filters passes its input signals through two parallel bandpass filter. The center frequency, bandwidth, and stereo position of each filter is independently adjustable. In addition, there are four low-frequency oscillators that can be used to modulate the frequency and stereo position of the filters. Each LFO provides three wave shapes, which are available simultaneously.



Parameter	Min	Max	Value
F1 Frequency			2000 Hertz
F1 Freq. Mod. Source			LFO A Sine
F1 Freq. Mod. Amount			1000 Hertz
F1 Q			200.0000
F1 Pan			Middle
F1 Pan Mod. Source			LFO A Ramp
F1 Pan Mod. Amount			80 Percent
F1 Gain			2.0000
F2 Frequency			200 Hertz
F2 Freq. Mod. Source			LFO B Sine
F2 Freq. Mod. Amount			50 Hertz
F2 Q			99.9999
F2 Pan			Middle
F2 Pan Mod. Source			LFO C Sine
F2 Pan Mod. Amount			100 Percent
F2 Gain			2.0000

Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
F1 Frequency	10	10000	Hz	Sets the center frequency of the first filter.
F1 Freq Mod Src	LFO A Sine	LFO D Ramp		Selects an LFO output to modulate the first filter’s frequency.

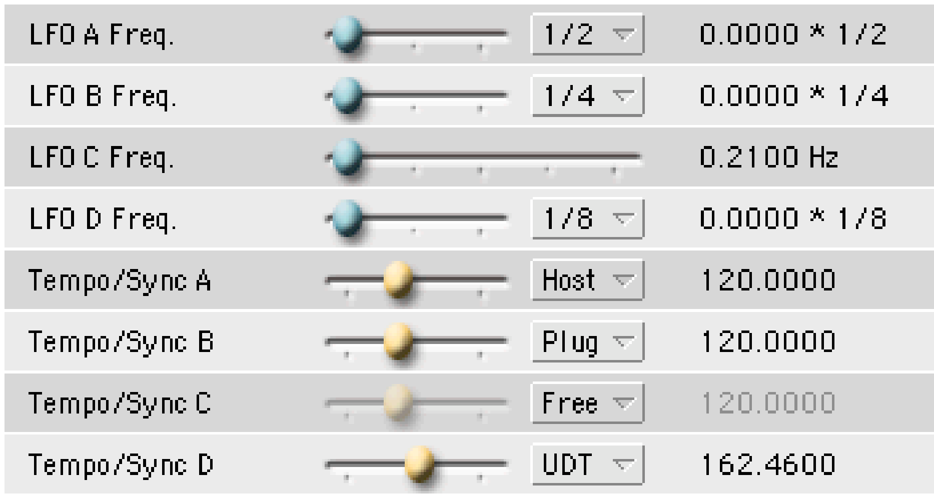
Moving Filters

category: filter

audio input: mono/stereo *cpu: light* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
F1 Freq Mod Amt	-10000	10000	Hz	Sets the range of filter center frequency modulation. Negative values invert the phase of the modulating LFO's signal.
F1 Q	1	400		Sets the resonance or Q factor of the first filter. Q is defined as the center frequency divided by the bandwidth.
F1 Pan	Far Left	Far Right		Sets the stereo position of the first filter's output.
F1 Pan Mod Src	LFO A Sine	LFO D Ramp		Selects an LFO output to modulate the first filter's stereo position.
F1 Pan Mod Amt	-100	100	percent	Sets the range of the first filter's stereo position modulation. Negative values invert the phase of the modulating LFO's signal.
F1 Gain	0	5		Sets the overall gain of the first filter.
F2 Frequency	10	10000	Hz	Sets the center frequency of the second filter.
F2 Freq Mod Src	LFO A Sine	LFO D Ramp		Selects an LFO output to modulate the second filter's frequency.
F2 Freq Mod Amt	-10000	10000	Hz	Sets the range of filter center frequency modulation. Negative values invert the phase of the modulating LFO's signal.
F2 Q	1	400		Sets the resonance or Q factor of the second filter. Q is defined as the center frequency divided by the bandwidth.
F2 Pan	Far Left	Far Right		Sets the stereo position of the second filter's output.
F2 Pan Mod Src	LFO A Sine	LFO D Ramp		Selects an LFO output to modulate the second filter's stereo position.
F2 Pan Mod Amt	-100	100	percent	Sets the range of the second filter's stereo position modulation. Negative values invert the phase of the modulating LFO's signal.
F2 Gain	0	5		Sets the overall gain of the second filter.

LFO Parameters



Name	*Min*	*Max*	*Units*	*Description*
LFO Freq	0	50	Hz	Sets the frequency of the modulating oscillator.
LFO Freq Multiplier	0	25		This parameter is displayed as a slider under LFO Freq when the Sync mode is set to Host, Plug, or UDT. It sets a multiplication factor on the Coarse Pan Freq Units parameter to produce the effective rate of panning relative to the tempo. For example, a value of 2 when the Units parameter is 1/4 gives an effective panning interval of a half note at the current tempo. This parameter is disabled when the Sync mode is set to Free.
LFO Freq Note Value	1	1/64t		This parameter is displayed as a pop-up menu under LFO Freq when the Sync mode is set to Host, Plug, or UDT. It sets a base note duration value that determines the effective rate of panning relative to the tempo. The note duration value is multiplied by the Coarse Pan Freq Mult parameter. For example, a value of 1/4 when the Mult is set to 2 gives an effective panning interval of a half note at the current tempo. This parameter is disabled when the Sync mode is set to Free.
Tempo	1	300	BPM	Tempo is displayed as a slider in the Tempo/Sync parameter. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the rate of panning. When in Host and Plug modes, the tempo is an indicator that cannot be changed--it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range and Audio Rate Pan will calculate the panning rate based on the values of the Coarse Pan Freq Mult and Coarse Pan Freq Units parameters. This parameter is disabled when the Sync mode is set to Free.

Moving Filters

category: filter

audio input: mono/stereo *cpu: light* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Tempo/Sync	Free	UDT		Sets the tempo and synchronization mode for the filter. The pop-up menu lets you choose from four sync modes: • Free mode lets you set LFO rate independent of the host sequencer. • Host mode synchronizes the LFO with the host tempo. • Plug mode sets the LFO from the PluggoSync plug-in if it is loaded. • UDT (User-Defined Tempo) mode lets you select the LFO rate with the Tempo slider. Note: Host mode is only available in VST 2.0 and MAS applications that support it.

Insights

- The negative ranges of the Modulation Amount parameters invert the signals of the LFOs. This lets you modulate two or more parameters with the same wave, but in opposite directions.
- The *Fake Stereoizer* preset sets both filters to a wide bandwidth, one with a low center frequency and one with a high frequency. The filters are set to the far left and right panning settings, spreading a monophonic signal's source across the stereo field.
- The *Insects* preset needs an input signal with a lot of high-frequency content. The filters are both set to a medium bandwidth and fairly high frequency. The A and B LFOs modulate the filter frequencies rapidly to create a sound reminiscent of those annoying insects that congregate in trees in the summer. The C and D LFOs pan the filters rapidly back and forth, so that the insects seem to be all around you.

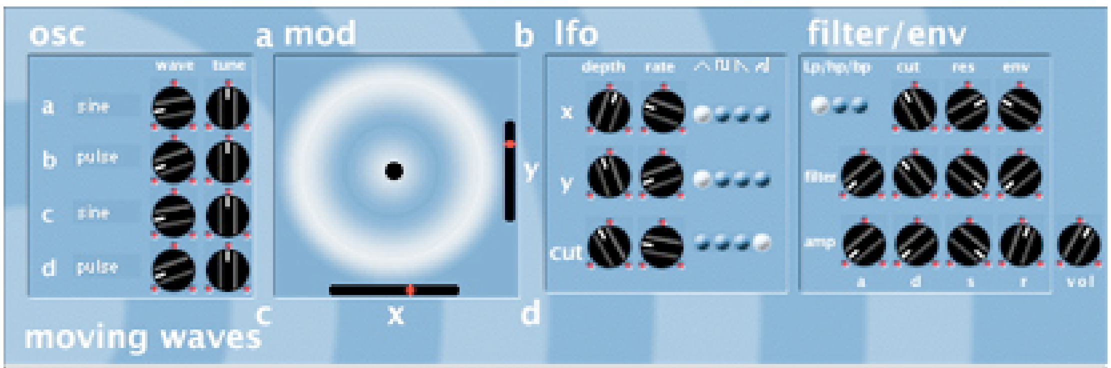
category: synthesis

moving waves

input: MIDI | *cpu: heavy* | *audio output: mono*

What It Does

moving waves is a crossfading synth. Its unique tone is created by using a four-corner (joystick-like) fading system, with X- and Y-axis automation (using built-in LFOs). The four oscillators have 32 different waveforms for a broader range of core tones.



Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Oscillator Waveform (a-d)	0 (saw)	32 (wav27)	18-21 (a-d)	Sets the waveform for each of the four available oscillators. The options are sawtooth, square, triangle, pulse, sine and digital waves 0-27.
Oscillator Tuning (a-d)	0 (-1 semi)	127 (+1 semi)	22-25 (a-d)	Sets the fine tuning of each of the four oscillators.
Joystick Control				Provides crossfading between the four oscillators (in a four-corners orientation). The visual indicators along the right side and bottom provide feedback on the LFO depth and rate effects on oscillator levels. The joystick is not exposed to MIDI Controllers, but can be altered with Pluggo modifiers.
X-axis LFO Depth	0 (0%)	127 (100%)	41	Sets the depth of the X-axis LFO - the amount of change in the crossfade between the A/B and C/D oscillators.
X-axis LFO Rate	0 (0.8 Hz)	127 (20 Hz)	42	Sets the rate of the X-axis LFO.
X-axis LFO Shape	0 (tri)	3 (random)	43	Sets the shape of the X-axis LFO. Values include triangle, square, sawtooth and random.

Name	*Min*	*Max*	*CC#*	*Description*
Y-axis LFO Depth	0 (0%)	127 (100%)	44	Sets the depth of the Y-axis LFO - the amount of change in the crossfade between the A/C and B/D oscillators.
Y-axis LFO Rate	0 (0.8 Hz)	127 (20 Hz)	45	Sets the rate of the Y-axis LFO.
Y-axis LFO Shape	0 (tri)	3 (random)	46	Sets the shape of the Y-axis LFO. Values include triangle, square, sawtooth and random.
Filter Cutoff LFO Depth	0 (0%)	127 (100%)	30	Sets the depth of the LFO's effect on the filter cutoff.
Filter Cutoff LFO Rate	0 (0.8 Hz)	127 (20 Hz)	31	Sets the rate of the filter cutoff LFO.
Filter Cutoff LFO Shape	0 (tri)	3 (random)	32	Sets the shape of the filter cutoff LFO. Values include triangle, square, sawtooth and random.
Filter Type	0 (lowpass)	2 (bandpass)	26	Sets the type of filter response, with options of lowpass, highpass and bandpass.
Filter Cutoff	0 (0 Hz)	127 (10 kHz)	27	Sets the cutoff frequency of the filter.
Filter Resonance	0 (0.0)	127 (1.0)	28	Sets the resonance value (sometimes called Q or pre-emphasis) of the filter. Resonance values range from 0.0 (no resonance) to 1.0 (self-oscillation).
Filter Envelope Modulation Depth	0 (0%)	127 (100%)	29	Sets the amount that the Filter Envelope modulates the filter cutoff.
Filter Envelope Attack	0 (0 ms)	127 (5000 ms)	37	Sets the filter envelope attack rate. When a note is played on the synthesizer, the filter envelope is triggered, and the attack time determines how long it takes the modulation output to move from zero to its maximum value.
Filter Envelope Decay	0 (0 ms)	127 (4000 ms)	38	Sets the filter envelope decay rate. After the envelope has reached its maximum value (based on the Attack parameter), the decay time determines how long it takes for the modulation output to move to its sustain level.
Filter Envelope Sustain	0 (0%)	127 (100%)	39	Sets the filter envelope sustain level. The modulation output will remain at this level until the note is released.
Filter Envelope Release	0 (0 ms)	127 (6000 ms)	40	Sets the filter envelope release rate. This is the amount of time it takes for the modulation output to move to zero after the note has been released.

Name	*Min*	*Max*	*CC#*	*Description*
Amplitude Envelope Attack	0 (0 ms)	127 (5000 ms)	33	Sets the amplitude envelope attack rate. When a note is played on the synthesizer, the amplitude envelope is triggered, and the attack time determines how long it takes the amplitude to move from zero to its maximum value.
Amplitude Envelope Decay	0 (0 ms)	127 (4000 ms)	34	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the amplitude to move to its sustain level.
Amplitude Envelope Sustain	0 (0%)	127 (100%)	35	Sets the amplitude envelope sustain level. The amplitude will remain at this level until the note is released.
Amplitude Envelope Release	0 (0 ms)	127 (6000 ms)	36	Sets the amplitude envelope release rate. This is the amount of time it takes for the amplitude to move back to zero after a note has been released.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

moving waves

category: synthesis

input: MIDI *cpu: heavy* *audio output: mono*

Global Parameters

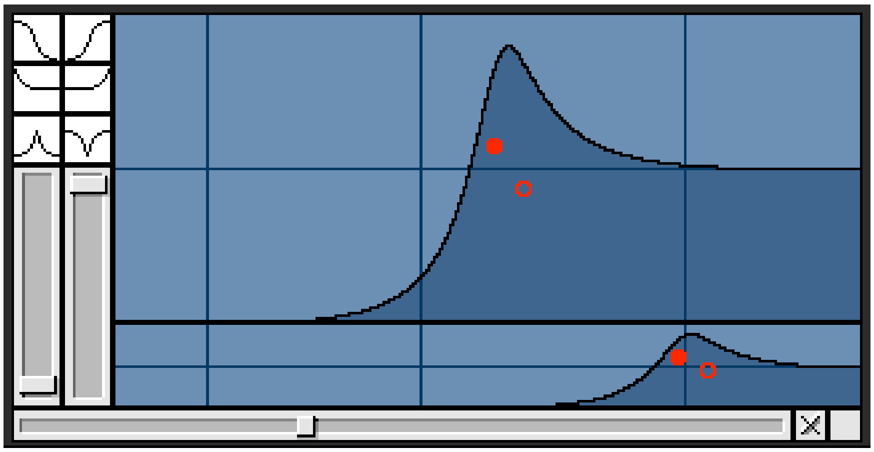
Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.
Polyphony	1	32	voices	Adjusts the number of available sample playback voices. Reducing this number may reduce CPU use.

Insights

- As the name might suggest, *moving waves* is all about motion. Don't treat this like a simple synthesizer - take full advantage of the LFO's, joystick controller and envelopes to add some real-time animation.

What It Does

Multi-Filter gives you two multi-mode filters with variable resonance. The cutoff frequencies of the two filters can be linked, with a variable separation. Each of the filters has a 12 dB/octave response, so linking them with no separation can give you a single, 24 dB/octave filter.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the output gain of the filtered signal.
Dry Level	0	1		Sets the output gain of the original input signal.
Filter Type	Low Pass	Notch		Selects the mode of the two filters. The choices are Low Pass, High Pass, Low Shelf, High Shelf, Band Pass, and Notch.
Filter 1 Cutoff	20	20000	Hz	Sets the cutoff frequency of the first filter. If Link Filters is enabled, this will set the cutoff frequency for the second filter as well, offset by the Separation parameter (see below).
Filter 1 Resonance	0	15	Q	Sets the resonance ("Q") characteristic of the first filter. Q is defined to be cutoff frequency divided by filter bandwidth. If Link Filters is enabled, this will set the resonance for the second filter as well.
Filter 1 Gain	0	2		Sets the gain of the first filter. If Link Filters is enabled, this will set the gain for the second filter as well.
Separation	0	10000	Hz	This sets the separation between the cutoff frequencies of the two filters. This will only have an effect when Link Filters is enabled.
Filter 2 Cutoff	20	20000	Hz	Sets the cutoff frequency of the second filter. The value of this parameter will be overridden by the Filter 1 Cutoff parameter if Link Filters is enabled.

Multi-Filter

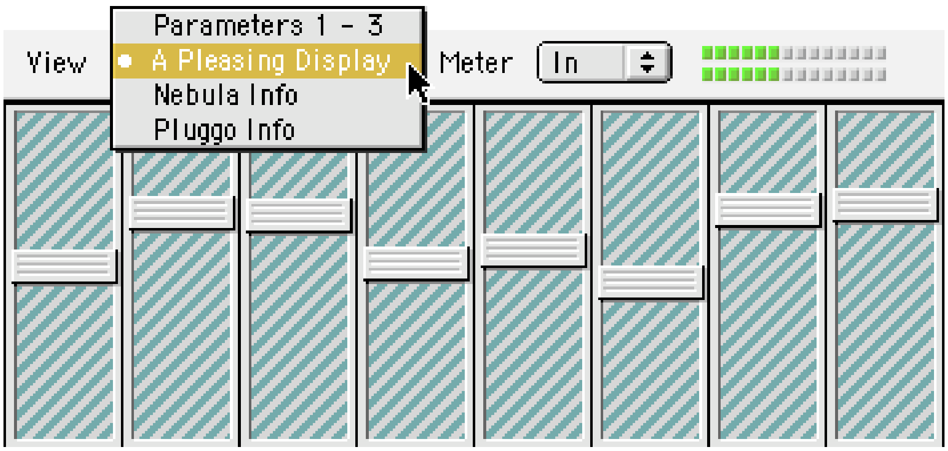
category: filter

audio input: mono *cpu: light* *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Filter 2 Resonance	0	15	Q	Sets the resonance (“Q”) characteristic of the second filter. Q is defined to be cutoff frequency divided by filter bandwidth. The value of this parameter will be overridden by the Filter 1 Resonance parameter if Link Filters is enabled.
Filter 2 Gain	0	2		Sets the gain of the second filter. The value of this parameter will be overridden by the Filter 1 Gain parameter if Link Filters is enabled.
Link Filters	0	1		This is a switch that sets the cutoff, resonance, and gain parameters of filter 2 to the corresponding values of filter 1.
Mute Filter	0	1		This turns off the output of the second filter. This is handy to hear the 12 dB/octave response that a single filter gives you.

What It Does

Nebula applies amplitude and phase-inversion changes to the input sound to create a subtle swirling, stereo illusion. It works well on textural sounds, like pads and beds. And bed pads.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Output	0	2		Sets the output signal level.
Nebula Depth	0	127		Sets the amount by which the volume level of the nebula arms can change.
Swirl Activity	0	1		Controls of the rate of movement within the nebula.

Insights

- You might be wondering what the Pleasing Display (available in the View menu) represents. These are the eight volume levels that are being randomly generated and then mixed together. From left to right, we have: left into left, left phase-reversed into left, right into left, right phase-reversed into left, left into right, left phase-reversed into right, right into right, and right phase-reversed into right.

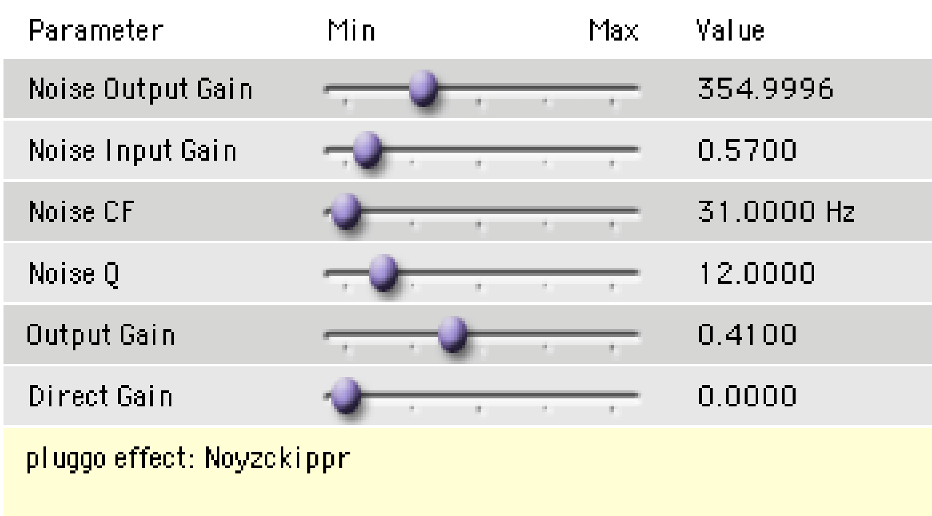
Noyzckippr

category: distortion

audio input: mono | *cpu: light* | *audio output: mono*

What It Does

Noyzckippr multiplies—yes, multiplies—the input signal by bandpass-filtered white noise. It's a highly non-linear effect that ranges from irregular tremolo at very low filter center frequencies to relatively useless simulations of poor radio reception at higher center frequencies.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Noise Output Gain	0	1200		Sets the gain on the filtered noise before it multiplies the input signal.
Noise Input Gain	0	7		Sets the level of the noise before filtering.
Noise CF	0	4000	Hz	Sets the center frequency of the filter applied to the noise. The most useful effects will be found in the lower ranges of this parameter.
Noise Q	0	80		Sets the resonance or Q factor of the filter applied to the noise. Watch out: the output level of the filter will increase dramatically when the Q is lowered.
Output Gain	0	1		Sets the overall output gain of the effect after the input signal has been multiplied by the filtered noise.
Direct Gain	0	1		Sets the output gain of the original input signal.

Insights

- The *Harsh Tremolo* preset creates an appealing AM effect that is not as regular as a simple sine wave. The secret is filtering the noise with low CF and very high Q.
- The *Soft Distortion* preset resembles poor radio reception in its inability to keep the signal level constant. There is also a bit of distortion when you actually do hear the signal.

- The *Noise Mod* preset is the sort of thing that's useful for adding some edge to sweet vocals.

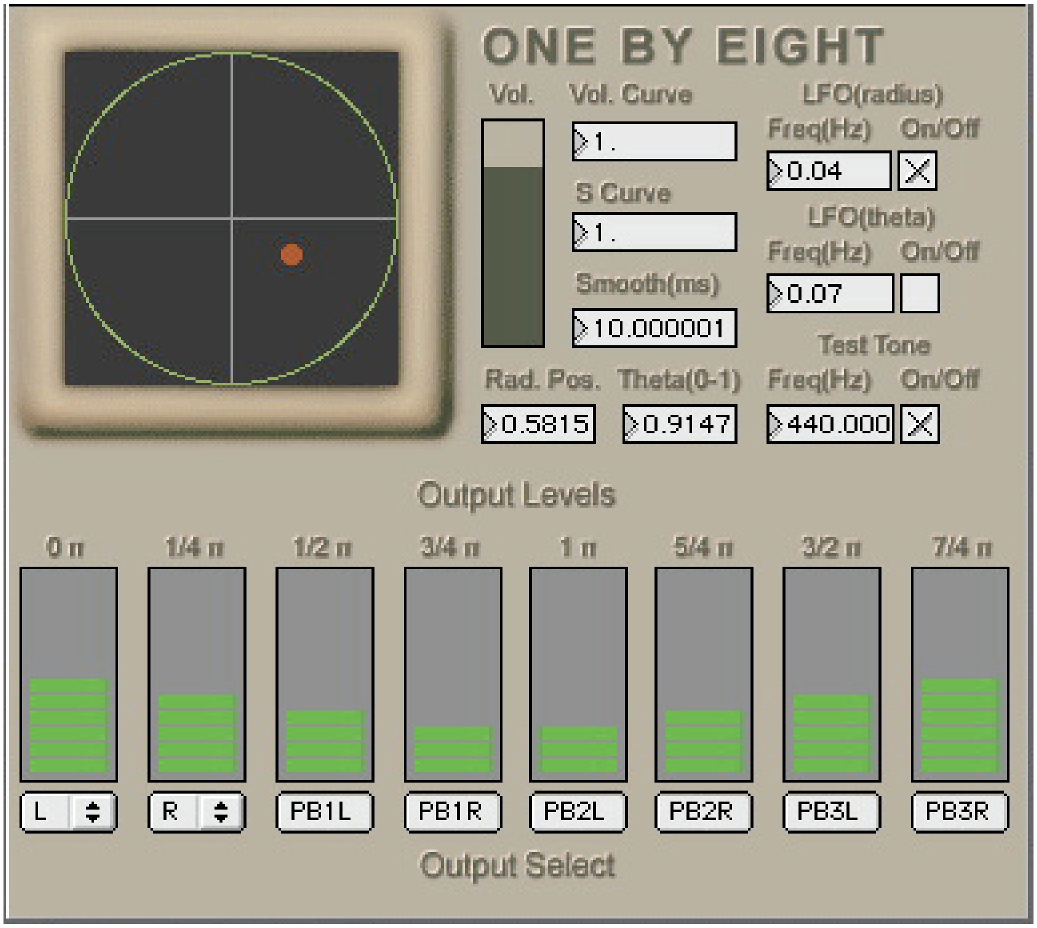
OneByEight

category: multichannel

audio input: stereo *cpu: light* *audio output: stereo*

What It Does

OneByEight is a one-input, eight-output matrix mixer. It uses polar coordinates to mix its input to eight outputs (a stereo input is mixed to a mono signal). The amount sent to each of the eight outputs is calculated using r (radius) and t (theta). The eight outputs are positioned equidistant along the perimeter of the unit circle at angles 0, /4, /2, 3/4, , 5/4, 3/2, and 7/4. Each output can be assigned to either the left or right audio output channels, or to a *PluggoBus* channel.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Vol.				Sets the overall output amplitude of the plug-in (i.e. the maximum amplitude which will be routed to any selected output channels
Vol. Curve	0.0	10.0		Sets the curve for input volume. If the volume curve value is less than 1.0 and greater than 0.0, the curve will be logarithmic. A value of 1.0 will set a linear volume curve, and values greater than 1.0 will produce exponential volume curves.

audio input: stereo *cpu: light* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
S Curve	0.0	100.0		Sets the spatialization curve for various output points on the unit circle whose outputs are assigned using the Output Select. volume. S curve values less than 1.0 and greater than 0.0 will produce logarithmic output curves. A value of 1.0 will set a produce linear output, and values greater than 1.0 will produce exponential S curves.
Smooth	0	2000	msec.	Smoothing is used to provide a smoother and more gradual transition time in milliseconds as the output signal is routed between the various points on the unit circle.
LFO (radius)	0	20	Hz.	Sets the frequency of an LFO whose output provides the radius value used in routing the input signal to the various output positions on the unit circle. The LFO toggle enables and disables the LFO. If the LFO is not selected, the radius value can be entered manually in the Rad. Pos box, or by dragging the display dot on the unit circle display.
LFO (theta)	0	20	Hz.	Sets the frequency of an LFO whose output provides the theta value used in routing the input signal to the various output positions on the unit circle. The LFO toggle enables and disables the LFO. If the LFO is not selected, the theta value can be entered manually in the Theta(0-1) box, or by dragging the display dot on the unit circle display.
Rad. Pos	0.0	1.0		Sets the radius value for output generation if the LFO(radius) parameter is not enabled. A value of 0 will position the display dot in the center of unit circle display. Values between 0 and 1.0 will move the display dot outward toward the edge of the unit circle (1.0) along a line whose angle is set by the current theta value.
Theta(0-1)	0.0	1.0	Hz.	Sets the theta value for output generation if the LFO(theta) parameter is not enabled. A value of 0 or 1.0 will position the display dot at the three-o'clock position on the unit circle display. Values between 0 and 1.0 will rotate the display dot in a counter-clockwise direction.
Test tone	0	2000	Hz.	Sets the frequency for a test tone. The test tone toggle turns the text tone on and off.
Output Select	Off	Pluggo Bus 4R		Each of the eight positions on the Unit Circle has a pop-up menu which selects the *PluggoBus* source for the output channel. You can mute or route each output to the left or right audio outputs of your plug-in, or any *PluggoBus* output channel.

Insights

- Use the test tone and the led monitors to get an idea of the signal level being routed to each output.
- You can use *OneByEight* to route output rhythmically to PluggoBus channels by setting the output selects to different audio channels which are each running a different *PluggoBusRcv* plug-in.

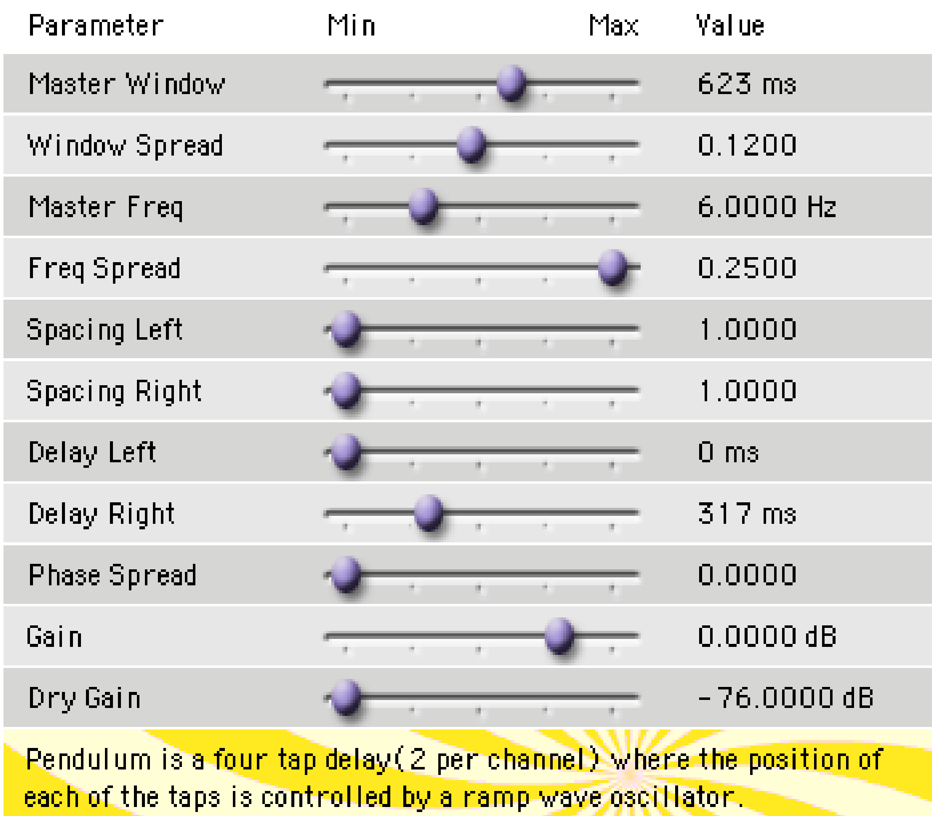
Pendulum

category: granular

audio input: mono, stereo | *cpu: light* | *audio output: stereo*

What It Does

Pendulum is a four-tap delay with two taps per channel where the delay time each of the taps is controlled by a ramp wave oscillator. The window parameters define the range of delay times generated by the ramp oscillator. The spread controls determine how much each tap's parameters differ from one another.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Master Window	1	1000	ms	Sets master range of delay times.
Window Spread	0	0.25		Sets the difference between the delay time window sizes.
Master Freq	0	20	Hz	Master rate of tap delay time movements.
Frequency Spread	0	0.25		Sets the difference between the delay time modulation frequencies.
Spacing Left	1	10		Sets the amount of silence between left delay time ramps.
Spacing Right	1	10		Sets the amount of silence between right delay time ramps.
Delay Left	0	1000	ms	Sets the delay offset (before modulation) of the left delay.
Delay Right	0	1000	ms	Sets the delay offset (before modulation) of the right delay.

Name	*Min*	*Max*	*Units*	*Description*
Phase Spread	0	0.25		Sets the amount of difference between the initial phases of the delay times.
Gain	-76	+18	dB	Sets the output level of the effect.
Dry Gain	-76	+18	dB	Sets the level of the unprocessed input signal.

Insights

- If you think in analog metaphors, it might be helpful to think of the variable delay time as a moving playback head on a tape recorder. The tape moves through the machine at a constant speed, but the head can move backwards and forwards. When you move the head, the effective playback speed of the tape changes, *but only as long as the head keeps moving*. Have you ever walked while riding on an escalator or moving sidewalk? If you walk forward, you move faster than the escalator, but if you walk backward, you move more slowly. If you can walk backward faster than the forward speed of the escalator, the effective speed is negative. In the case of tape playback, that would mean that the sound started to play backwards.
- You can achieve panning and phase effects with high Phase Spread settings.
- You can make rhythmic effects with high Spacing settings.

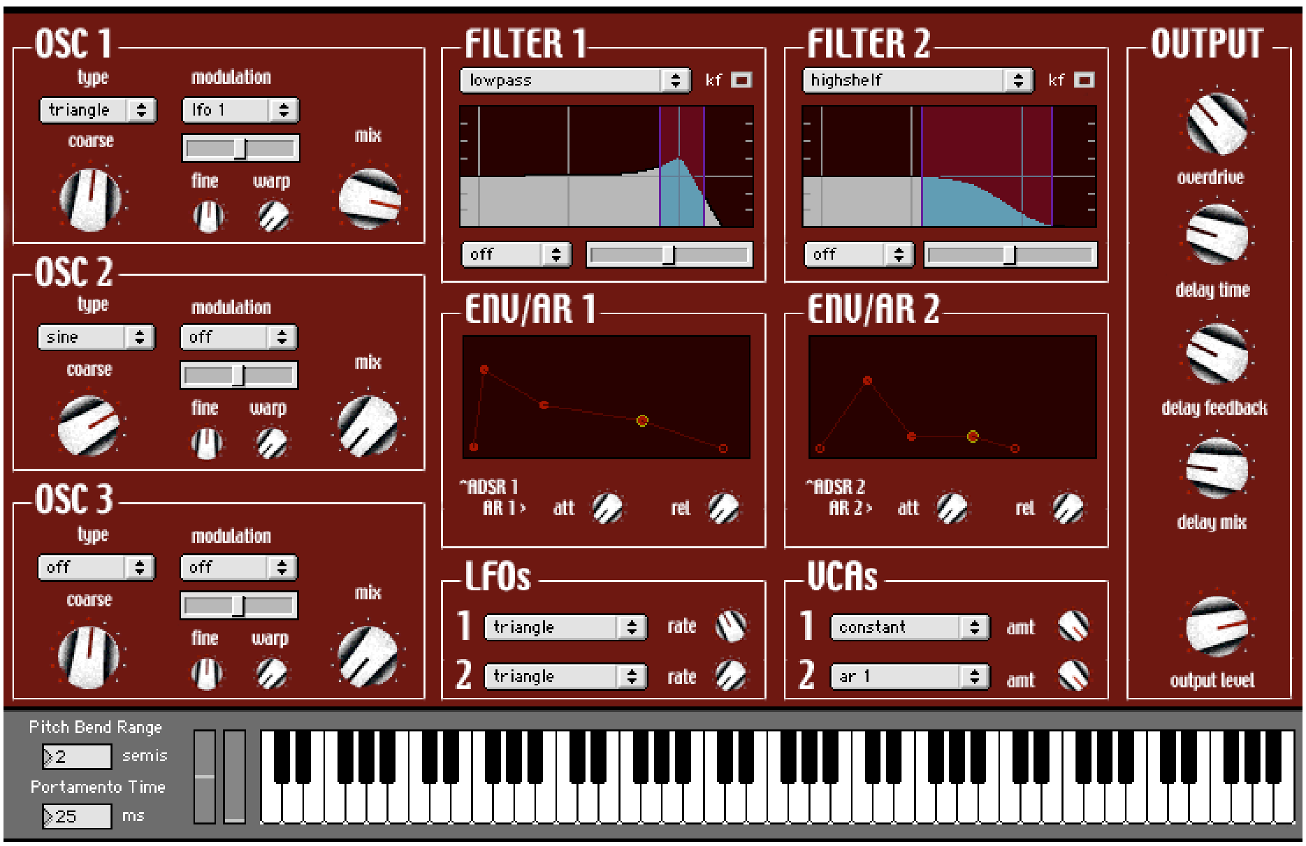
pgs-1

category: synthesis

input: MIDI | *cpu: heavy* | *audio output: mono*

What It Does

pgs-1 (Pretty Good Synth version 1) is an analog-style monosynth with some extreme flexibility and a whole lotta filtering. Three oscillators, two filters and four envelopes are the start of this machine. An integrated effects section (with overdrive and digital delay) allows presets to provide a total instrument experience.



Visible Parameters

Name	*Min*	*Max*	*Description*
Oscillator Type	0 (off)	5 (noise)	Sets the waveform of each of the oscillators. Options include off, sine, pulse, saw, triangle and noise.
Coarse Tuning	0 (-32 semi)	64 (+31 semi)	Sets the coarse pitch offset applied to the oscillators.
Oscillator Modulation Source	0 (off)	6 (lfo 2)	Sets the source of pitch modulation for the oscillators. Options include off, adsr 1, adsr 2, ar 1, ar 2, lfo 1 and lfo 2.
Oscillator Modulator Amount	0 (-100 %)	200 (+ 100%)	Sets the amount of pitch modulation to an oscillator by the selected source. A negative value provides inverted modulation.

Name	*Min*	*Max*	*Description*
Fine Tuning	0 (-20 cents)	40 (+20 cents)	Sets the fine pitch offset applied to the oscillators.
Oscillator Warp	0 (0%)	100 (100%)	Alters the waveform by “warping” the waveform playback rate. The effect varies depending on the Oscillator Type, but generally the sound becomes more complex.
Oscillator Mix	0 (0%)	127 (100%)	Adjusts the individual levels of the three oscillators to create the oscillator mix.
Filter Type	0 (lowpass)	7 (hi-shelf)	Sets the filter type for the individual filters. Options include display-only, lowpass, highpass, bandpass, bandstop, peaknotch, lowshelf and highshelf.
Filter Key Follow	0 (off)	1 (on)	Determines whether the filter cutoff will follow the current MIDI note. This will allow the filter to “brighten” when the incoming MIDI note is higher.
Filter Edit Display			The Filter Edit Display provides complete control of the filter with visual feedback. The filter cutoff can be adjusted by clicking in the center of the edit “selection” area. The filter Q is adjusted by clicking on one of the selection “fences” and extending / contracting it. The filter gain (for shelving and peaking filter types) is altered by selecting the horizontal gain indicator and dragging it up-or-down.
Filter Modulation Source	0 (off)	7 (velocity)	Sets the source of each filter modifier. Options include off, adsr 1, adsr 2, ar 1, ar 2, lfo 1, lfo 2 and velocity.
Filter Modulation Depth	0 (-100%)	200 (+100%)	Sets the amount of modulation of the selected source on the filter cutoff value.
ADSR Envelope Editor			The ADSR envelope editors provide a five-point graphic display of the envelope stages. You can set the values by clicking and dragging the “points” of the graph. The yellow-ringed point represents the sustain level of the envelopes.
AR Attack	0 (0 ms)	99 (2000 ms)	When a new MIDI note is struck, the AR envelopes start running. The Attack rate determines the time for the modulation level to move from 0% to 100%.
AR Release	0 (0 ms)	99 (2000 ms)	When a MIDI note is released, the AR envelope begins its release stage. The Release rate determines the time for the modulation level to drop from 100% to 0%.
LFO Shape	0 (off)	5 (down-saw)	Determines the shape of the LFO modifier. Options include off, triangle, sine, square, up-saw and down-saw.

pgs-1

category: synthesis

input: MIDI | *cpu: heavy* | *audio output: mono*

Name	*Min*	*Max*	*Description*
LFO Rate	0 (0.01 Hz)	127 (10 Hz)	Sets the rate of the LFO modifier.
VCA Source	0 (constant)	6 (lfo 2)	Determines the source of VCA modulation. Options are constant (set value), adsr 1, adsr 2, ar 1, ar 2, lfo 1 and lfo 2.
VCA Amount	0%	99%	Sets the amount of modulation (or the base value for constant settings) by the source to the output level.
Overdrive	0%	100%	Sets the amount of overdrive (soft distortion) applied to the output signal.
Delay Time	0 ms	2560 ms	Sets the delay time of the digital delay
Delay Feedback	0%	99%	Sets the amount of delay feedback in the delay circuit.
Delay Mix	0 (0:100)	127 (100:0)	Sets the ration of wet to dry signal.
Output Level	0 (0%)	127 (100%)	Sets the output level of the synthesizer signal.
Pitch Bend Range	2 semis	12 semis	Sets the extent of pitchbend provided by the pitch bend controller.
Portamento Time	0 ms	5000 ms	Determines the amount of time required for the output to reach the MIDI note pitch.

Hidden Parameters

Name	*Min*	*Max*	*Description*
Filter Frequency	20 Hz	17000 Hz	Sets the cutoff frequency of the filter. In the user interface, this is normally set using the filter edit display.
Filter Gain	0.0001	16.0	Sets the filter gain (determines the positive or negative extent of the filter). In the user interface, this is normally set using the filter edit display.
Filter Resonance	0.01	25.0	Sets the filter resonance of the filter at the cutoff point. In the user interface, this is normally set using the filter edit display.
Control Simulations			The two controllers (for pitchbend and mod wheel) and the keyboard allow you to create and modify synth sounds without having to use a MIDI keyboard.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Mod Wheel Range – Oscillator 1	-100	+100	percent	Sets the amount of modulation offset provided by the mod wheel control to the Oscillator 1 depth control.
Mod Wheel Range – Oscillator 2	-100	+100	percent	Sets the amount of modulation offset provided by the mod wheel control to the Oscillator 2 depth control.
Mod Wheel Range – Oscillator 3	-100	+100	percent	Sets the amount of modulation offset provided by the mod wheel control to the Oscillator 3 depth control.
Mod Wheel Range – Filter 1	-100	+100	percent	Sets the amount of modulation offset provided by the mod wheel control to the Filter 1 depth control.
Mod Wheel Range – Filter 2	-100	+100	percent	Sets the amount of modulation offset provided by the mod wheel control to the Filter 2 depth control.

Insights

- The keyboard, mod wheel and pitch bend wheel on the user interface are working controllers for the *pgs-1* synthesizer. They are handy when you are doing sound design (or laptop work).
- By far, the most important element in the sound of this module is the filter bank. In addition to the two filters, there are some “hidden” overdrive circuits that attempt to emulate the fuzziness of real analog synthesizers.
- Using the Mod Wheel to affect the oscillators and filters can change an ordinary sound into a living and breathing sound. Make sure to flip over to the “egg slider” interface and add some mod wheel control.
- Many classic synthesizers (such as the Roland JD-800 and Access Virus) are defined as much by their effects settings as their raw sound. The *pgs-1* includes an effects process for just that reason: to create presets that interface to add some mod wheel control.

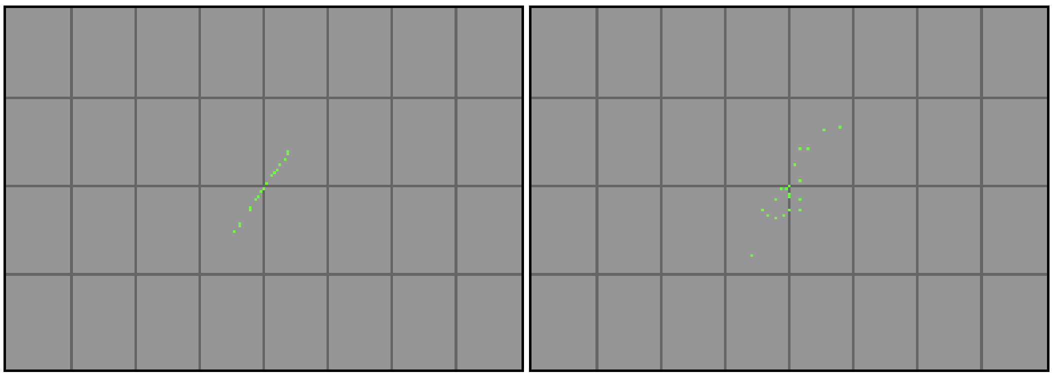
Phase Scope

category: visual display

audio input: stereo | *cpu: light* | *audio output: none*

What It Does

Phase Scope is a utility that lets you check the phase alignment of a stereo signal. If the left and right inputs are in phase, you'll see a line similar to the one shown in the left screen shot. If the left and right inputs are not in phase, you might see something like the screen shot on the right.



Insights

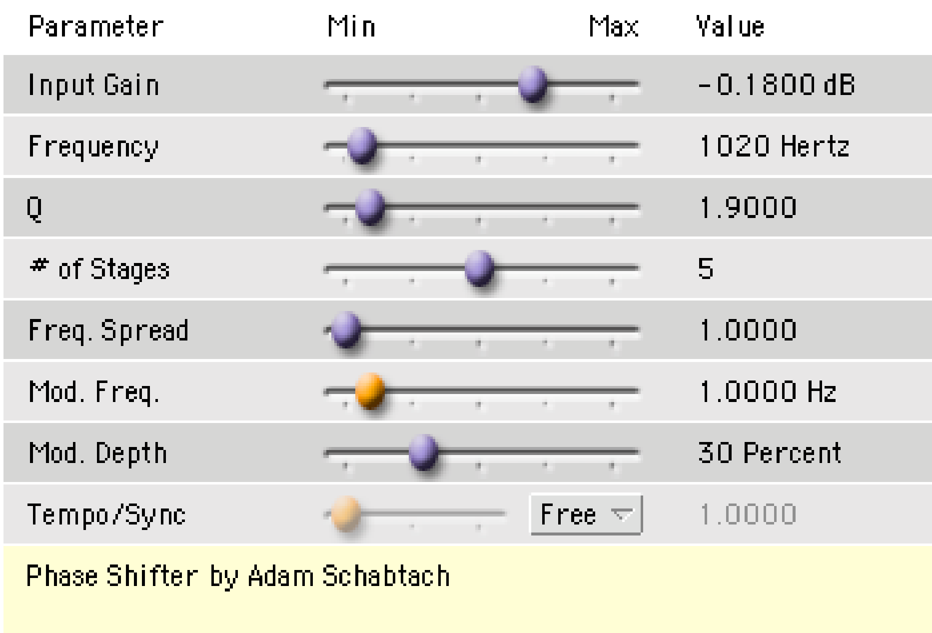
- The incoherence of effects such as *Swirl* and *Rye* that use phase inversion is quite visible when displayed with *Phase Scope*.

category: filter	*accepts sync*	**Phase Shifter**
audio input: mono	*cpu: light*	*audio output: mono*

What It Does

Phase Shifter is a digital emulation of an old analog stomp-box effect popular with guitarists of yesteryear. A phase shifter creates its characteristic sound by applying very short, frequency-dependent delays to a signal and mixing the delayed signal with the original. The effect is similar to a flanger, but usually more subtle.

Rather than using a delay line, which shifts all frequencies equally, a phase shifter uses *allpass* filters. Allpass filters have an equal amplitude response to all frequencies, but a phase response which varies with the frequency of the input signal.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Gain	0	2		Adjusts the level of the input signal. Turn this up if you have a weak signal; turn it down if you hear clipping.
Frequency	20	15000	Hz	Tunes the allpass filters. Loosely speaking, this setting is the center of the frequency range over which the allpass filters have a changing amount of phase shift.
Q	1	10		Sets the Q factor of the filters. In this case, Q refers to the range of frequencies over which the allpass filter has a changing amount of phase shift. Generally speaking, a lower Q produces a more dramatic effect.
# of Stages	1	8		Sets the number of allpass filters through which the signal passes.

Phase Shifter

accepts sync | *category: filter*

audio input: mono | *cpu: light* | *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Freq Spread	0	1		Adjusts the ratio of frequencies between the filters. Zero means that the filters all have the same frequency.
Mod Freq	0	10	Hz	Adjusts the frequency of the oscillator that modulates the frequencies of the allpass filters.
Mod Freq Mult	0	25		This parameter is displayed as a slider under Mod Freq when the Sync mode is set to Host, Plug, or UDT. It sets a multiplication factor on the Mod Freq Units parameter to produce the effective rate of the oscillator that modulates the frequencies of the allpass filters relative to the tempo. For example, a value of 2 when the Units parameter is 1/4 gives an effective rate of a half note at the current tempo. This parameter is disabled when the Sync mode is set to Free.
Mod Freq Note Value	1	1/64t		This parameter is displayed as a pop-up menu under Mod Freq when the Sync mode is set to Host, Plug, or UDT. It sets a base note duration value that determines the rate of the oscillator that modulates the frequencies of the allpass filters. The note duration value is multiplied by the Mod Freq Mult parameter. For example, a value of 1/4 when the Mult is set to 2 gives an effective panning rate of a half note at the current tempo. This parameter is disabled when the Sync mode is set to Free.
Mod Depth	0	100	percent	Adjusts the amount of modulation applied to the frequency of the allpass filters.
Tempo	1	300	BPM	Tempo is displayed as a slider in the Tempo/Sync parameter. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the modulating oscillator rate. When in Host and Plug modes, the tempo is an indicator that cannot be changed--it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range and Audio Rate Pan will calculate the modulating oscillator rate based on the values of the Coarse Pan Freq Mult and Coarse Pan Freq Units parameters. This parameter is disabled when the Sync mode is set to Free.

Name	*Min*	*Max*	*Units*	*Description*
Sync Mode	Free	UDT		Sync Mode is a pop-up menu in the Tempo/Sync parameter. There are four modes of sync available: • Free mode lets you set the modulating oscillator rate in Hertz independent of the host sequencer. • Host, mode synchronizes the modulating oscillator rate to the host tempo. • Plug mode synchronizes the modulating oscillator rate to the beat output of PluggoSync • UDT (User-Defined Tempo) mode is equivalent to Free mode but allows you to set the modulating oscillator rate in terms of a tempo and note unit values. Note: Host mode is only available in VST 2.0 and MAS applications that support it.

Insights

- Hardware phase shifters differ largely by the number of allpass filters they contained, and their (usually fixed) frequency and Q values. The *Phase Shifter* effect gives you far more control than most (or all?) of its hardware counterparts.
- Setting the Frequency Spread parameter to zero usually creates the most dramatic phase-shifting effect. Setting it to some other value diffuses the effect somewhat, which can be interesting when applied to signals with broad frequency content.

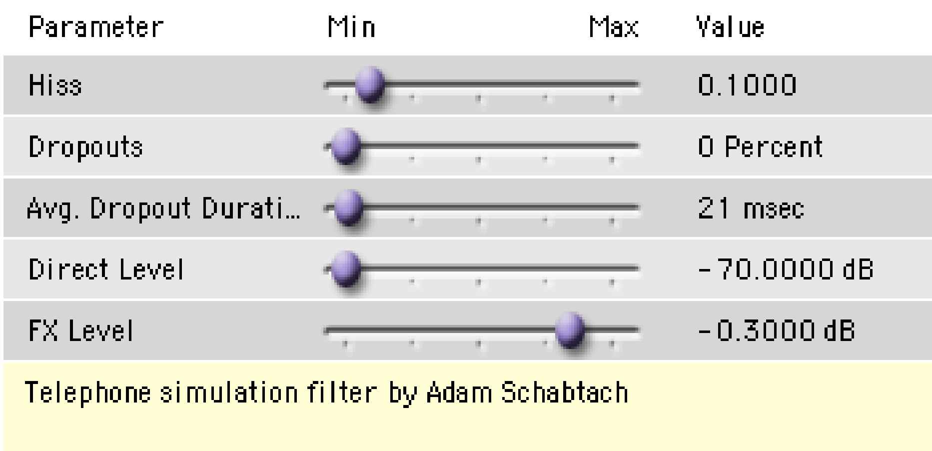
Phone Filter

category: filter

audio input: mono *cpu: light* *audio output: mono*

What It Does

Phone Filter makes its input sound like it's coming over the phone by distorting it slightly and limiting its bandwidth and dynamic range.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Hiss	0	1		Adds background hiss.
Dropouts	0	100	percent	When non-zero, the signal will drop out randomly a specified percentage of the time. Useful in emulating the effect of talking on a cellular phone.
Avg Dropout Duration	0	1000	ms	Sets the average length of time that the signal will stay off when it drops out.
Direct Level	-70	+12	dB	Sets the gain on the unprocessed input signal.
FX Level	-70	+12	dB	Sets the output gain on the effect.

Insights

- The "telephone voice" effect is something of an overused cliché in pop music, and should probably be avoided. But then, the same is true of the I-IV-V chord progression. Let your artistic conscience be your guide.

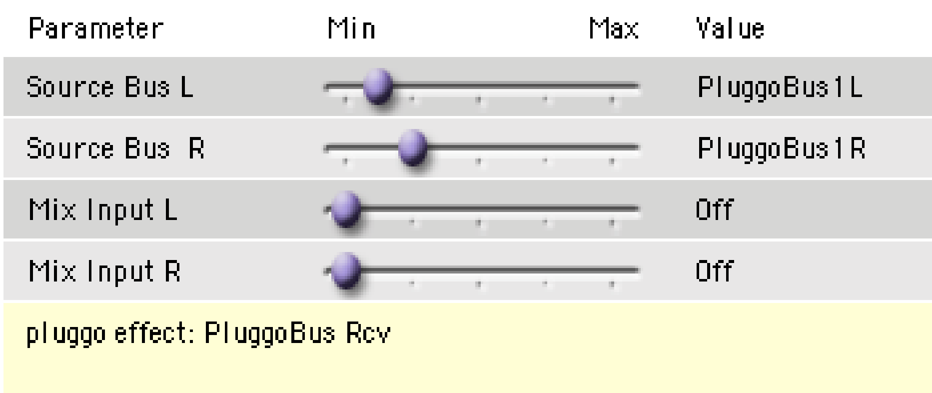
category: audio routing

PluggoBus Rcv

audio input: pluggoBus | *cpu: light* | *audio output: mono, stereo*

What It Does

PluggoBus Rcv accepts audio input from the PluggoBus and sends it to one or both of its plug-in outputs.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Source Bus L	Pluggo Bus 1L	Pluggo Bus 4R		Select the PluggoBus source for the left output channel.
Source Bus R	Pluggo Bus 1L	Pluggo Bus 4R		Select the PluggoBus source for the right output channel.
Mix Input L	Off	On		Sets whether the plug-in’s left input channel is mixed with the PluggoBus audio for the left output channel.
Mix Input R	Off	On		Sets whether the plug-in’s right input channel is mixed with the PluggoBus audio for the right output channel.

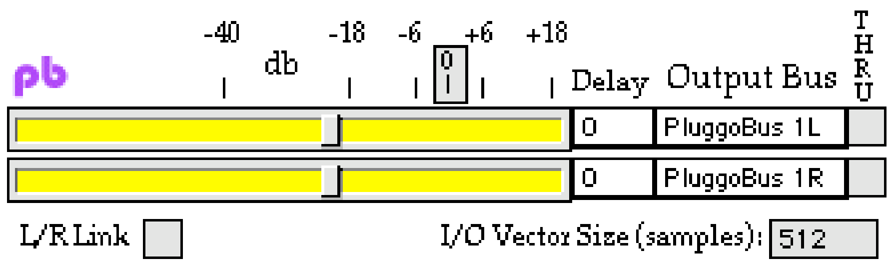
PluggoBus Send

category: audio routing

audio input: mono, stereo | *cpu: light* | *audio output: pluggoBus*

What It Does

PluggoBus Send puts its audio input signal(s) on the PluggoBus so that they can be sent to other plug-ins that are PluggoBus aware. Examples include the *Vocoder* and *Convolver*, where you want to be able to use two input sources that wouldn't necessarily be grouped together in a stereo pair.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Send Level L	-inf	+18	dB	Set the level for the bus sent from the plug-in's left input channel.
Send Level R	-inf	+18	dB	Set the level for the bus sent from the plug-in's right input channel.
Output Bus L	Pluggo Bus 1L	Pluggo Bus 4R		Select the PluggoBus that will receive the input from the left channel.
Output Bus R	Pluggo Bus 1L	Pluggo Bus 4R		Select the PluggoBus that will receive the input from the right channel.
Delay to Bus L	0	1024	samples	Sets a delay between the plug-in's left input and the bus output.
Delay to Bus R	0	1024	samples	Sets a delay between the plug-in's right input and the bus output.
Audio Thru L	Off	On		Selects whether the left audio input channel is echoed through the plug-in's left output.
Audio Thru R	Off	On		Selects whether the right audio input channel is echoed through the plug-in's right output.
Stereo Link	Off	On		If checked, settings are identical for both left and right channels.

Insights

- *PluggoBus Send* lists the I/O Vector Size in the bottom corner of its window. This value may be used as a guide for how many samples you may need to delay the input before sending it to the PluggoBus in order to achieve the proper synchronization when mixing something from the PluggoBus with a plug-in input. Why is this necessary? Because the sequencer's mixer processes audio in a "chain" and you may want to send something from a plug-in that occurs later in the chain to a plug-in that occurs earlier in the chain. That will introduce a delay, because the next opportunity that the earlier plug-in will have to receive the audio will be after the entire audio processing chain has been completed and starts over again.

PluggoFuzz

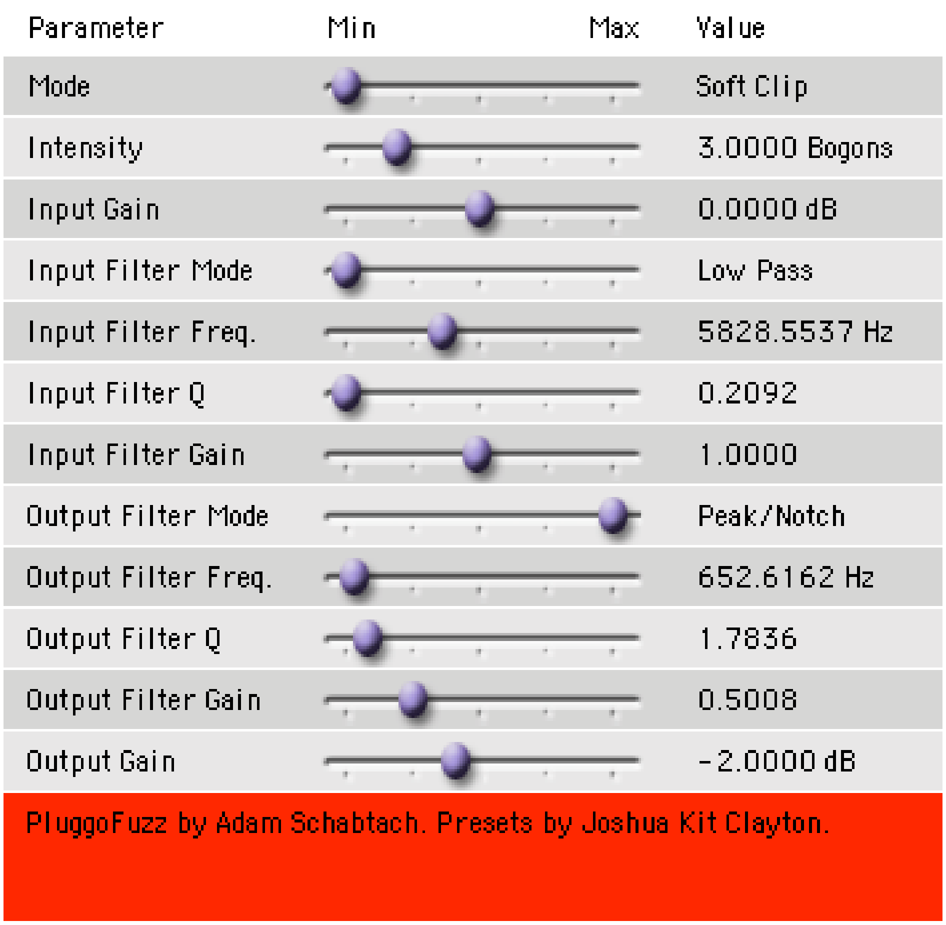
category: distortion

audio input: mono | *cpu: light* | *audio output: mono*

What It Does

PluggoFuzz is a distortion plug-in. It distorts signals and makes them fuzzy. And it does this with four clipping modes and multi-mode filters at its input and output stages.

Distortion effects are, of course, primarily of interest to guitarists, but there's no need to let them have all the fun. Particularly in recent years, distortion has become a popular signal-enhancement effect for synthesizers, drums loops, vocals, and cellos.



Visible Parameters

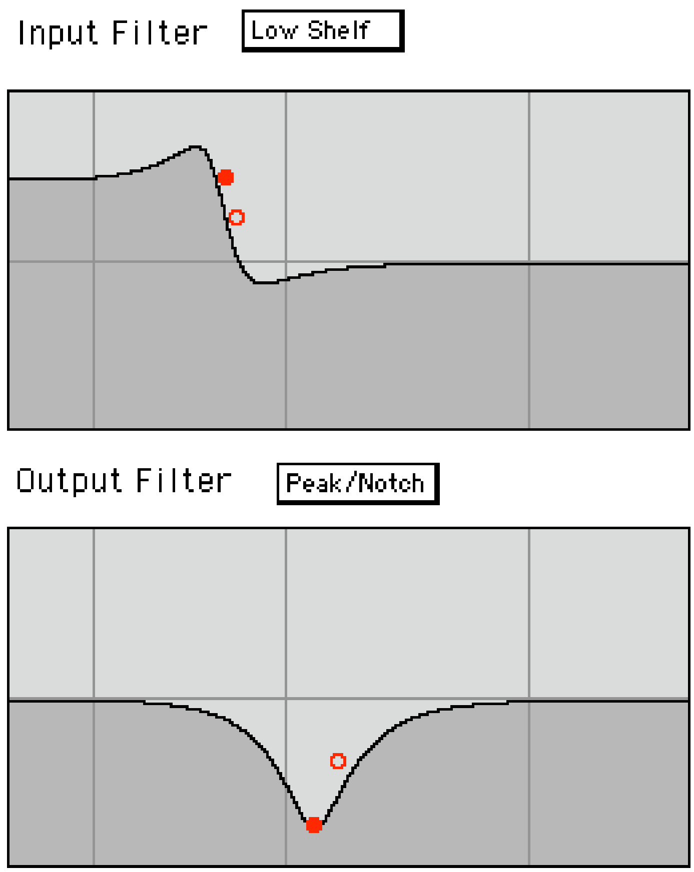
Name	*Min*	*Max*	*Units*	*Description*
Mode	Soft Clip	Wrap		Sets the distortion mode. See below for descriptions of the different modes
Intensity	1	11	Bogons	Sets the amount of distortion applied to the input signal. 1 is the minimum amount of distortion, and 11 is the maximum amount. A Bogon is a unit of distortion not yet a part of the metric system.

PluggoFuzz

audio input: mono *cpu: light* *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Input Gain	-12	12	dB	Boosts or attenuates the input signal before it enters the input filter.
Input Filter Mode	Low Pass	Peak/ Notch		Chooses the frequency-response characteristic of the input filter.
Input Filter Frequency	20	16000	Hz	Sets the center or corner frequency of the input filter.
Input Filter Q	0.01	20		Sets the resonance or Q factor of the input filter. The effect of this parameter varies with the filter mode; see the filter graph displays to get an idea of its effect.
Input Filter Gain	0.01	2		Adjusts the output level of the filter.
Output Filter Mode	Low Pass	Peak/ Notch		Chooses the frequency-response characteristic of the output filter.
Output Filter Frequency	20	16000	Hz	Sets the center or corner frequency of the output filter.
Output Filter Q	0.01	20		Sets the resonance or Q factor of the output filter. The effect of this parameter varies with the filter mode; see the filter graph displays to get an idea of its effect.
Output Filter Gain	0.01	2		Adjusts the output level of the filter.
Output Gain	-12	12	dB	Boosts or attenuates the final output signal.

Equalization View



- In its Equalization view, *PluggoFuzz* displays two multi-mode filters that can be applied to the signal input—before it is distorted—as well as the signal output—after it is distorted.

Insights

- In the ongoing quest for The Ultimate Guitar Tone, it has been discovered that how you filter or EQ the distorted signal is at least as important as how you distort it. Commercial stomp-box distortion pedals typically have some rudimentary filters for "tone controls", and often have non-adjustable input filters. With flexible filtering both before and after the distortion stage, *PluggoFuzz* attempts to provide a wide range of distortion characteristics.
- Hardware distortion units all work more or less the same way: they amplify a signal beyond the limits of some circuit, so that the peaks of the signal are chopped off when those limits are reached. The difference in tone between tone between tube-based and transistor-based amplifiers, when overdriven into distortion, is partly based on how the circuits behave as they approach their limits. Transistor-based circuits hit their maximum output levels abruptly, and chop the signals off with sharp corners. (Essentially they turn sine waves into square waves.) Tube-based circuits start to distort in a more gradual fashion, and produce rounded corners on the clipped output signal (turning sine waves into square waves with round corners.) The sharper corners create more harmonics, and hence a brighter and/or harsher distortion characteristic.

- *PluggoFuzz* offers several different ways of distorting the signal, set by the Mode parameter:

  Soft Clip: the signal is boosted and clipped, but with a rounded-corner characteristic similar to an overdriven tube circuit.

  Hard Clip: the signal is boosted and clipped, leaving sharp corners. (The difference between the Soft Clip and Hard Clip modes is most evident at lower distortion levels. At high levels they're pretty much doing the same thing: unmercifully chopping off the peaks of your signals.)

  Foldover: the signal is boosted and clipped, and the clipped-off portion is folded downwards below the clipping level. (Imagine drawing the signal on paper, then folding the top quarter down, and holding the drawing up to the light so you can see the folded-over part.) The difference between this mode and the Hard Clip mode varies somewhat with the input signal, but often the Foldover mode has a bit more oomph.

  Wrap: the signal is boosted and clipped (bet you saw that coming), and the clipped signals are inverted and added back in. This simulates the distortion exhibited by the input stages of some less-well-designed analog-to-digital converters. Generally speaking, this kind of distortion does not produce nice sounds, but sometimes we're not in a mood for nice sounds, are we?
- To create a wah pedal effect: use the Bandpass or Peak/Notch mode for the output filter, and change its center frequency with a Modulator plug-in.

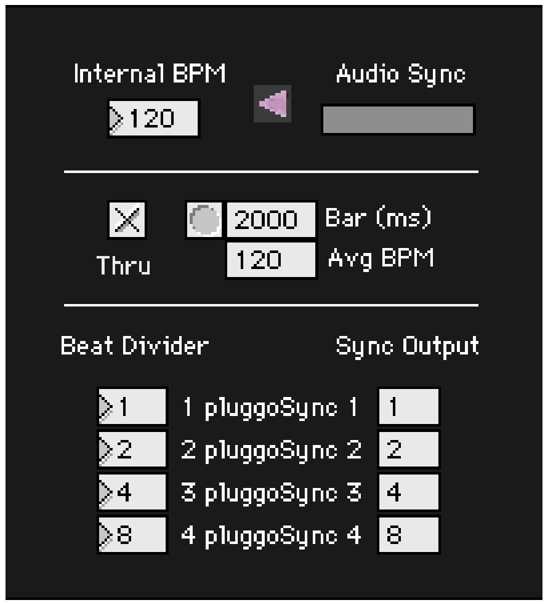
PluggoSync

category: synchronization

audio input: mono | *cpu: light* | *audio output: thru*

What It Does

PluggoSync accepts an audio signal and generates synchronization control signals. Other plug-ins can use this synchronization information to trigger events, such as restarting a step sequence or an LFO waveform. *PluggoSync* provides five different sync outputs, each with its own Beat Division, that run at multiples of the master sync signal generated from the audio input. *PluggoSync* can also run in an Internal Clock mode that generates the synchronization information whether or not an audio signal is present.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Clock Source	Int	Audio Sync		If this parameter, set by the purplish triangle near the top of the *PluggoSync* editing window, points to the Internal Clock, *PluggoSync* generates sync signals according to the Internal BPM parameter. If the triangle points to Audio Sync, *PluggoSync* will attempt to generate sync signals b y detecting peaks in its audio input. In either case, the clock generates a master tempo that is displayed in the Average BPM number box in the *PluggoSync* editing window.
Internal BPM	5	500	BPM	Sets the master tempo of the sync generator, used when *PluggoSync* when its Clock Source is the Internal Clock. If the Clock Source is Audio Sync, this parameter has no effect.

Name	*Min*	*Max*	*Units*	*Description*
Sync 1 Division (Beat Divider)	1	64		Sets the rate of the synchronization signal output for the sync source called *PluggoSync 1* relative to the master tempo. If the value of this parameter is 1, the sync signal for *PluggoSync 1* is generated at the same rate as the master tempo. If the value of this parameter is 2, the sync signal for *PluggoSync 1* is generated at twice the rate of the master tempo. If the value is 64, the sync signal for *PluggoSync 1* is generated at 64 times the master tempo.
Sync 2 Division (Beat Divider)	1	64		Sets the rate of the synchronization signal output for the sync source called *PluggoSync 2* relative to the master tempo. See the discussion for Sync 1 Division for more details.
Sync 3 Division (Beat Divider)	1	64		Sets the rate of the synchronization signal output for the sync source called *PluggoSync 3* relative to the master tempo. See the discussion for Sync 1 Division for more details.
Sync 4 Division (Beat Divider)	1	64		Sets the rate of the synchronization signal output for the sync source called *PluggoSync 4* relative to the master tempo. See the discussion for Sync 1 Division for more details.
Thru	Off	On		Enabling the Thru checkbox echoes the audio input to the audio output. This can be useful if you want to hear the click track input to *PluggoSync*, or to send the audio to another plug-in inserted after *PluggoSync*.

Interface Elements

- The number box to the left of the Bar (ms) displays the interval (in milliseconds) between sync pulses in the audio input (or internal clock) generated by *PluggoSync*.
- The round yellow button to the left of the Bar (ms) number box flashes when *PluggoSync* has generated a sync signal, resetting its beat count. Clicking this button resets the beat count and sends a sync signal.
- Beneath the Bar (ms) number box, pluggoSync displays the Average BPM (beats per minute) of the current master tempo, whether generated by the Internal Clock or from Audio Sync.
- To the right of each of the Beat Divider number boxes, the Sync Output number boxes show the beat count for each of the five sync outputs.

Insights

- Ideally, you'll want to use a special sync sample to get the best results with *PluggoSync*. The file *sync.aiff* supplied in the PluggoSync Examples in the Pluggo Stuff folder is a suitable trigger signal that's pleasant to listen to as well if you need to check whether the sync is working. You assemble a "sync track" containing the audio signals at the beat divisions you're interested in using a basis for effect synchronization. The *Pluggo Getting Started* manual has a chapter on the use of *PluggoSync*. You might also look at the PluggoSync Examples folder, where you'll find a example documents that employ *PluggoSync* acting to modify the playback of the *Synth* plug-in.

PlugLogic

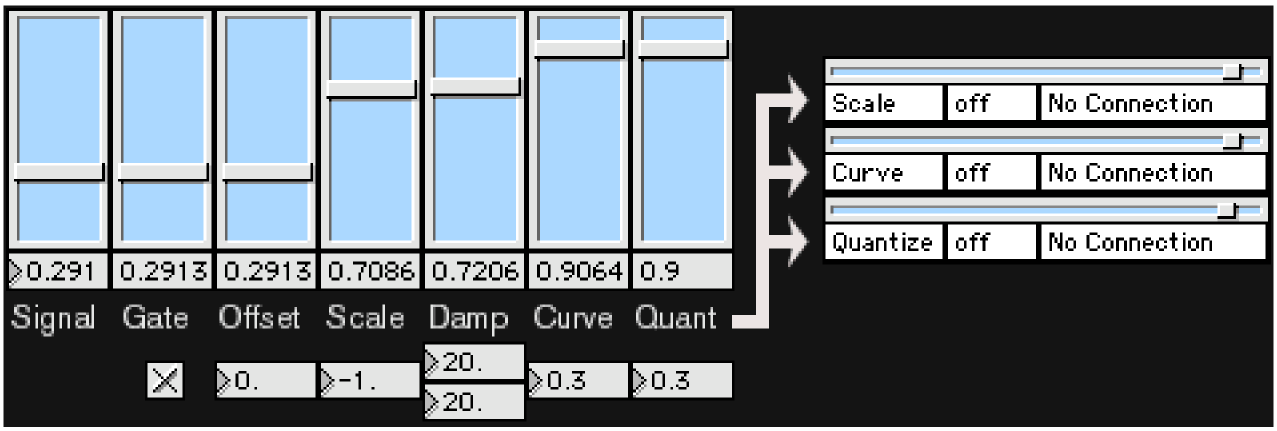
category: modulator

audio input: none *cpu: light* *audio output: thru*

What It Does

PlugLogic allows you to shape, massage, and multiply modulator control signals.

First, send a modulator signal to the *PlugLogic* Control Signal input. For example, choose PlugLogic 1 Control Signal in the modulator destinations menu of an LFO plug-in. You can also move the Control Signal slider manually to generate a signal. Next, you can apply up to six different modifiers to the Control Signal—Gate, Offset, Scale, Damp, Curve, and Quantize. At each modifier stage, the value of the Control Signal is displayed by a slider and a number box. You can not change these sliders directly - they just reflect the effects that the modifiers are having. Finally, you can send the Control Signal to up to three different destinations. Each destination can use any modifier as its source. This means, for example, you can send the scaled signal to one destination, the damped signal to another, and the quantized signal to a third.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Control Signal	0	1		This distinctively colored slider is the input to *PlugLogic*. In another Modulator plug-in, assign this Control Signal parameter of *PlugLogic* to one of the modulation outputs. The modulation signal will then be modified by *PlugLogic* according to the settings of the other parameters, and assigned to control up to three parameters of other plug-ins.
Gate	Off	On		Turns the Control Signal on and off. If the gate is open (On), the Control Signal will be passed through. Otherwise, if it is closed (Off), nothing will get through.
Offset	-1	1		This value is added to the Control Signal. A value of 0 will pass the Control Signal unchanged.
Scale	-5	5		This value is multiplied by the Control Signal. A value of 1 will pass the Control Signal unchanged. A value of -1 will reverse it.
Damp-up	0	100		Sets the rate of increase allowed in the Control Signal. Higher values make the Control Signal move more slowly in an upward direction. A value of 0 will pass the Control Signal unchanged.

Name	*Min*	*Max*	*Units*	*Description*
Damp-down	0	100		Sets the rate of decrease allowed in the Control Signal. Higher values make the Control Signal move more slowly in a downward direction. A value of 0 will pass the Control Signal unchanged.
Curve	0	5		Sets the slope of an exponential curve applied to the Control Signal. A value of 0 will pass the Control Signal unchanged.
Quantize	0	1		Limits the Control Signal values to multiples of this parameter. A value of 0 will pass the Control Signal unchanged.
Source 1-3	0	6		Selects a Control Signal modifier as the source for output to a parameter destination. Any modifier can be chosen - Gate, Offset, Scale, Damp, Curve, or Quantize.

Parameters for Each Modulation Destination

Name	*Min*	*Max*	*Units*	*Description*
Source	Signal	Quantize		Selects one of the Control Signal modifiers (Signal, Gate, Offset, Scale, Damp, Curve, Quantize) as the source for output to a modulation destination.
Mode	Off	Scale		When the mode is Set (the most common), the modulator's value directly sets the assigned parameter. When the mode is Offset, the modulator's value is added to the current value of the parameter (before modulation). When the mode is Scale, the modulator's value multiplies the current value of the parameter. When the mode is Off, no changes to the parameter occur.
Assign				Choose a plug-in and a parameter to modulate from this menu. The parameters listed will depend on the plug-ins that are currently inserted.

Insights

- Use *PlugLogic* to "mult" a control signal. For example, let's say you want an LFO plug-in to control five different plug-in parameters. It only has three modulator destinations —so assign two of its modulator destinations as desired, and then send the third to *PlugLogic*. From there you can assign three more modulator destinations—either straight through, or modified by one of the Control Signal modifiers.

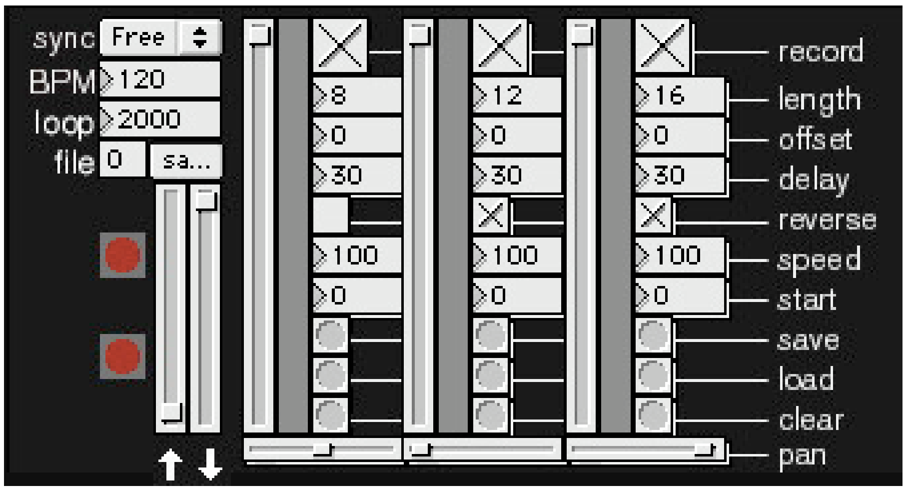
PlugLoop

accepts sync | *category: filter/delay*

audio input: mono, stereo | *cpu: medium* | *audio output: stereo*

What It Does

PlugLoop is an audio looping and sampling tool based on *Looper*, a Max/MSP program from jhno/Delicate Ear. It gives you three virtual tape loops, into which you can record fragments of audio, or continuous streams. The speed, direction, and retriggering of the loops are all user-controllable.



Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Sync Source	UDT	Plug		Selects the synchronization source for PlugLoop. The pop-up menu lets you choose from three sync modes: • Free mode lets you set the PlugLoop Tempo parameter. • Host mode synchronizes the Tempo to the host tempo. • Plug mode sets the Tempo to match the PluggoSync plug-in if it is loaded. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Tempo	5	500	BPM	Determines how fast the loops will be triggered, according to their Loop Length (below).
Max Loop	1	20000	ms	Sets the length of the virtual tape loops. This is limited by the amount of memory available.

Name	*Min*	*Max*	*Units*	*Description*
Load Set	0	99		*PlugLoops* lets you save and load the contents of the three virtual tape loops as a set. To use this feature, there must be a folder called *PlugLoops* in the VSTPlugIns folder (for VST and MAS applications) or the Pluggo Plug-ins folder (for Pro Tools). To save the loops to disk, choose a number with the File parameter, and select save from the popup menu. If, for example, you set File to 7 and select save, three files will be written into the *PlugLoops* folder: *plugloop.7.1*, *plugloop.7.2*, and *plugloop.7.3*. To load a loop set, choose a number with the File parameter, and select "load" from the popup menu. The corresponding files, if they exist, will be read into the loops. The File parameter is saved with your program, which allows you to restore the contents of your loops whenever you load the preset. If the File parameter is greater than 0, *PlugLoop* will try to load the corresponding plugloop files when the preset is loaded. If the File Parameter is 0, will not attempt to restore the contents of the loops.
Wet Level	0	1		Sets the overall output volume of the loop modules.
Dry Level	0	1		Sets the output level of the original input signal.

Parameters for Each Loop

Name	*Min*	*Max*	*Units*	*Description*
Loop Volume	0	127		Sets the output volume for this tape loop.
Loop Pan	0	127		Sets the pan position for this tape loop where 0 = left, 127 = right, and 64 = center.
Loop Record	Off	On		Starts/stops recording of the input audio signal into this tape loop.
Loop Length	1	74	beats	Sets the frequency at which this loop will be retriggered. A "beat" is defined as a sixteenth note in the current tempo. For example, if the Tempo is set to 120 BPM, a bar is 2000 milliseconds and a sixteenth note is 62.5 ms. If Loop Length is set to 4, then the loop will be retriggered every 4 sixteenth notes, which is once every 250 ms.
Loop Offset	-74	74	beats	Adjusts the timing of this loop in relation to the other loops. Think of this as a phase parameter—it offsets the loop triggers in single-beat increments.
Loop Delay	0	1000	ms	Sets the delay time for the audio output of this loop. This can be handy to fine-tune the timing between loops.
Loop Reverse	0	1		Sets the playback direction of this loop.
Loop Speed	1	1000	percent	Sets the playback speed of this loop. A value of 100 percent plays back the audio at its original pitch; a value of 1000 is ten times faster. A value of 50 causes half-speed playback.

PlugLoop

accepts sync *category: filter/delay*

audio input: mono, stereo *cpu: medium* *audio output: stereo*

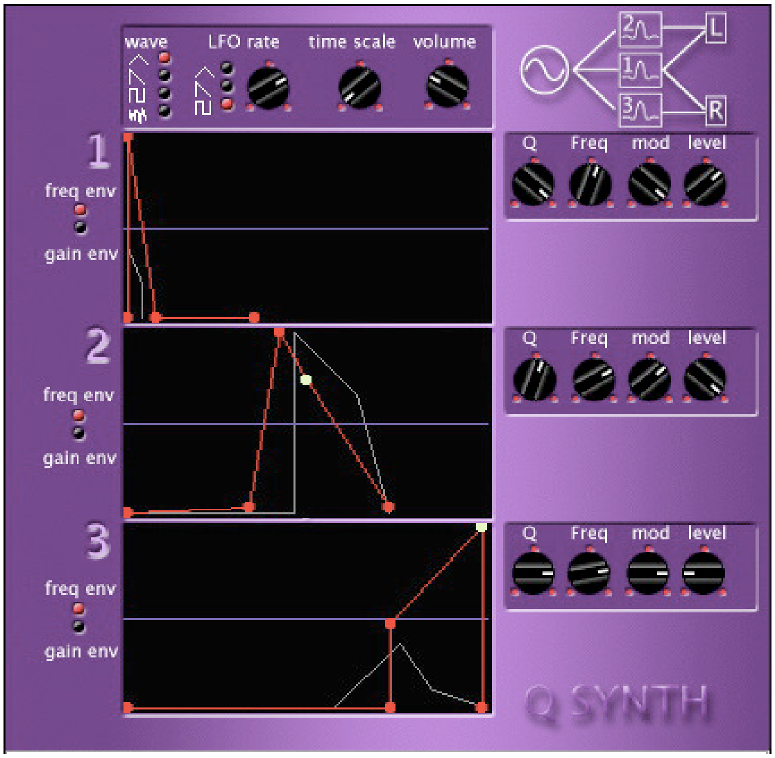
Name	*Min*	*Max*	*Units*	*Description*
Loop Start	0	2000	ms	Sets the position in the tape loop which playback will begin when the loop is triggered. This is useful to adjust the precise point in the loop where the sound starts. For example, you might adjust this parameter so that the loop starts right on a particular beat.

Insights

- Use Modulator plug-ins (especially *Randomizer*) to automate *PlugLoop* parameters to create algorithmic, evolving textures. For example, you can automate the Record checkbox to punch sounds in and out of the loop. The Length, Speed, and Reverse parameters are also prime candidates for automation.
- You can use the Load Set parameter to save looping sounds with your song. Let's say you have three loops going on with different settings, and you want this sound to be restored next time you open your song file. To do this, you need to save each loop by hitting the Save button in each channel. Name the files *plugloop.1.1*, *plugloop.1.2*, and *plugloop.1.3*, and make sure they go into a folder called *PlugLoops* that is inside the same VSTPlugIns folder (for VST or MAS applications) or the Pluggo Plug-ins folder (for Pro Tools). Then, once your sounds are saved, change the Load Set parameter to 1. *PlugLoop* will read the sounds you just saved into the loop channels. Save your song, and the next time you open it, all of the *PlugLoop* parameters will be restored—including the Load Set parameter, which will cause *PlugLoop* to read in those files.

What It Does

qsynth is a slick little synth that combines a single oscillator with three discrete resonant filters.



Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Waveform	0 (tri)	3 (random)	14	Sets the waveform of the single oscillator.
LFO Shape	0 (tri)	2 (square)	25	Sets the waveform of the filter modulation LFO.
LFO Rate	0 (0 Hz)	127 (20 Hz)	21	Sets the rate of the filter modulation LFO.
Envelope Time Scale	0 (0 sec)	127 (20 sec)	26	Determines the implied duration of the envelope editing display areas.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

qsynth

category: synthesis

input: MIDI | *cpu: medium/heavy* | *audio output: mono*

Name	*Min*	*Max*	*CC#*	*Description*
Editing Envelopes				The three editing envelopes provide five point editing of both cutoff frequency (freq env) and amplitude (gain env). The "white" editing point is the sustain position, the level that will be held until the MIDI key is released. The editing envelopes are not accessible via MIDI or Pluggo modifiers.
Envelope Q (Resonance)	0 (0.0)	127 (1.0)	15-17 (1-3)	Sets the resonance values (sometimes called Q or pre-emphasis) of the three filters. Resonance values range from 0.0 (no resonance) to 1.0 (self-oscillation).
Envelope Cutoff Frequency	0 (0 Hz)	127 (10 kHz)	18-20 (1-3)	Sets the cutoff frequencies of the three filters.
Envelope Cutoff Modulation	0 (0%)	127 (100%)	22-24 (1-3)	Determines the depth of LFO modulation of the cutoff frequency.
Envelope Output Level	0 (0%)	127 (100%)	27-29 (1-3)	Sets the three individual channel levels.

Global Parameters

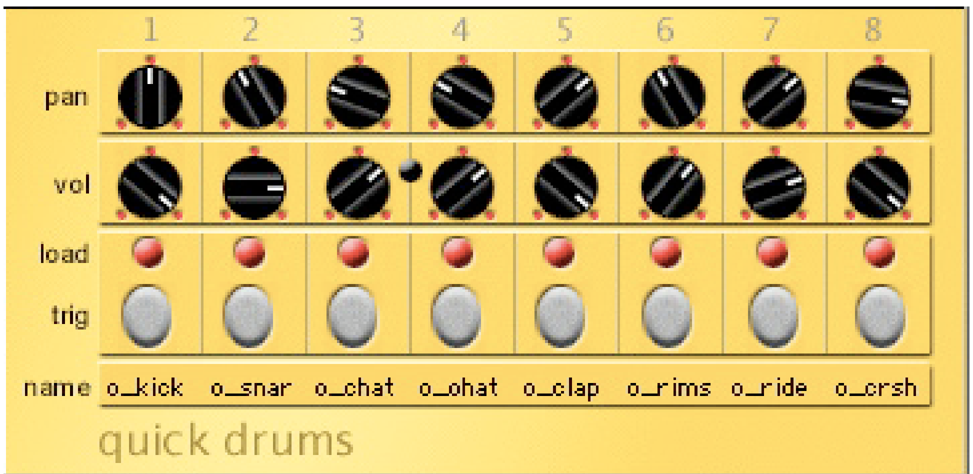
Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.
Polyphony	1	32	voices	Adjusts the number of available sample playback voices. Reducing this number may reduce CPU use.

Insights

- *qsynth* answers the question "What would I do if I had more filters than oscillators?" The diagram at the upper-right of the interface gives a quick overview of the oscillator-to-filters-to-output routing.
- The filter outputs are routed to different pan positions (see the diagram on the synthesizer interface), allowing for interesting stereo movements through filter and gain envelope handling. Almost all of the presets take advantage of this phenomenon.

What It Does

quick drums is a simple eight-channel sampled drum module, with pan and volume control on each channel. This module uses almost no CPU, and is a great way to inexpensively add some percussion sounds to the mix.



Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Pan Position	0 (Left)	127 (Right)	22-29 (1-8)	Adjusts the pan setting of each of the drum channels. A value of 64 (a vertical knob orientation) represents panning to the center.
Volume	0 (0%)	127 (100%)	14-21 (1-8)	Adjusts the individual channel levels output.
Channel 3/4 Exclusive Toggle	0 (off)	1 (on)	30	If engaged, the samples on channels 3 and 4 will "cut off" the decay of the other channel. This is useful for hi-hat samples (where open hats have to be damped by closed hat hits).
Load Button			N/A	Press the Load Button to choose a new sample for a given drum channel. An Open File dialog will be displayed, allowing a new sample to be selected.
Trigger Button			N/A	Click on the Trigger Button for a drum sample to audition that channel's settings.
Name Display			N/A	The name display will show the first six characters of each drum channel's sample.

Insights

- Quick and easy, with almost no CPU hit. That's the real insight on quick drums…

- The MIDI notes used to fire the drums sounds are:

  Channel 1: 36 (generally kick)
  Channel 2: 38 (generally snare)
  Channel 3: 42 (generally closed hi-hat)
  Channel 4: 46 (generally open hi-hat)
  Channel 5: 40 (generally 2nd snare or percussion)
  Channel 6: 41 (generally low tom)
  Channel 7: 43 (generally mid tom)
  Channel 8: 45 (generally high tom)

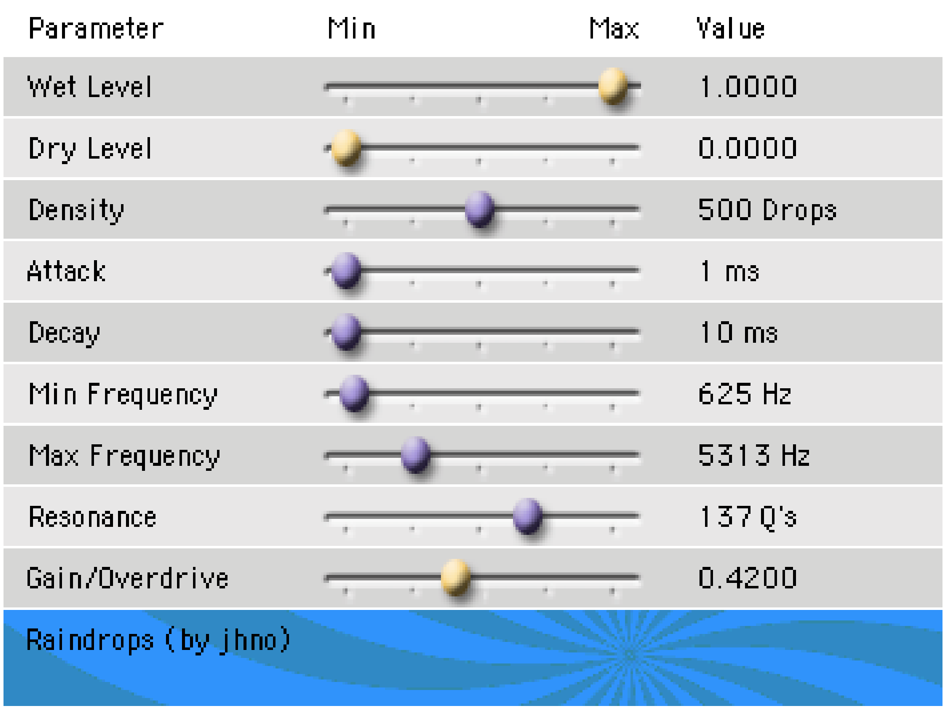
category: filter/delay

Raindrops

audio input: mono | *cpu: light* | *audio output: mono*

What It Does

"*Raindrops* was inspired by watching the rain fall late at night in front of the warehouse loft where I live in San Francisco. I wondered how you could turn a sound into a field of moving particles. So, when I went back inside I tried making a network of bandpass filters, each one offering a tiny peek into the frequency spectrum. The result is *Raindrops*." — jhno



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the output gain of the processed signal. In this case, the word "wet" is more descriptive than it is with the average signal processing algorithm.
Dry Level				Sets the output gain of the unprocessed signal.
Density	0	1000	drops	Adjusts the rate of the appearance of bandpass filter droplets.
Attack	0	1000	ms	Sets the onset time of the droplet envelope.
Decay	0	1000	ms	Sets the decay time of the droplet envelope.
Min Frequency	1	20000	Hz	Sets the low end of a range within which a random center frequency for each droplet will be generated.
Max Frequency	1	20000	Hz	Sets the upper end of a range within which a random center frequency for each droplet will be generated.

Raindrops

category: filter/delay

audio input: mono *cpu: light* *audio output: mono*

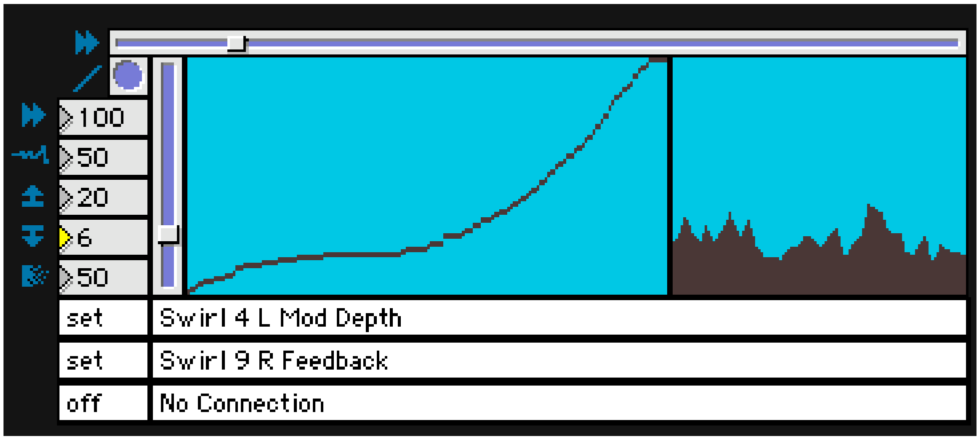
Name	*Min*	*Max*	*Units*	*Description*
Resonance	0	200	Q	Sets the global Q, or resonance, for all particles. Q is defined as bandwidth divided by center frequency.
Gain/Overdrive	0	1		Adjusts the overall volume, and distorts the particles if set too high.

Insights

- "By *network*, John is referring to *two* bandpass filters. He says it's an illusion of a network." —David Z.
- The *Exciter* preset uses distorted high-frequency particles. Distortion is achieved with a high Gain/Overdrive setting.

What It Does

Randomizer is a Modulator plug-in that generates random control signals that can be used to change the parameters of other plug-ins. It passes its audio input to its output without modification.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Rate	5	5000	ms	Sets the base interval between random numbers.
Rate Randomness	0	5000	ms	Sets a random value to be added to each base interval, for a more random timing. For example, with a Rate of 100 ms and a Rate Randomness of 200 ms, the time between generated values will vary randomly between 100 and 300 ms.
Damp Up	0	1000	ms	Sets the time required for the output to reach a new value, if the new value is higher than the old one. A value of 0 indicates no damping.
Damp Down	0	1000	ms	Sets the time required for the output to reach a new value, if the new value is lower than the old one. A value of 0 indicates no damping.
Output Grain	0	1000	ms	Sets the output rate when damping is active.

Random Distribution

Randomizer allows you to draw the distribution function for the numbers that will be randomly generated. The parameters listed above essentially affect the output and modification of random numbers after they have been generated. A distribution function is something that indicates, over the long term, how the numbers generated will be distributed over the range of possible values. A straight line (which you can get by clicking the Straight Distribution button) should generate a range of values between 0 and 1 equally often. The histogram display shows the values that have been generated recently. The slider to the immediate left of the distribution function indicates the most recent value generated.

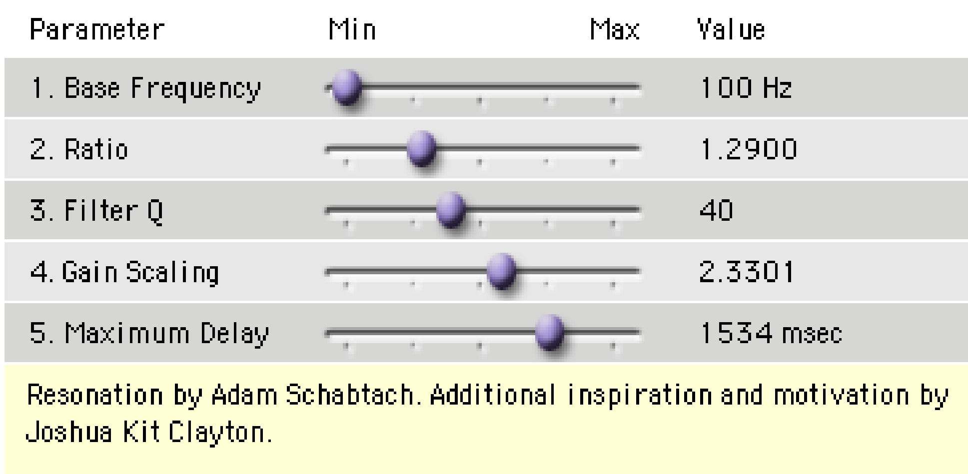
Resonation

category: filter/delay

audio input: mono | *cpu: medium* | *audio output: stereo*

What It Does

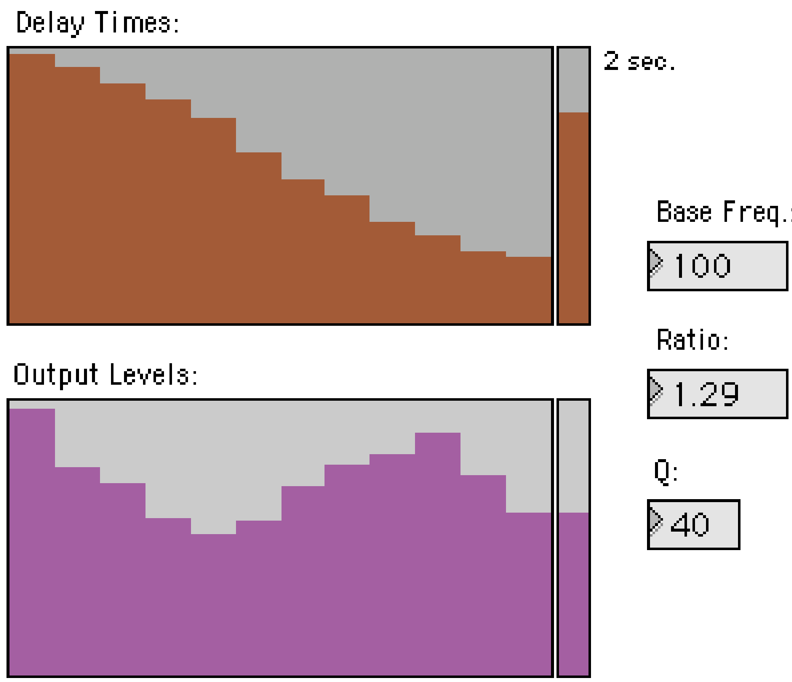
Resonation sends its input signal to 12 bandpass filters (resonators) in parallel. The filters are tuned in semitone intervals. Each filter is followed by a delay element, with a delay time of up to two seconds. You can adjust the frequency of the lowest filter and the ratio between frequencies of adjacent filters, from which the frequencies of the other filters are derived., the time of the longest delay, and the spread of delay times relative to the longest.



Visible Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Base Frequency	20	10000	Hz	Center frequency of the lowest-pitched filter.
Ratio	1	2		Ratio between successive filter frequencies. If this parameter is 1.0, all filters have the same frequency. If it is 2.0, the filters are tuned in octaves.
Filter Q	1	100		Sets the resonance or Q of the filters. Q is defined as the filter’s center frequency divided by its bandwidth.
Gain Scaling	0	4		Boosts or attenuates the output levels of the filters. Turn this up if the output signal is weak; turn it down if you hear clipping.
Maximum Delay	0	2000	ms	Sets the longest delay time available for the filter delays.

Interface View Parameters



Name	*Min*	*Max*	*Units*	*Description*
Delay Times (multislider)	0	100	%	Sets the delay times of the filter delays. Each slider shows the corresponding filter's delay time as a percentage of the maximum delay time, set by parameter 5 (above).
Maximum Delay Time (slider)	0	2000	ms	Same as parameter 5, above.
Output Levels (multislider)	0	100	%	Sets the output levels of the filters. These levels are scaled by the gain scaling, set by parameter 4 (above).
Gain Scaling (slider)	0	4		Same as parameter 4, above.
Base Freq.	20	10000	Hz	Same as parameter 1, above.
Ratio	1	2		Same as parameter 2, above.
Q	1	100		Same as parameter 3, above.

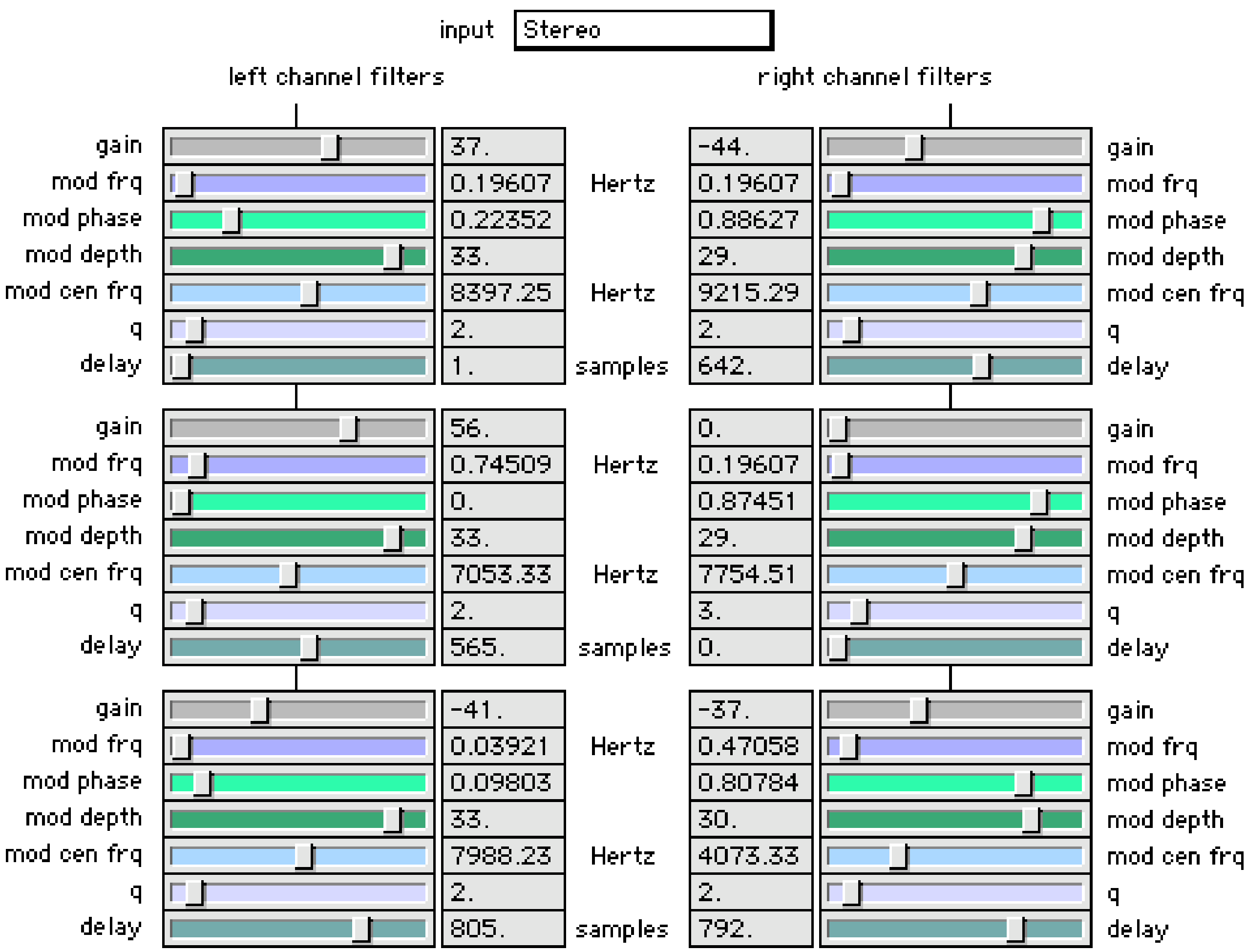
Insights

- The *Up* preset creates rising glissandos by tuning the filters in semitones, and varying the delay times linearly from zero for the first filter to maximum for the last. Hence, you hear the output of the lowest-pitched filter first, followed by the next-lowest filter, on up to the highest filter. The *Down* preset works in a similar fashion, only in the other direction.

- Varying the Base Frequency and Ratio parameters, either by hand or with a Modulator plug-in, can create some lovely sweeping and swirling effects.
- The outputs from the filters are sent alternately to the left and right output channels of the plug-in. This creates a synthetic stereo effect. If this effect isn't useful in your musical context, use the plug-in in a mono-in/mono-out context.

What It Does

Resosweep provides you with six bandpass filters, each of which can be modulated independently. In addition, each filter has a short delay line that can be used for subtle stereo positioning effects. By arranging the modulations so they're out of sync with each other, you can create complex, evolving timbres.



Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Input Select	Mono (Left)	Mono (Right)		Selects whether the input to the filters comes from the left channel only, both channels (the left to the left three filter, and the right to the right three filters), or from the right channel only.

Resosweep

category: filter/delay

audio input: mono, stereo | *cpu: medium* | *audio output: stereo*

Parameters for Each Filter

Name	*Min*	*Max*	*Units*	*Description*
Gain	-128	127		Sets the input gain to the filter. Negative values invert the phase of the signal.
Mod Freq	0	10	Hz	Sets the rate of modulation of the filter's center frequency.
Mod Phase	0	1		Allows you to offset the modulation waveform relative to others at the same frequency.
Mod Depth	0	36	ms	Sets the amount of modulation of the filter's center frequency.
Mod Center Freq	100	15000	Hz	Sets the base center frequency of the filter before modulation.
Q	1	30		Sets the sharpness of the filter. A value of 1 is the least sharp, and a value of 30 is the sharpest. Q is defined as bandwidth divided by center frequency.
Delay	0	1024	samples	Sets the delay of the filter's output. Small delays can have the effect of "positioning" the filters in a stereo image.

Insights

- If the center frequencies of several filters are close together, you may not hear one or more of them, due to masking effects. This can be used to interesting effect when the filters are modulated however, since as the filter center frequencies begin to separate, you will suddenly hear a filtered sound as it passes out of the "auditory masking filter" region of another filtered sound. Another reason you may not be able to hear one of the filters is that the input signal does not contain significant spectral information in the region you have chosen. In this case it will be necessary to turn up the gain. Finally, note that the output level of a filter decreases as you increase its Q
- *Kneeling at the Wall* is particularly useful preset for emulating a certain kind of processing on cymbals some think of as the "jhno sound."
- "Yeah, there are a lot of sliders. But just start moving them."

category: distortion

Ring Modulator

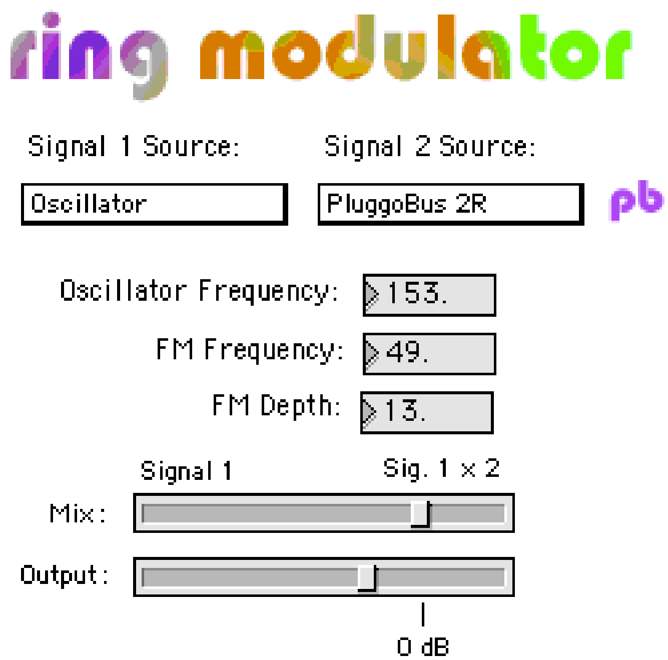
audio input: mono, pluggoBus *cpu: light* *audio output: mono*

What It Does

Ring Modulator multiplies two signals together. The resulting signal contains only frequency components equal to the sum(s) and difference(s) of the frequencies present in the input signals.

For example, if both input signals are sine waves, of frequencies 300 and 500 Hz respectively, the output signal will consist of two sine waves, one with a frequency of 200 Hz and one with a frequency of 800 Hz. If the input signals are more complex than sine waves, the output will be correspondingly more complex, and often clangorous or inharmonic.

Typical hardware ring modulators provide an internal oscillator as signal, and hence have only one input. In *Ring Modulator*, the two signals may be any of the following: either input channel for this plug-in, an internal sine-wave oscillator, or any of the PluggoBus signals.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Signal 1 Source	Off	Pluggo Bus 4R		Selects an input signal. The signals designated “left” and “right” are the inputs to this plug-in. They will be the same if the plug-in is used in a mono context.
Signal 2 Source	Off	Pluggo Bus 4R		Selects an input signal. The signals designated “left” and “right” are the inputs to this plug-in. They will be the same if the plug-in is used in a mono context.
Oscillator Frequency	0	5000	Hz	Sets the frequency of the internal sine-wave oscillator.
FM Frequency	0	100	Hz	Sets the frequency of a second oscillator which modulates the frequency of the first oscillator.

Ring Modulator

category: distortion

audio input: mono, pluggoBus *cpu: light* *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
FM Depth	0	100	Hz	Sets the amount that the second oscillator changes the frequency of the first oscillator.
Mix	0	100	%	Blends the output of the ring modulator with the Signal 1 input. Setting this slider to the far left sends only Signal 1 to the output. Setting it to the far right sends only the output of the ring modulator.
Output	-70	18	dB	Sets the overall output level of the effect.

Insights

- Ring modulation is a form of amplitude modulation (AM). The distinction between ring modulation and AM is that in AM, one input signal (designated the *modulator*) has a DC component. This means that the output signal will contain some amount of the input signal, in addition to the sum and difference frequencies. *Ring Modulator* makes no attempt to block DC offsets in either of its inputs, and hence will operate as an amplitude modulator. If you're using the internal oscillator and hear its tone in the output, it means that the input signal has a DC offset (assuming that you've set the Mix slider to the far right).
- Setting the oscillator to a sub-audio frequency creates tremolo effects. (Tremolo is another special name for amplitude modulation.)
- Ring modulation often creates sounds that are described as “clangorous” or “metallic.” Using two relatively simple signals that rise and fall in pitch independently can create sounds often associated with shortwave radios or invasions of alien beings which took place during the 1950s.
- Ring modulators are so named because the first such devices contained four diodes connected together in a ring-like configuration. That electrical engineers named the device a “ring modulator” because its internals contained four small electronic components laid out in a square may suggest something to you about the communication skills of electrical engineers.

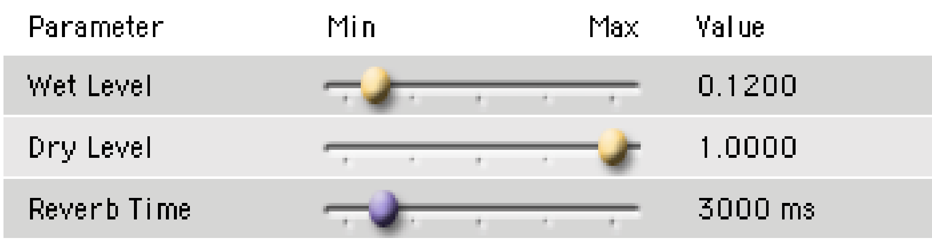
category: reverb

Rough Reverb

audio input: stereo *cpu: medium* *audio output: stereo*

What It Does

Rough Reverb is a simple effect that can be especially useful when a super-smooth, lush sound is simply not what you are looking for. It is fully stereo (both channels are independent).



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the level of the reverberated input signal.
Dry Level	0	1		Sets the level of the direct input signal.
Reverb Time	1	20000	ms	Sets the delay time inside the reverberator.

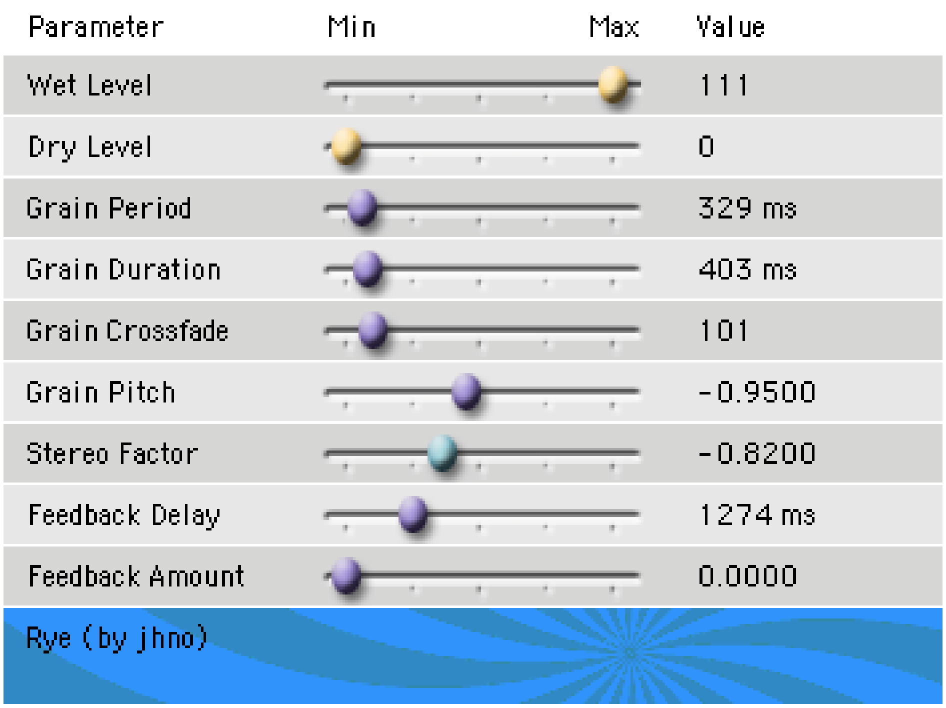
Rye

category: granular

audio input: mono, stereo | *cpu: medium* | *audio output: stereo*

What It Does

Rye is part of the Pluggo Signature Series Granular Synthesizers. *Rye* offers a basic palette of controls for real-time granular synthesis, including a unique stereo effect that uses phase reversal.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	127		Sets the output level of the granular synthesis.
Dry Level	0	127		Sets the output level of the original input signal.
Grain Period	5	5000	ms	Sets the time interval between grains—i.e., the speed of the granular synthesis.
Grain Duration	1	5000	ms	Sets the length of each grain.
Grain Crossfade	0	1000	ms	Sets the crossfade time between successive grains.
Grain Pitch	-10	10		Sets the playback speed of the grains. A value of 1 causes playback at original speed; -1 causes playback at original speed, backward. A value of 2 is double speed; -0.5 is half speed backward.
Stereo Factor	-3	3		Adjusts the stereo image of the granular synthesis. A value of 1 causes mono output. Values greater than 1 give stereo separation, while values less than 1 give phase-reversed stereo separation.

audio input: mono, stereo *cpu: medium* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Feedback Delay	0	5000	ms	Sets the delay time for feedback of the granular synthesis output back into its input.
Feedback Amount	0	7		Sets the gain of the feedback signal.

Insights

- Pumpernickel?

shape synth

category: synthesis

input: MIDI *cpu: medium/heavy* *audio output: mono*

What It Does

shape synth is a killer synthesizer featuring a waveshaper for extensive real-time sound manipulation. Dual oscillators feed a waveshaper, followed by a resonant lowpass filter. Mod Wheel control of the waveshape makes this an unsurpassed real-time sound machine.

Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Waveshape	0 (square)	127 (pulse)	1	Sets the shape used for the waveshaped voicing system. This setting can be controlled in real-time using the Mod Wheel (CC# 1).
Waveshape Noise Mix-in	0 (0%)	127 (100%)	31	Mixes in some white noise into the waveshape. This will create a grittier or more metallic sound.
Waveshape Phase Adjustment	0 (0 degrees)	127 (180 degrees)	32	Sets the phase (waveform offset) into the waveshape. This provides a wider array of shapes for audio processing.
Oscillator 2 Detune	0 (-12 semi)	127 (+12 semi)	14	Sets the coarse pitch offset of the second oscillator.
Oscillator 2 Fine Tune	0 (-1 semi)	127 (+1 semi)	15	Sets the fine tuning of the second oscillator.
LFO Shape	0 (tri)	2 (square)	26	Sets the waveform of the LFO.

Name	*Min*	*Max*	*CC#*	*Description*
LFO Speed (Rate)	0 (0 Hz)	127 (20 Hz)	27	Sets the rate of the LFO.
LFO Waveshape Phase Depth	0 (0%)	127 (100%)	33	Sets the amount of LFO modulation applied to the waveshape phase.
Filter Cutoff	0 (0 Hz)	127 (10 kHz)	16	Sets the cutoff frequency of the filter.
Filter Resonance	0 (0.0)	127 (1.0)	17	Sets the resonance value (sometimes called Q or pre-emphasis) of the filter. Resonance values range from 0.0 (no resonance) to 1.0 (self-oscillation).
Filter Envelope Depth	0 (0%)	127 (100%)	28	Sets the amount that the Filter Envelope modulates the filter cutoff.
Filter LFO Depth	0 (0%)	127 (100%)	29	Sets the amount that the LFO modulates the filter cutoff.
Amplitude Envelope Attack	0 (0 ms)	127 (5000 ms)	18	Sets the amplitude envelope attack rate. When a note is played on the synthesizer, the amplitude envelope is triggered, and the attack time determines how long it takes the amplitude to move from zero to its maximum value.
Amplitude Envelope Decay	0 (0 ms)	127 (4000 ms)	19	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the amplitude to move to its sustain level.
Amplitude Envelope Sustain	0 (0%)	127 (100%)	20	Sets the amplitude envelope sustain level. The amplitude will remain at this level until the note is released.
Amplitude Envelope Release	0 (0 ms)	127 (6000 ms)	21	Sets the amplitude envelope release rate. This is the amount of time it takes for the amplitude to move back to zero after a note has been released.
Filter Envelope Attack	0 (0 ms)	127 (5000 ms)	22	Sets the filter envelope attack rate. When a note is played on the synthesizer, the filter envelope is triggered, and the attack time determines how long it takes the modulation output to move from zero to its maximum value.
Filter Envelope Decay	0 (0 ms)	127 (4000 ms)	23	Sets the filter envelope decay rate. After the envelope has reached its maximum value (based on the Attack parameter), the decay time determines how long it takes for the modulation output to move to its sustain level.
Filter Envelope Sustain	0 (0%)	127 (100%)	24	Sets the filter envelope sustain level. The modulation output will remain at this level until the note is released.

shape synth

category: synthesis

input: MIDI *cpu: medium/heavy* *audio output: mono*

Name	*Min*	*Max*	*CC#*	*Description*
Filter Envelope Release	0 (0 ms)	127 (6000 ms)	25	Sets the filter envelope release rate. This is the amount of time it takes for the modulation output to move to zero after the note has been released.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.

Insights

- If you aren't using the mod wheel, you are missing half the fun of this synthesizer. Mod wheel control will adjust from near-pulse to rounded sawtooth waveforms.
- The phase adjustment provides a wide array of oddball waveforms for waveshaping. Presets *Dr. Detuna*, *Raft-bound* and *Man from K.L.A.M.P.* all show off some interesting waveforms

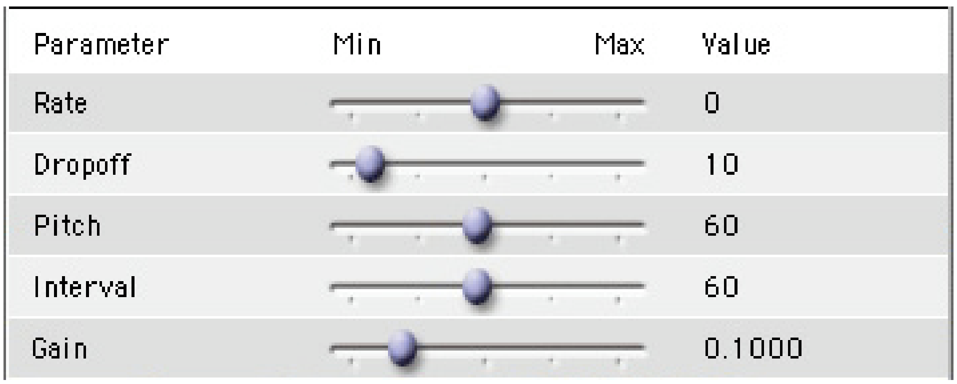
category: synthesis

ShepardTones

audio input: none *cpu: light* *audio output: mono*

What It Does

ShepardTones is a plug-in that produces its namesake as output – a tone whose timbral balance and rate of pitch change produces the illusion of a continually rising (or falling) tone. Use it to get a rise on the dancefloor, or to construct large, mournful slabs of ascending/descending textures.



Visible Parameters

Name	*Min*	*Max*	*Description*
Rate	-20	20	Sets the rate at which the tone appears to rise or fall and the “direction” of the tone. Positive values produce a rising tone, and negative values produce a falling tone
Dropoff	0	127	Sets the dropoff rate for the oscillator mix. Adjusting the mix will produce a more convincing illusion when the Rate and Interval values are changed.
Pitch	0	127	Sets the pitch of the illusory tone. The pitch is set using MIDI note numbers (60 = Middle C).
Interval	0	127	Used to adjust the interval between the two tones whose crossfading is used to produce the Shepard Tone illusion. The interval is set using MIDI note numbers (60 = Middle C).
Gain	0.0	5.0	Sets the overall gain of the output.

Insights

- You can create interesting textures by using multiple ShepardTones patches set to pitches which constitute the notes of a chord and whose rates are set to different values.

Shuffler

accepts sync | *category: granular*

audio input: mono | *cpu: medium* | *audio output: mono*

What It Does

Shuffler can re-arrange its audio input signal with astonishing dexterity. It works by recording incoming audio into a loop, and then playing back slices of that loop in real-time. For example, if you set the Loop Time to 1000 ms and the Number of Slices to 4, then each second you will hear the last second of audio cut up into 250 ms slices. Using the slider interface, you can specify the position of each slice within the playback sequence, as well as the slice's playback speed. Once you get the hang of it, you can do things like reversing single drum beats in a pattern, stutter effects, and just plain random mangling.

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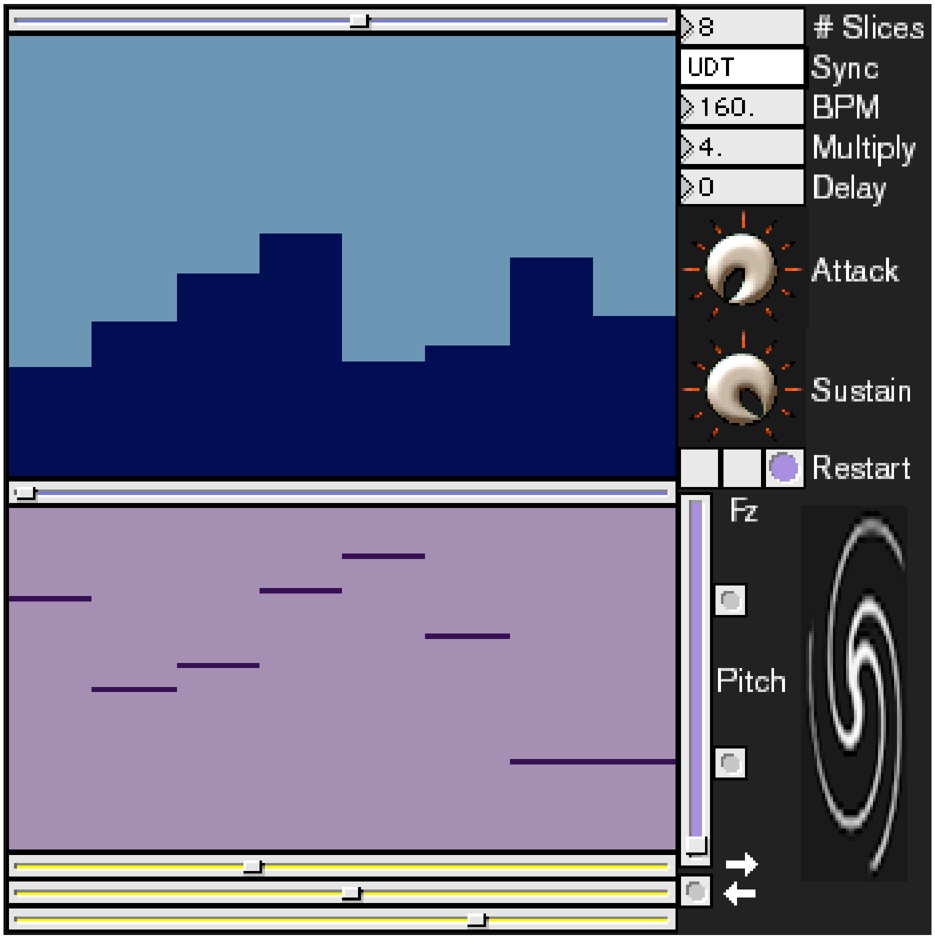


Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the output level of the shuffled audio.
Dry Level	0	1		Sets the output level of the original input signal.
Steps	1	16		Sets the number of slices to cut the loop into.

Name	*Min*	*Max*	*Units*	*Description*
Sync Source	UDT	Plug		Selects the synchronization source for Shuffler. The pop-up menu lets you choose from three sync modes: • UDT (User-Defined Tempo) mode lets you select the Tempo with the Tempo slider. • Host mode synchronizes the Tempo with the host tempo. • Plug mode sets the Tempo from the PluggoSync plug-in if it is loaded. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Tempo	1	300	BPM	Sets the tempo of Shuffler in beats per minute. This parameter, along with the Multiply parameter below, determines Shuffler's speed.
Multiply	0.01	8.0		Scales Shuffler's speed. With Multiply set to 1, Shuffler will cycle once per bar at the current Tempo. If you changed this to 2, Shuffler would cycle once every two bars, which is twice as slow, while a setting of 0.25 would be four times as fast. Note that Shuffler assumes a 4/4 time signature. You can compensate for other time signatures with judicious settings of Multiply, if necessary.
Mode	0	1		This changes the way that the Attack and Sustain parameters affect playback of the slices. For basic playback, be sure this is turned off.
Attack	0	1		Sets the time required to reach the current slice position. For basic playback, be sure this is set to 0.
Sustain	0	1		Sets the slice position target to reach after the Attack segment. For basic playback, be sure this is set to 1.
Freeze	0	1		Stops recording new audio into the loop. Whatever sound is currently recorded will remain until Freeze is turned off and recording of incoming audio resumes.
Trigger	0	1		Restarts the step sequencers, triggering a new slice cycle.
Delay	0	500	ms	Sets the delay time for the audio output. This can be useful to keep it in sync with other sounds.
Stereo	-2	2		Adjusts the output stereo image. A value of 1 causes mono output. Values greater than 1 provide stereo separation, while values less than 1 provide phase-reversed stereo separation.

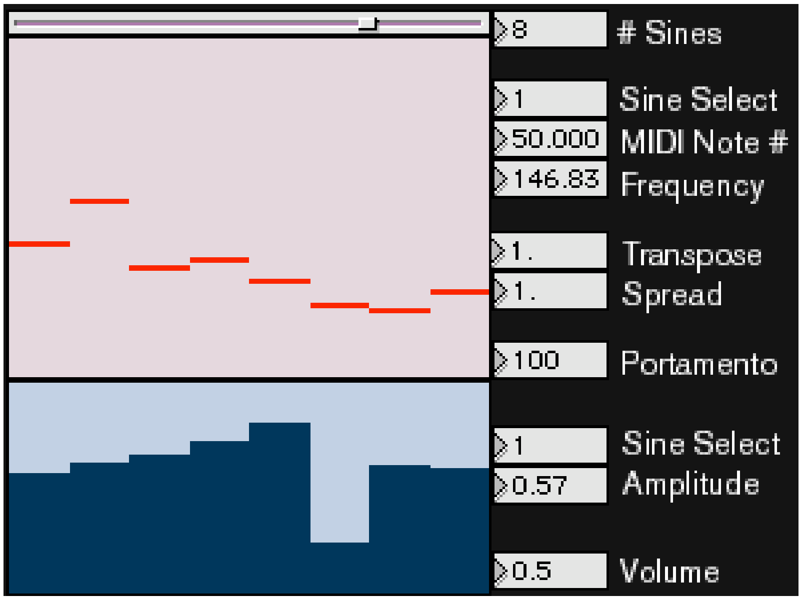
Sine Bank

category: synthesis

audio input: none *cpu: medium* *audio output: mono*

What It Does

Sine Bank is a synthesizer that plays up to 32 sine waves. Global parameters such as Transpose and Spread make it an ideal animal to control from a Modulator plug-in. You might use *Sine Bank* as an input to the *Convolver* or *Vocoder*, or simply on its own as a background texture. You can use the upper set of sliders to set rough frequencies for the entire collection of sine waves, then use the MIDI Note number and Frequency controls to fine tune each oscillator. The lower set of sliders sets an amplitude value for each oscillator. This value can be fine-tuned with the Amplitude control.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Output Volume	0	1		Sets the overall output volume of the sine waves. When using more sine waves, it will be necessary to reduce the overall volume to avoid clipping.
# of Sines	1	32		Sets the number of sine waves that are present in the output.
Frequency Sine Select	1	32		Selects a sine wave to be changed by the Frequency and MIDI Note Number controls.
Frequency Fine Tune	8.1757	12543	Hz	Sets the oscillator frequency for the selected sine wave.
Frequency Transpose	-5	5		Sets a global transposition factor on the frequencies of all the sine waves. A value of 1 means no transposition.
Frequency Spread	0	5		Sets the spacing of the sine waves. When this parameter is set to 1, the frequencies are left unchanged. When set to 0, all frequencies are set to the average frequency value As the value of the parameter increases, the frequencies are spread out increasingly from the average value.

Sine Bank

audio input: none *cpu: medium* *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Portamento	0	5000	ms	Sets the rate at which frequencies change from their current value to a new value; for instance, when changing programs or drawing with the multislider. A value of 0 means that the oscillators change frequency immediately.
Amplitude Sine Select	1	32		Selects a sine wave to be changed by the Amplitude control.
Amplitude Fine Tune	0	1		Sets the amplitude of the selected sine wave.

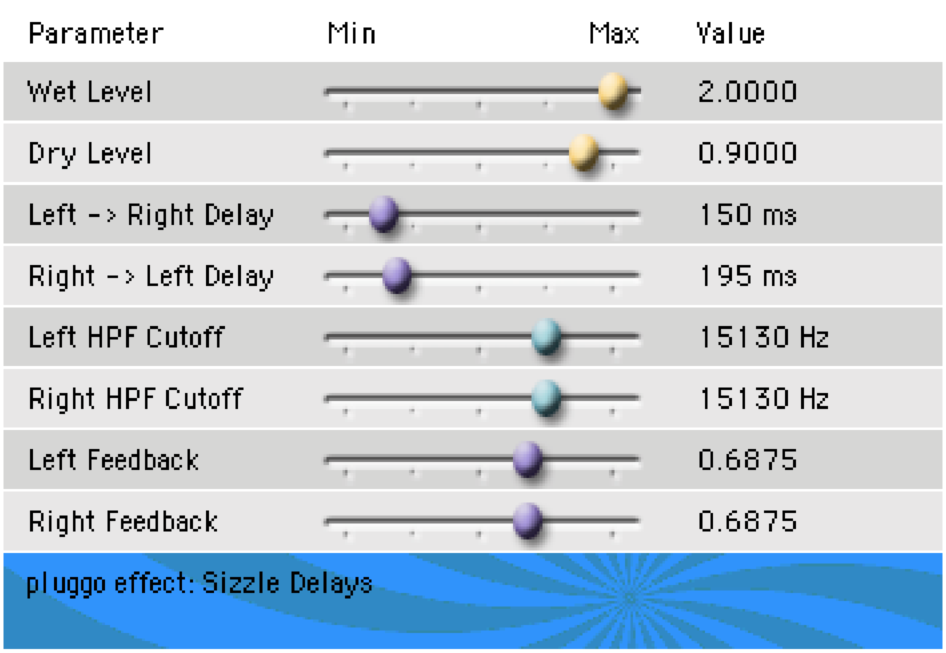
Sizzle Delays

category: filter/delay

audio input: stereo | *cpu: light* | *audio output: stereo*

What It Does

Sizzle Delays creates a broad range of interesting effects by a simple method. The input signal is fed into a stereo delay line. Two high-pass filters are applied to the delay outputs, which are then fed back into the delay lines. Exciter effects can be achieved with short delays and high feedback levels that are clipped within the feedback loop.



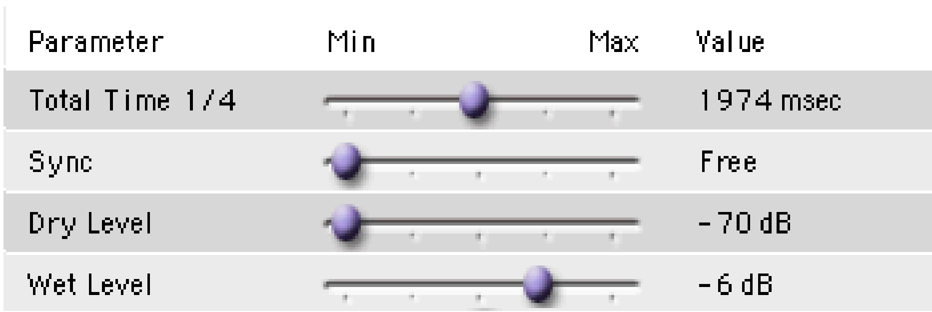
Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	2		Sets the output gain of the processed signal.
Dry Level	0	1		Sets the output gain of the unprocessed input signal.
Left -> Right Delay	1	1000	ms	Sets the delay time on the left channel that is fed back to the right channel.
Right -> Left Delay	1	1000	ms	Sets the delay time on the right channel that is fed back to the left channel.
Left HPF Cutoff	20	20000	Hz	Determines the cutoff frequency of a high-pass filter that is applied within the feedback loop of the left delay line.
Right HPF Cutoff	20	20000	Hz	Determines the cutoff frequency of a high-pass filter that is applied within the feedback loop of the right delay line.
Left Feedback	0	1		Sets the gain on the output of the filter that is fed back from the left delay line to the right channel.
Right Feedback	0	1		Sets the gain on the output of the filter that is fed back from the right delay line to the left channel.

category: granular	*accepts sync*	**Slice-n-Dice**
audio input: mono	*cpu: light*	*audio output: mono*

What It Does

Slice-n-Dice continuously records its input signal and slices it into 32 pieces of equal length, while at the same time playing back one or more of the previously recorded 32 slices. You use a multislider to determine which slices are played as well as the order in which they're played back.



Visible Parameters

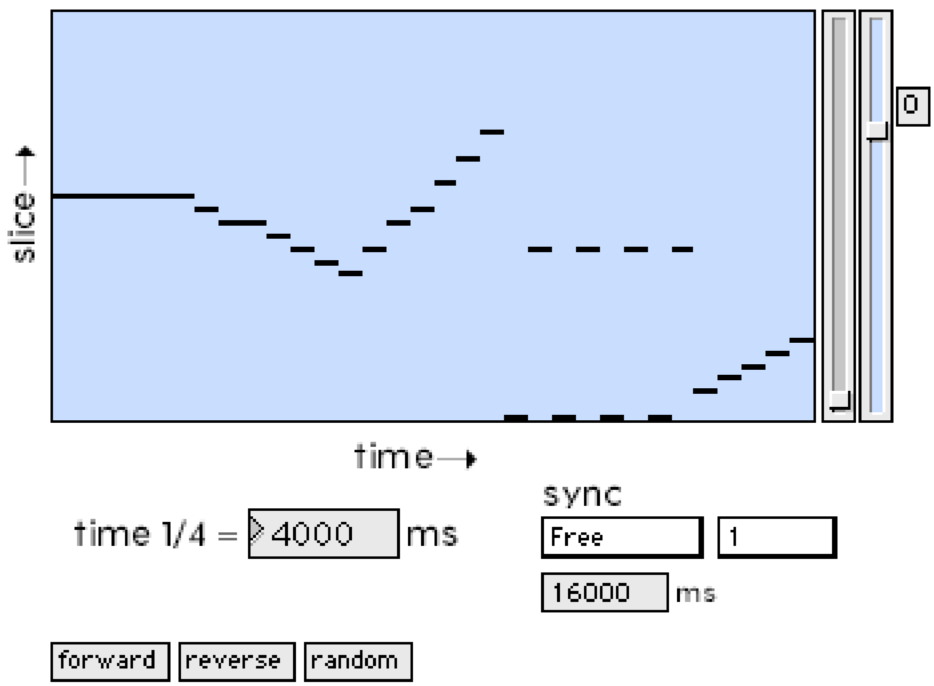
Name	*Min*	*Max*	*Units*	*Description*
Total Time 1/4	32	4000	ms	Sets the total delay time for all the slices in Free mode. This value is further modified by the Note Value parameter, and the Total Time 1/4 parameter is taken to be the time for a quarter note in the Note Value parameter. The resulting time for all the slices is displayed beneath the Sync pop-up menu. In the screen shown above it is 16000ms because the Total Time 1/4 is multiplied by 4 due to the Note Value parameter being a whole note (1).
Sync	Internal	Pluggo Sync		Sets the synchronization mode. • In Free mode, the Total Time 1/4 parameter sets the total delay time for the slices. • In Host mode, the host tempo and the Note Value parameter determine the interval of a slice. • In Plug mode, the output of the PluggoSync plug-in determines the total delay time for all the slices. In all modes, the current total delay time is displayed beneath the Sync pop-up menu. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Dry Level	0	1		Sets the level of the unprocessed input signal.
Wet Level	0	1		Sets the overall output level of the effect.

Slice-n-Dice

accepts sync | *category: granular*

audio input: mono | *cpu: light* | *audio output: mono*

Slicer View



Slicer View Parameters

Name	*Min*	*Max*	*Units*	*Description*
Multislider				Determines which slices are played, and in what order. Each slider's vertical position indicates which slice will be played. The lowest vertical position selects the first of the 32 slices; the highest position selects the last.
Dry Level (gray slider)	0	1		Sets the level of the unprocessed input signal.
Wet Level (blue slider)	0	1		Sets the overall output level of the effect.

Name	*Min*	*Max*	*Units*	*Description*
Sync Mode				Selects the synchronization source for Shuffler. The pop-up menu lets you choose from three sync modes: • In Free mode, the Total Time 1/4 parameter sets the total delay time for all the slices. • In Host mode, the host tempo and the Note Value parameter determine the total delay time for all the slices. • In Plug mode, the output of the PluggoSync plug-in determines the total delay time for all the slices. In all modes, the current total delay time is displayed beneath the Sync pop-up menu. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Note Value	1	1/64t		Sets the time of the slices by multiplying or dividing the current total delay time (set either by the Total Time 1/4 parameter or by the host tempo).

Interface Elements

- The **forward** button sets the slices to play back in order. The **reverse** button sets the slices to play in reverse order. The **random** button randomizes the slices. You can click this button repeatedly to try different randomly generated combinations.

Insights

- Setting the sliders so that they form a diagonal line from the lower left to the upper right will produce an output signal that is the same as the input. The *Forward* preset sets the sliders in this manner.
- Setting the sliders so that they form a diagonal line from the upper left to the lower right—as seen in the *Backward* preset—will produce an output signal that sounds like the input signal played backwards, sort of.
- *Slice-n-Dice* was created partly in anticipation of a future era in which holographic storage modules will have replaced CDs, and the sound of a skipping CD player will be a much sought-after “vintage” effect.

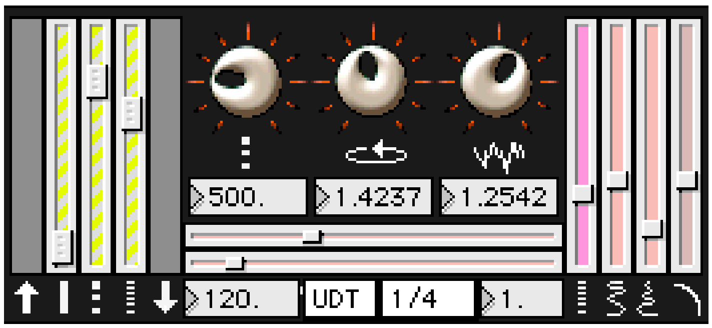
Space Echo

accepts sync | *category: delay*

audio input: mono | *cpu: medium* | *audio output: mono*

What It Does

Space Echo was inspired by the classic Roland RE-201 tape delay/reverb unit. But it has a sound all its own. It features tape speed effects (warble and inertia) that can be made to sound quite a bit more extreme than a correctly functioning tape delay unit. As for a comparison with a malfunctioning tape delay, that's a matter of speculation.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Dry Mix	0	157		Sets the output level of the original input signal.
Echo Mix	0	157		Sets the output level of the delay signal.
Reverb Mix	0	157		Sets the output level of the reverb signal.
Tape Speed	0	1		Sets the delay time by varying the virtual tape speed.
Feedback	0	1		Sets the amount of feedback in the tape delay.
Clip	0	1		Adjusts the amount of clipping that takes place in the tape delay feedback. This is analogous to the saturation that occurs with analog tape.
Lowpass	0	127		Sets the cutoff frequency of the output signal low-pass filter. This is effective in simulating the limited frequency response of tape delay.
Highpass	0	127		Sets the cutoff frequency of the output signal high-pass filter. This is effective in simulating the limited frequency response of tape delay.
Reverb Decay	0	127		Sets the decay time of the reverb.
Warble Speed	0	127		Sets the rate at which the tape playback speed is randomized.
Warble Amount	0	127		Sets the amount by which the tape playback speed is randomized.
Tape Inertia	0	127		Sets the rate at which the tape speed changes.

Name	*Min*	*Max*	*Units*	*Description*
Sync Source	UDT	Plug		Selects the synchronization source for Space Echo. The pop-up menu lets you choose from three sync modes: • UDT (User-Defined Tempo) lets you set the Tempo for Space Echo in BPM. • Host mode sets the Tempo to match the host tempo. • Plug mode sets the Tempo to match the PluggoSync plug-in if it is loaded. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Tempo	1	300	BPM	Sets the Tempo for Space Echo in beats per minute. This parameter, along with the Note and Multiplier parameters below, determines the delay time.
Sync Multiplier	0.01	75		Sets the multiplication factor that scales the delay time, as described below.
Sync Note Value	1	1/64t		Sets the base note duration for Space Echo. This note duration is multiplied by the Sync Multiplier to set the delay time, according to the current Tempo. For example, if you select a Sync Unit of 1/4 (a quarter note), and a Sync Multiplier of 1, the delay time will be set to once quarter note. With a Tempo of 120 BPM, this would be 500 ms. If you change the Sync Multiplier to 2, the delay time will be set to two quarter notes, or 1000 ms at 120 BPM. On the other hand, a Multiplier of 0.5 would set the delay time to one half of a quarter note, or 250 ms.

Spectral Filter

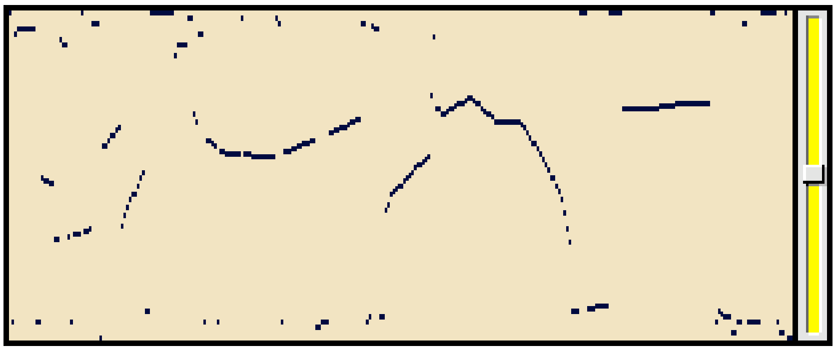
category: spectral domain

audio input: mono	*cpu: heavy*	*audio output: mono*

What It Does

Spectral Filter is like having a 253-band graphic equalizer. The input signal is processed by a 1024-point FFT-based analysis/resynthesis algorithm that adjusts the level of each frequency band according to the graph you draw. If your sequencing environment supports real-time parameter changes, you can record and play back changes in the graph as you scribble with the mouse.

Zack Settel and Cort Lippe developed the filtering technique employed by Spectral Filter. Watch for their forthcoming book or catch the *Convolution Brothers* on tour to hear spectral filtering used on a variety of texts.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Output Level	0	1		Sets an overall gain on the filter.

Insights

- The Randomize and Evolve commands in the parameter change pop-up menu can do some nice things that you couldn't otherwise draw.
- Try setting all the bands to zero and then drawing widely spaced dots. This allows only isolated frequency bands of the input to pass through the filter. The widely spaced dots inspired the original name of the effect— Forbidden Planet—which is supplied as an example with MSP.

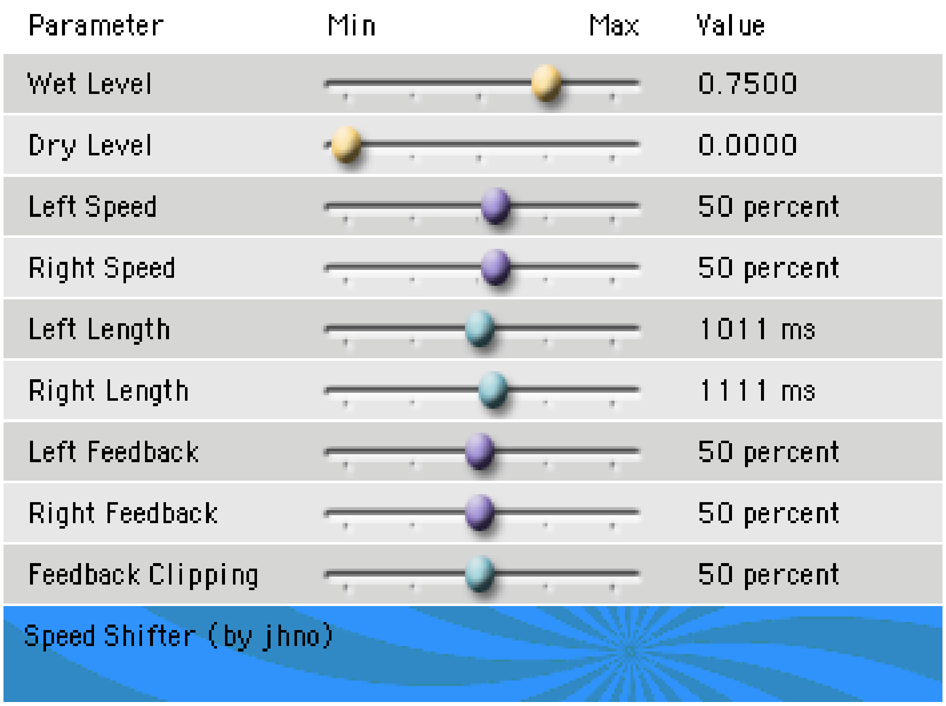
category: delay

Speed Shifter

audio input: mono | *cpu: light* | *audio output: stereo*

What It Does

Speed Shifter is like running your signal through two tape loops, each running at a different speed with a different length of tape, feeding back into each other. It's capable of some extreme effects as well as subtle doubling/thickening treatments.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the output gain of the processed signal.
Dry Level	0	1		Sets the output gain of the unprocessed signal.
Left Speed	-400	400	percent	Sets the speed of the left tape loop, specified as a percentage of the original pitch. Negative values will play the tape backward - this is often a good idea.
Right Speed	-400	400	percent	Sets the speed of the right tape loop, specified as a percentage of the original pitch. Negative values will play the tape backward - this is sometimes a good idea.
Left Length	0	2000	ms	Specifies the length of the left tape loop. The Speed setting and the type of input material will affect what works best here.
Right Length	0	2000	ms	Specifies the length of the right tape loop. The Speed setting and the type of input material will affect what works best here.

Speed Shifter

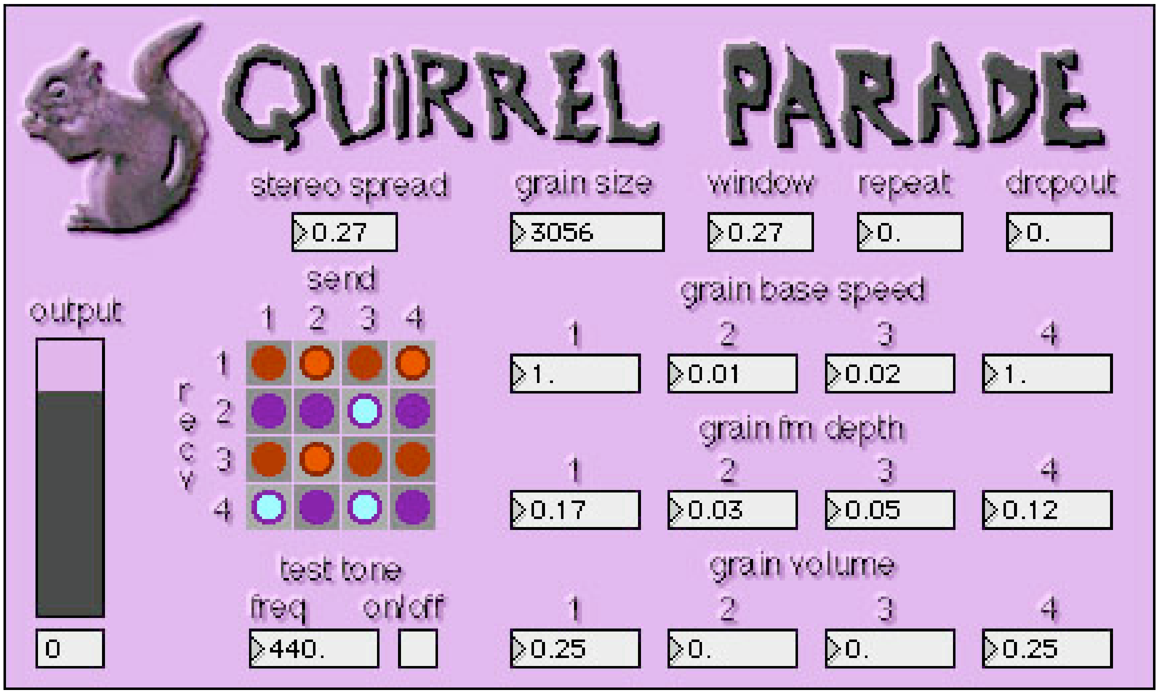
category: delay

audio input: mono *cpu: light* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Left Feedback	0	100	percent	Determines the amount of the left tape loop fed back into the Speed Shifter input.
Right Feedback	0	100	percent	Determines the amount of the right tape loop fed back into the Speed Shifter input.
Feedback Clipping	0	100	percent	Sets the output signal level at which clipping occurs in the feedback loop. This can be useful to limit the level of feedback.

What It Does

Squirrel Parade produces its unique output by means of Animal Modeling, which is actually the process of dividing an input channel into four granular channels. The playback of these granular channels is controlled by the grain base speed and frequency modulation parameters. A modulation matrix(that grid of red and blue dots) controls which granular channels modulate one another's playback speed. And best of all, no squirrels are harmed in the creation of the resulting fracas.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
grain size	4	44100	samples	Sets the size of each grain.
window	0.0	1.0	percent	Sets the duration of a linear fade in/out at the beginning/end of each grain.
stereo spread	-1.0	1.0		Specifies how widely the output is spread across the stereo field. A spread value of 0 produces monaural output. Negative values spread the output signal from left to right, and positive values spread the output from right to left.
repeat	0 .0	1.0	percent	Sets the percentage chance that the frequency-modulated granular output will be repeated.
dropout	0 .0	1.0	percent	Sets the percent chance that the output grains will be muted.

Squirrel Parade

category: granular

audio input: mono *cpu: medium* *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
send/recv matrix				The send/recv matrix is used to set which granular channel or group of granular channels will modulate another granular channel. Clicking on a granular channel number in the send column will select the channel as a carrier for frequency modulation, and the vertical receive column identifies the granular channel that fill function as the modulator waveform.
grain base speed	0.0	10.0		Sets the playback rate for each of the four granular channels. A playback rate of 1.0 is normal speed.
grain fm depth	0.0	10.0		Sets the amount of modulation that will be applied to the frequency modulation for each of the four grain channels. The mix of each of the four granular channels to be modulated is set using the send/receive matrix. The summed send channel outputs for each of the four granular channels act as the carrier, and the receive channel of each of the four granular channels acts as the modulator.
grain volume	0.0	0.25	percent	Sets the volume of each of the four granular channels.
output	-76	+18	dB	Sets the output amplitude of the effect.
test tone freq	0.0	2000.0	Hz	Sets the frequency in Hz of a test tone for the effect.
test tone toggle	off	on		Turns the test tone on and off.

Insights

- The test tone can be used to get a rough idea of how the effect is warping the input.

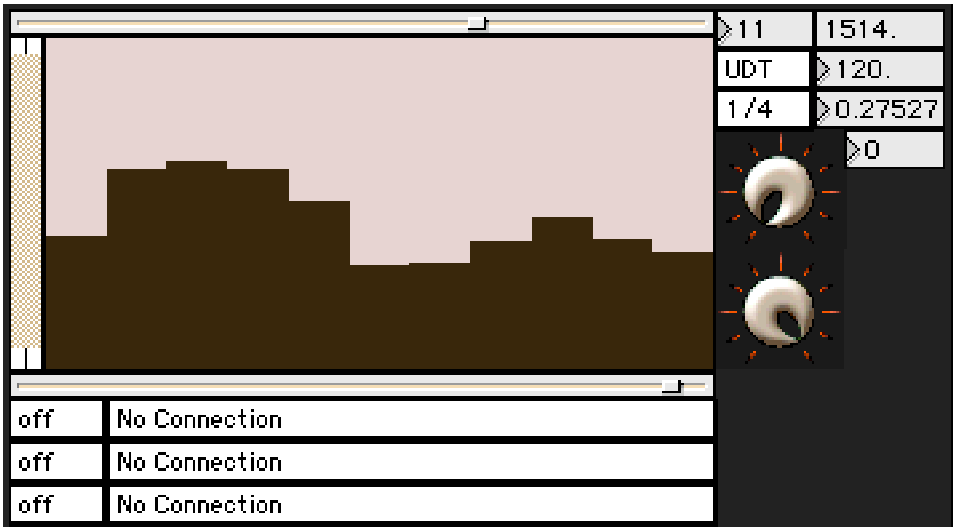
category: modulator — *accepts sync*

Step Sequencer

audio input: none — *cpu: light* — *audio output: thru*

What It Does

Step Sequencer is a Modulator plug-in that generates a control signal from a sequencer inspired by an analog control voltage system. *Step Sequencer* does not process audio, it passes its input signal to its output.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Min	0	127		Sets the minimum output value of the control signal.
Max	0	127		Sets the maximum output value of the control signal.
Steps	0	31		Sets the number of steps in the sequence.
Loop Time				Displays the length of a single cycle of the step sequencer in milliseconds. You can not change this value directly - set the speed of the sequencer using the BPM, Note, and Multiplier Parameters.
				Selects the synchronization source for Space Echo. The pop-up menu lets you choose from three sync modes: • UDT (User-Defined Tempo) lets you set the Tempo for Space Echo in BPM. • Host mode sets the Tempo to match the host tempo. • Plug mode sets the Tempo to match the PluggoSync plug-in if it is loaded. Note: Host mode is only available in VST 2.0 and MAS applications that support it.

Step Sequencer

accepts sync | *category: modulator*

audio input: none | *cpu: light* | *audio output: thru*

Name	*Min*	*Max*	*Units*	*Description*
Sync Source	UDT	Plug		This pop-up menu lets you select the synchronization source for Step Sequencer. The pop-up menu lets you choose from three sync modes: • UDT (User-Defined Tempo) lets you set the Step Sequencer tempo in BPM. • Host mode sets the sequencer tempo to match the host tempo. • Plug mode sets the sequencer tempo to match the PluggoSync plug-in if it is loaded. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Tempo	1	300	BPM	Sets the tempo of Step Sequencer in beats per minute. This parameter, along with the Note and Multiplier parameters below, determines the rate of the sequencer.
Step Time Multiplier	0.001	100		Sets the multiplication factor that scales Step Sequencer's speed, as described below.
Step Note Unit	1	1/64		Sets the base note duration for a single step of the sequencer. This note duration is multiplied by the Step Time Multiplier to set the sequencer's speed, according to the current Tempo. For example, if you select a Step Note Unit of 1/16 (a quarter note), and a Step Time Multiplier of 1, the sequencer will play through one step every sixteenth note. At a Tempo of 120 BPM, each step would take 125 ms. So, if your sequence contained 8 steps, it would take 1000 ms to cycle through the sequence - which is half a bar in 4/4 time. If you then changed the Step Time Multiplier to 0.5, each step would take half as long, and the sequencer would run twice as fast.
Attack	0	1		Sets the time required to reach the current step value. This functions as a portamento.
Sustain	0	1		Sets the value to reach after the Attack segment. This causes a decay in the output.

Interface Elements

- When the Sync mode is set to Free, the sequence can be reset to the beginning by clicking the round Reset button beneath the Sync pop-up menu.

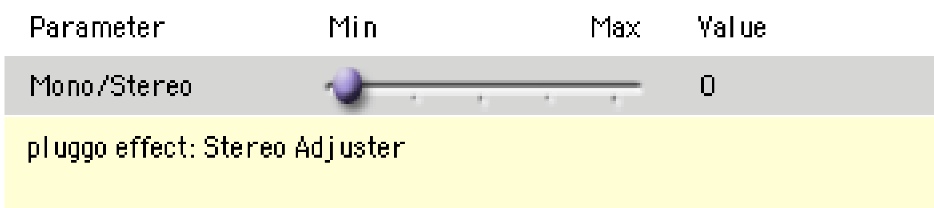
category: multichannel

Stereo Adjuster

audio input: stereo *cpu: light* *audio output: stereo*

What It Does

Stereo Adjuster adjusts the stereo spread of an existing stereo recording. Setting the slider in the middle means that stereo is unchanged. Move it to the right and the sound becomes distant (by mixing in a left-minus-right signal). Move it to the left and the sound becomes close and mono.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Mono/Stereo	0	127		When the value is 0, the two channels are mixed together to create a mono output. As the slider is moved to the right, the stereo effect is spread farther apart.

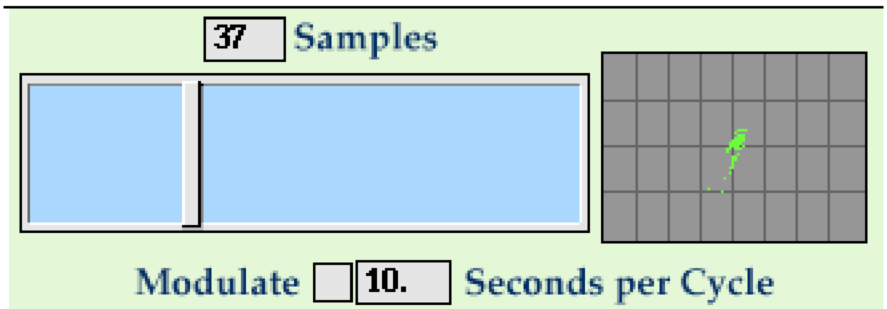
Stereo Faker

category: multichannel

audio input: mono *cpu: light* *audio output: stereo*

What It Does

Stereo Faker creates a stereo image from a mono signal splitting it into a comb filter and its complement comb filter. Different images are possible by adjusting the sample delay.

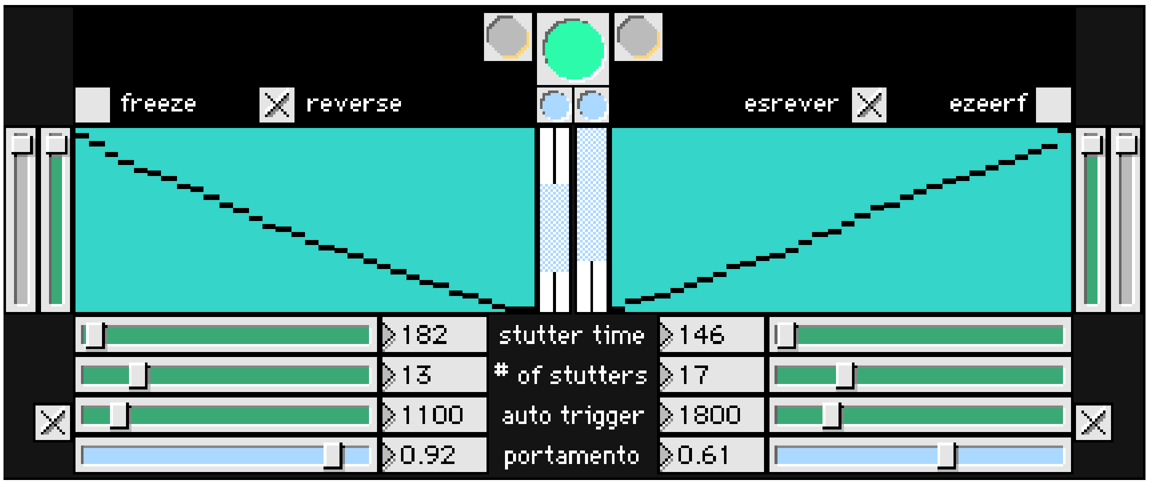


Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Delay Time	1	127	samples	Sets the delay time of the comb filter used to create the stereo image.
Sweep Time	10	10	secs/cycle	Sets rate of modulation of the comb filter delay time.
Modulate	Off	On		Turns modulation of the comb filter delay time on and off.

What It Does

Stutterer is designed to play back fragments of its input signal in a number of different ways. Operated manually, you can grab a "snapshot" of the audio and loop it a specified number of times. In automatic mode, it will do this for you. You can change the pitch of the fragments, play them backward, and more to get a variety of deconstructive effects. Applications of *Stutterer* include mangling of drum loops in the manner of experimental jungle/drum and bass composers, or getting that M-M-Max Headroom™ effect, etc.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
LeftDry	0	127		Sets the output level of the unstuttered signal for the left channel.
RightDry				Sets the output level of the unstuttered signal for the right channel.
LeftWet	0	127		Sets the level of the stutter output for the left channel.
RightWet	0	127		Sets the level of the stutter output for the right channel.
LeftNumber	0	74	stutters	Sets the number of stutters per trigger for the left channel.
RightNumber	0	74	stutters	Sets the number of stutters per trigger for the right channel.
Left Stutter Time	1	60000	ms	Sets the length, in milliseconds, of each stutter for the left channel. For example, if this parameter is set to 500, when you hit the stutter button you will hear the last half-second of the input signal.
Right Stutter Time				Sets the length, in milliseconds, of each stutter for the right channel.
Left AutoTrigger	Off	On		When On, the left channel stutters trigger by themselves repeatedly at an interval set by the Left Trigger Time.

Stutterer

category: delay

audio input: mono, stereo | *cpu: light* | *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Right AutoTrigger	Off	On		When On, the right channel stutters trigger by themselves repeatedly at an interval set by the Right Trigger Time.
Left TriggerTime	1	60000	ms	Sets the time interval between auto triggers of left channel stutters.
Right Trigger Time	1	60000	ms	Sets the time interval between auto triggers of right channel stutters.
Left Portamento	0	1		Specifies how fast the pitch of the left channel stutters will change. Set Portamento to zero to hear the individual steps of the Pitch Trajectory, and adjust it higher to smooth out the pitch motion
Right Portamento				Specifies how fast the pitch of the right channel stutters will change. Set Portamento to zero to hear the individual steps of the Pitch Trajectory, and adjust it higher to smooth out the pitch motion
Left Freeze	Off	On		Holds the current left channel stutter in place, so you can retrigger it without resampling the input signal.
Right Freeze				Holds the current right channel stutter in place, so you can retrigger it without resampling the input signal.
Left Reverse	Off	On		When On, the left channel will play its stutters backward.
Right Reverse	Off	On		When On, the right channel will play its stutters backward.
Left Pitch Min	0	127		Sets the low end of the range of the Pitch Trajectory for the left channel.
Right Pitch Min				Sets the low end of the range of the Pitch Trajectory for the right channel.
Left Pitch Max				Sets the upper end of the range of the Pitch Trajectory for the left channel.
Right Pitch Max				Sets the upper end of the range of the Pitch Trajectory for the right channel.
Left Bang	Off	On		A value of 1 (On) triggers a stutter for the left channel. Clicking on the round button above the left pitch trajectory will trigger a stutter.
Right Bang	Off	On		A value of 1 (On) triggers a stutter for the right channel. Clicking on the round button above the right pitch trajectory will trigger a stutter.
Big Bang	Off	On		A value of 1 (On) triggers the stutters of both channels in one fell swoop.

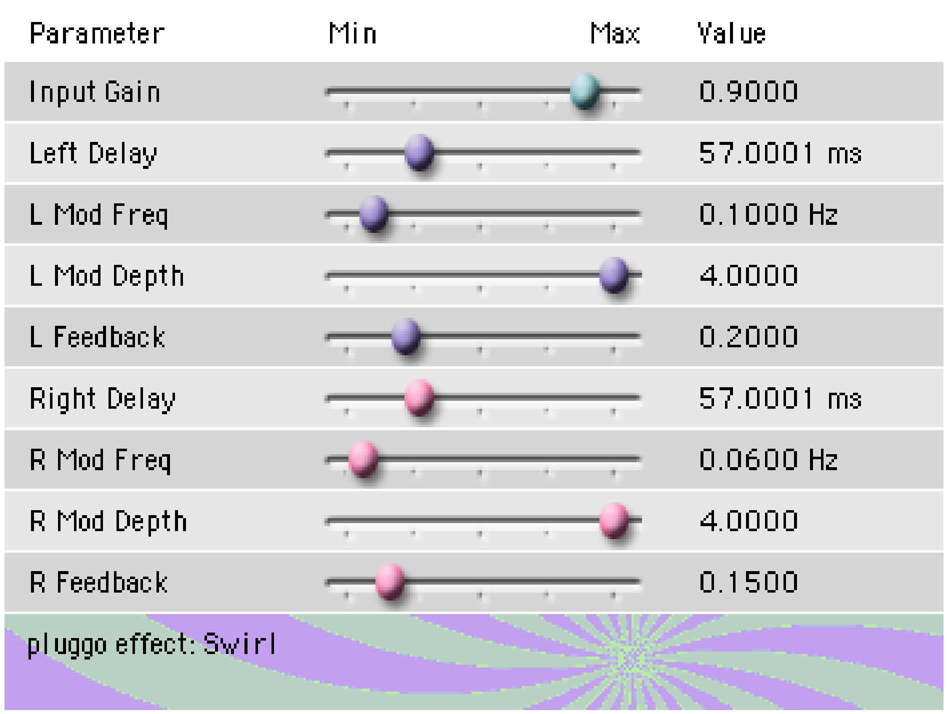
category: delay

Swirl

audio input: mono *cpu: light* *audio output: stereo*

What It Does

Swirl is a mono-to-stereo effect that uses only delay-time modulation to create a variety of moving ambiences. Delay-based "panning" uses the auditory system's sensitivity to interaural time differences, whereas traditional amplitude-based panning effects (exemplified by plug-ins such as *Audio Rate Pan* and *Nebula*) use our sensitivity to interaural intensity differences.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Input Gain	0	1		Sets the gain on the input to the delay network.
Left Delay	0	210	ms	Sets the delay time for the left output.
L Mod Freq	0	1	Hz	Sets the frequency of delay time modulation of the left output.
L Mod Depth	0	4		Sets the amount of delay time modulation of the left output. As the value of this parameter increases, the output will begin to exhibit vibrato characteristics.
L Feedback	0	.9		Sets the amount of feedback for the left output.
Right Delay	0	210	ms	Sets the delay time for the right output.
R Mod Freq	0	1	Hz	Sets the frequency of delay time modulation of the right output.
R Mod Depth	0	4		Sets the amount of delay time modulation of the right output. As the value of this parameter increases, the output will begin to exhibit vibrato characteristics.

Swirl

category: delay

audio input: mono *cpu: light* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
R Feedback	0	.9		Sets the amount of feedback for the right output before it is phase inverted. Note that the signal of the left delay line is not phase inverted before it is fed back.

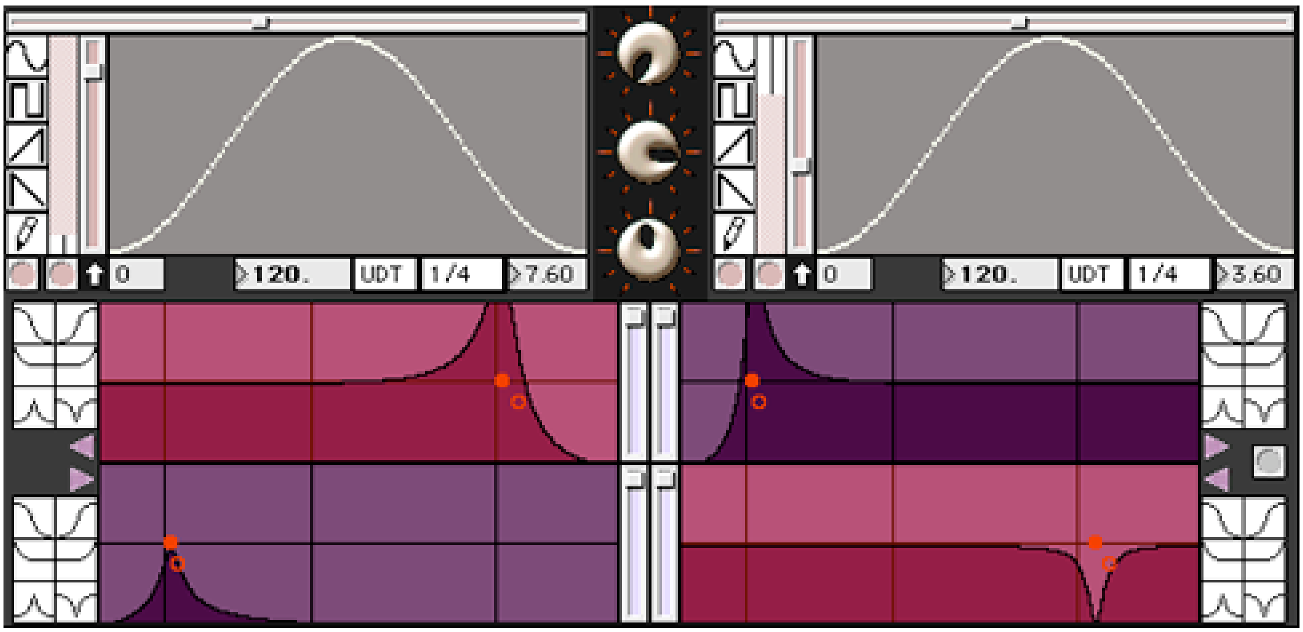
Insights

- Why doesn't Swirl have a "Dry Level" parameter? Because its delay-based panning effects are pretty much destroyed when you mix the original signal in with the processed one. It's recommended that you use *Swirl* as an insert effect for this reason. It's subtle and fickle.
- How do you control the rate and trajectory of the movement of the sound? There isn't a hard-and-fast rule, but consider the *Travelling* preset. This moves back and forth at the rate of about once a second. To increase the speed, hold down the option key while clicking on the Right Delay parameter so you can make precise adjustments. By increasing this parameter to 4.5 ms, you'll increase the swirl rate to about twice a second. But move it very far beyond this value, and the overall cue of one channel's delay time being significantly longer than the other takes over and the sound moves to one side of your head.
- The above example shows that there is an interaction between the slow modulation of the delay times and the difference between delay times in the right and left channels. To achieve the perception of the movement, you need both a slight difference in delay times and a (relatively slow) modulation of them.

category: filter	*accepts sync*	**Swish**
audio input: stereo	*cpu: medium*	*audio output: stereo*

What It Does

Swish consists of four filters and two LFOs that can create complex, sweeping filter effects. Either of the LFOs, or neither can control each of the filters. Other features include user-drawable LFO waveforms and six independent filter modes.



Filter Parameters

Name	*Min*	*Max*	*Units*	*Description*
Dry Level	0	1		Sets the output level of the original input signal.
Filter Level	0	1		Sets the output level of each filter.
Q	0	1		Sets the overall resonance characteristic of the four filters. They can be individually adjusted as well.
Gain	0	1		Sets the overall gain of the four filters. They can be individually adjusted as well.
Filter Frequency	20	20000	Hz	Sets the cutoff frequency of this filter.
Filter Gain	0	2		Sets the internal gain of this filter.
Filter Q	0	5		Sets the resonance characteristic of this filter.
Filter Mode	lowpass	notch		Select the filter mode for this filter. The choices are: low-pass, high-pass, low-shelf, high-shelf, band-pass, and notch.
Filter Switch	Off	LFO 2		Select LFO modulation for this filter. The choices are: Off, LFO 1, LFO 2.

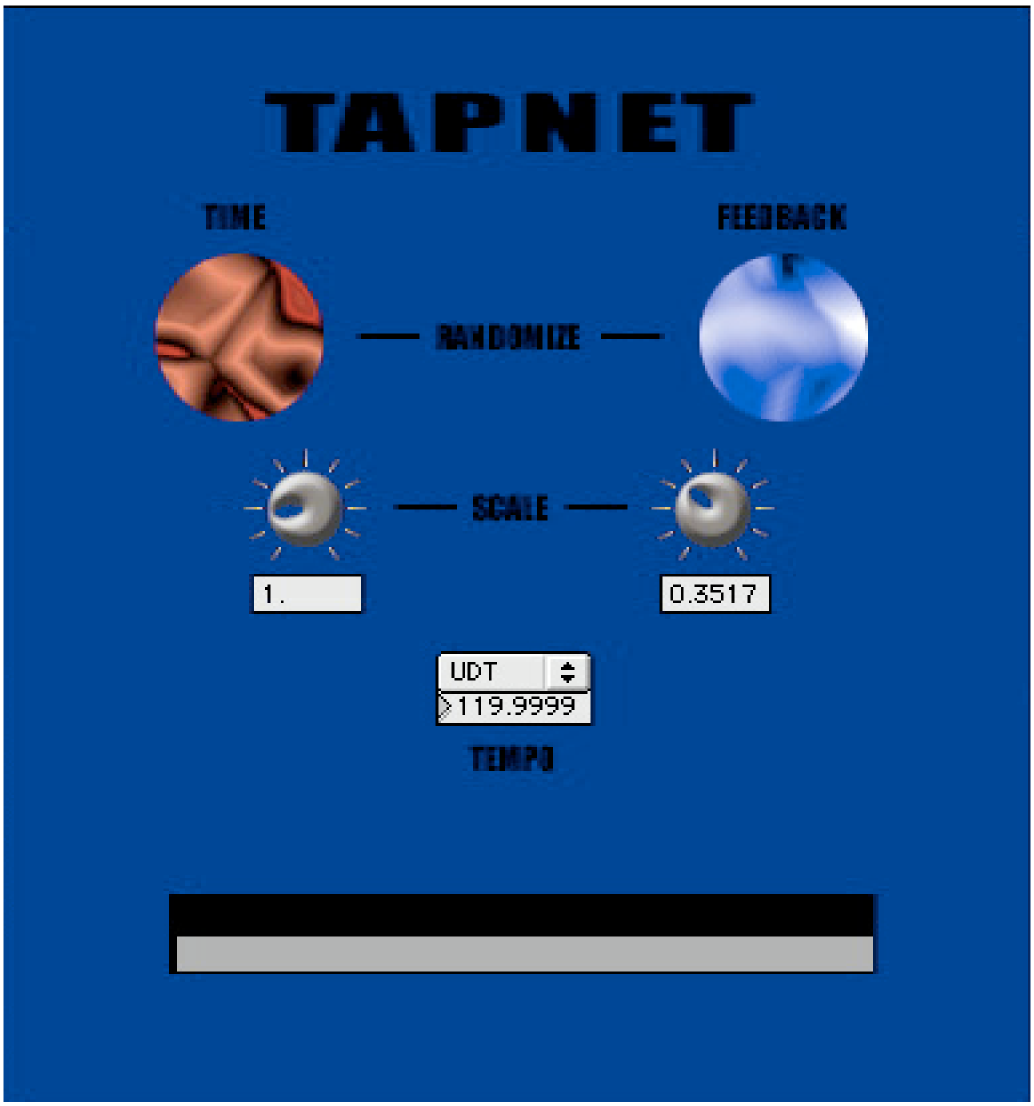
LFO Module Parameters

Name	*Min*	*Max*	*Units*	*Description*
LFO Waveform	Sine	Draw		Sets the current shape of the LFO. By clicking on the buttons to on the left side of the LFO module, you can select a Sine, Square, Saw Up, Saw Down, or User wave shape. When the User wave shape is selected, you can draw in the LFO Waveform display to set the LFO shape.
LFO Min	0	127		Sets the minimum value that the LFO will produce (when the LFO Waveform display shows its lowest value).
LFO Max	0	127		Sets the maximum value that the LFO will produce (when the LFO Waveform display shows its highest value).
LFO Phase	0	360	degrees	Offsets the start position of the LFO, in other words, where it starts in its waveform when it is retriggered. A value of 0 means the LFO starts at the beginning, and a value of 360 means it starts at the end. This parameter can be useful to employ when LFO modules are synchronized
LFO Sync Source	UDT	Plug		These pop-up menus let you select the synchronization source for each of Swish's two LFOs. The pop-up menus let you choose from three sync modes: • UDT (User-Defined Tempo) lets you set the LFO tempo in BPM. • Host mode sets the LFO tempo to match the host tempo. • Plug mode sets the LFO tempo to match the PluggoSync plug-in if it is loaded. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
LFO Tempo	1	300	BPM	Sets the LFO tempo in beats per minute. This parameter, along with the Note and Multiplier parameters below, determines the rate of the LFO cycle.
LFO Sync Multiplier	0.01	75		Sets the multiplication factor that scales the LFO speed, as described below.
LFO Sync Note Value	1	1/64t		Sets the base note duration for a single LFO cycle. This note duration is multiplied by the LFO Sync Multiplier to set the LFO speed, according to the current LFO Tempo. For example, if you select a Sync Unit of 1/4 (a quarter note), and a Sync Multiplier of 1, the LFO will cycle once every quarter note. With an LFO Tempo of 120 BPM, this would be one cycle every 500 ms, or 2 Hz. If you change the Sync Multiplier to 2, the LFO will cycle once every two quarter notes: twice as slow. On the other hand, a Multiplier of 0.25 will make it four times faster - one cycle every sixteenth note.

Note: Parameters labeled "301. LFO 1 Sync" and "306. LFO 2 Sync" will appear in the Assign menu of any Modulator plug-in when the *Swish* plug-in is loaded. Modulating these parameters will have no effect.

What It Does

TapNet is a four-tap delay line with a couple of unique features. Most importantly, instead of one feedback path from the output to the input, each individual tap has a feedback path to every other tap and itself, also. *TapNet* supports host sync, which can keep delay times in tempo - even when you Randomize them. And finally, each tap delay has an independent modulator that lets you do pitch-shifting and chorus-type effects.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*

TapNet

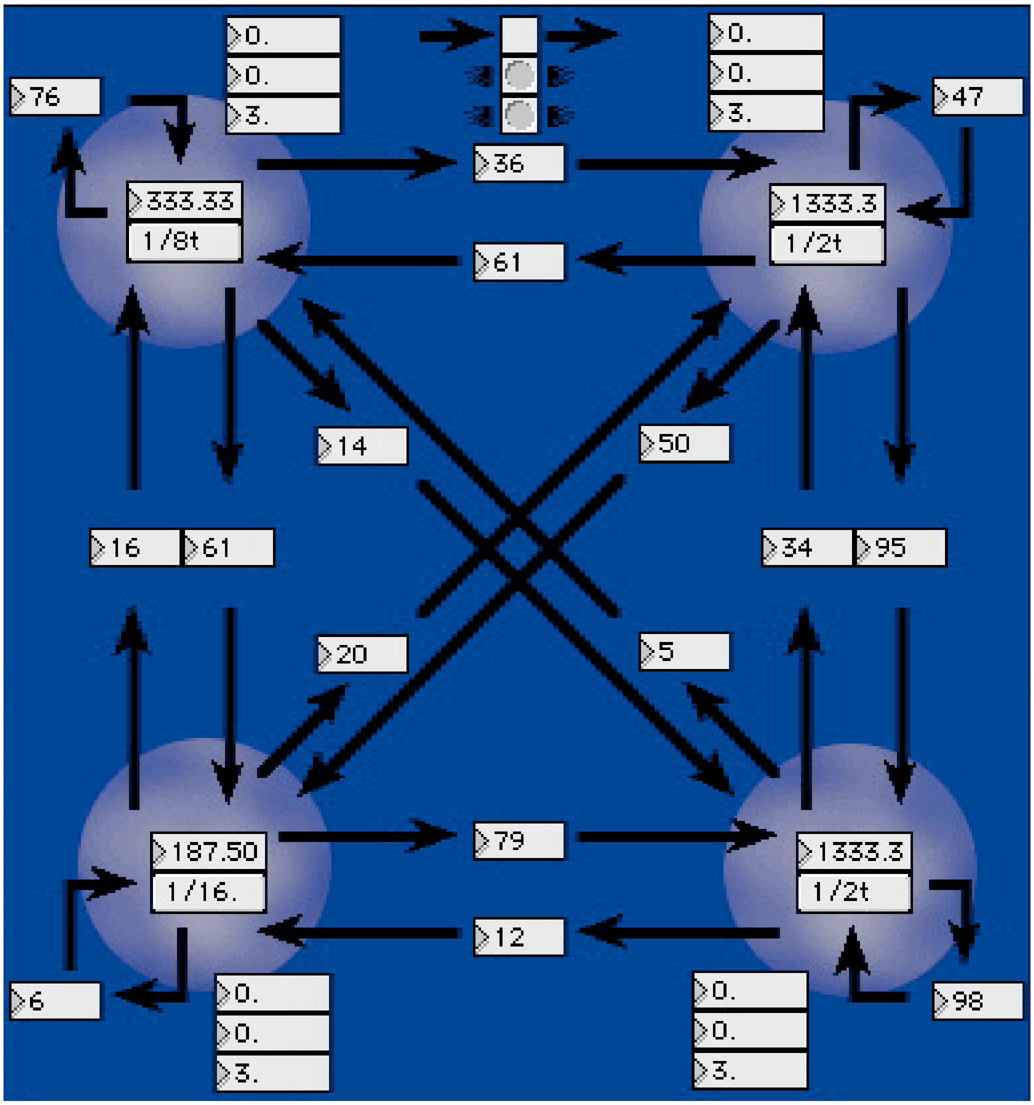
accepts sync | *category: delay*

audio input: mono | *cpu: medium* | *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Randomize Time				Clicking on this button will generate a new set of time values for the TapNet delay lines in the network. You can view the current delay network settings using the NN view, or using the eggslider views.
Randomize Feedback				Clicking on this button will generate a new set of feedback values for the TapNet delay line network. You can view the current feedback settings using the NN view, or using the eggslider views.
Scale Time	0.0	5.0		Scales the overall time of each tape in the delay network.
Scale Feedback	0	1.000		Scales the feedback amount of each tap in the delay network to a fraction of the total amplitude.
Tempo	1	300	BPM	Sets the global tempo. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the delay time. When in Host and Plug modes, the tempo is an indicator that cannot be changed-it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range and TapNet will calculate the delay times based on the values of each node in the delay network and the Scale Time Units parameter value. This parameter is disabled when the Sync mode is set to Free. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Sync Mode	Free	UDT		Sync Mode is a pop-up menu in the Tempo/Sync parameter. There are four modes of sync available: • Free mode lets you set the delay time independent of the host sequencer. • Host mode synchronizes the delay time to the host tempo. • Plug mode synchronizes the delay time to the beat output of PluggoSync • UDT (User-Defined Tempo) mode is equivalent to Free mode but allows you to set the delay time in terms of tempo and note unit values. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Dry Level	-70	+18	dB	Sets the output level of the original, undelayed signal.
Wet Level	-70	+18	dB	Sets the output level of the delay network signal.

Details View

The Details interface includes a visual representation of *TapNet*'s internal signal routing. It lets you see exactly what is happening and lets you adjust all the parameters of the delay lines. The Details interfaces shows how input and output are routed. The left channel of the plug-in's audio input is sent to delay 1 (shown at the top left), and the right channel to delay 2 (shown at the top right). Delay 3 is shown at the bottom left), and delay 4 is at the bottom right. All internal routing among the delays is set by the feedback parameters. Then, delays 1, 2, 3, and 4 are routed to left, right, left, and right outputs, respectively. Each delay also has three vertical number boxes that are used to modulate the delay lines for chorusing effects. These parameters are described in greater detail in the Eggslider View section below



Eggslider View

The Eggslider view shows all the parameters of the delay network. Since the delay network is complex, there are three eggslider interface panels.

TapNet

accepts sync | *category: delay*

audio input: mono | *cpu: medium* | *audio output: stereo*

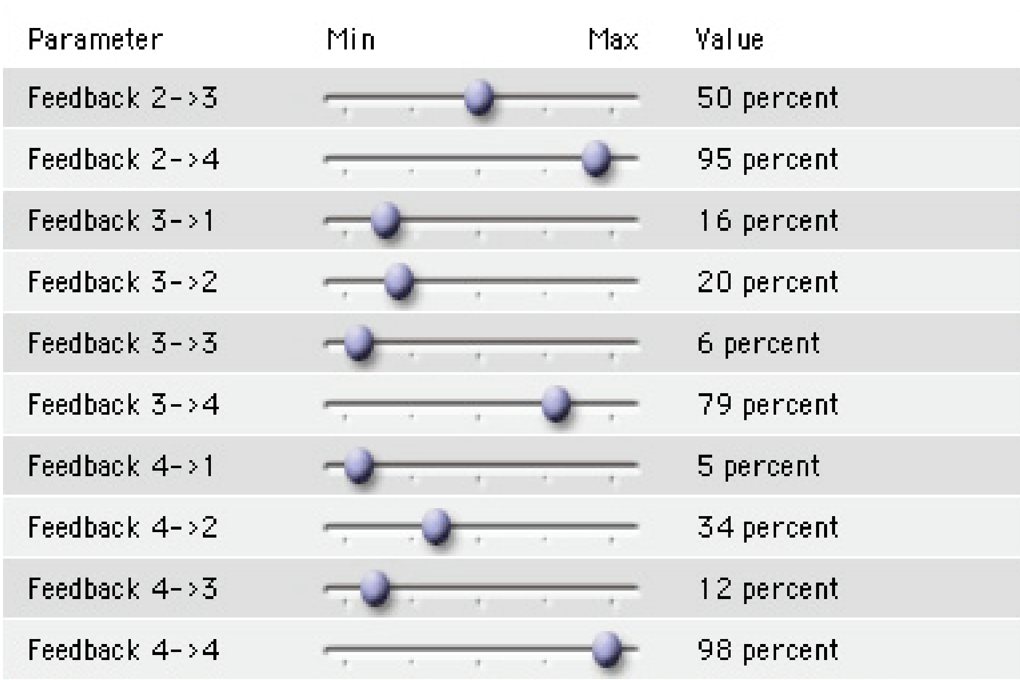
Parameter	Min	Max	Value
Wet Level			1.0000
Dry Level			0.0000
Delay Time Scale			1.0000 times
Feedback Scale			0.3517 gain
Random Delays			0
Random Feedback			0
Link Delays 1/2			off
Global Tempo		UDT	119.9999
Delay 1 Time		1/8t	1.0000 * 1/8t
Delay 1 Time		1/2t	1.0000 * 1/2t
Delay 1 Time		16.	1.0000 * 16.
Delay 1 Time		1/2t	1.0000 * 1/2t
Delay 1 Mod Amount			0.0000 units
Delay 1 Mod Amount			0.0000 units
Delay 1 Mod Amount			0.0000 units

Parameter	Min	Max	Value
Delay 1 Mod Amount			0.0000 units
Delay 1 Mod Speed			0.0000 Hz
Delay 1 Mod Speed			0.0000 Hz
Delay 1 Mod Speed			0.0000 Hz
Delay 1 Mod Speed			0.0000 Hz
Delay 1 Mod Duck			3.0000 percent
Delay 1 Mod Duck			3.0000 percent
Delay 1 Mod Duck			3.0000 percent
Delay 1 Mod Duck			3.0000 percent
Feedback 1->1			76 percent
Feedback 1->2			36 percent
Feedback 1->3			61 percent
Feedback 1->4			14 percent
Feedback 2->1			61 percent
Feedback 2->2			47 percent

TapNet

accepts sync | *category: delay*

audio input: mono | *cpu: medium* | *audio output: stereo*



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	-70	+18	dB	Sets the output level of the delay network signal.
Dry Level	-70	+18	dB	Sets the output level of the original, undelayed signal.
Delay time scale	0.0	5.0		Scales the overall time of each tape in the delay network.
Feedback scale	0	1.000		Scales the feedback amount of each tap in the delay network to a fraction of the total amplitude.
Random delays	0	1		Generates a new set of time values for the TapNet delay lines in the network.
Random feedback	0	1		Generates a new set of feedback values for the TapNet delay line network.
Link delays 1/2	0	1		When TapNet is operating in Host Sync mode, this value is set automatically.

Name	Min	Max	Units	Description
Global tempo	1	300	BPM	Sets the global tempo. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the delay time. When in Host and Plug modes, the tempo is an indicator that cannot be changed- it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range and TapNet will calculate the delay times based on the values of each node in the delay network and the Scale Time Units parameter value. This parameter is disabled when the Sync mode is set to Free. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Sync Mode	Free	UDT		Sync Mode is a pop-up menu in the Tempo/Sync parameter. There are four modes of sync available: • Free mode lets you set the delay time independent of the host sequencer. • Host mode synchronizes the delay time to the host tempo. • Plug mode synchronizes the delay time to the beat output of PluggoSync • UDT (User-Defined Tempo) mode is equivalent to Free mode but allows you to set the delay time in terms of tempo and note unit values. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Delay time (1-4)	0	1000	ms	Sets the delay time between taps. The delay time is set in milliseconds when the Sync Mode (shown in the Tempo/Sync parameter) is set to Free. In Host, Plug, and UDT modes, the delay time parameter is specified using note units and a multiplier.
Delay Time Mult (1-4)	0	1.000		Sets a multiple of the Delay Time Units currently selected that determine the delay time at the current tempo. This parameter is disabled when the Sync Mode is set to Free.
Note values (1-4)	1	64t.		Sets the base note duration value used in determining the delay time in relation to the current tempo. This value is multiplied by the Delay Time to obtain the total beat value used to calculate the delay time. This parameter is disabled when the Sync Mode is set to Free.
Delay mod amount (1-4)	-100.0	100.0	ms	Sets the modulation delay time in milliseconds.
Delay mod speed (1-4)	0	500	Hz	Sets the frequency of the sine wave, which modulates the delay tap.
Delay mod duck (1-4)	3	100	Percent	I have no idea what this is. Do you?

TapNet

accepts sync | *category: delay*

audio input: mono | *cpu: medium* | *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Feedback	0	100	percent	Sets the percentage of the delayed signal to be fed back into the delay network. There is one value for every node in the feedback network, for a total of 16 feedback loops.

Insights

- The easiest way to use *TapNet* is via the "Randomize" buttons on the Global interface page. These let you scramble all the internal delay time and feedback values, so you don't have to laboriously set them yourself. Just hit the "Randomize Delay Times" button until you like the resulting timing, and then hit "Randomize Feedback" to tweak the texture. The "Scale" parameters for Delay Time and Feedback are very useful to adjust all the delay lines at once.
- Beware of runaway feedback.
- *TapNet* can be set to any of the usual Sync modes. In "Free" mode all delay times are set in milliseconds, and the "Note Value" settings have no effect. In "Host," "PluggoSync," or "User-Defined Tempo (UDT)" modes, the delay times are determined by the specified Note Values. If you hit "Randomize Delay Times" while in a tempo sync mode, the Note Values will be randomized and all of the delays will stay in proportion to one another - a nice effect, especially if the tempo is in sync with your sounds.
- As in other sync-enabled plug-ins, each delay line has an individual multiplier that can scale its Note Value. To adjust the multipliers, use the eggslider interface pages. Otherwise, the multipliers are always set to "1", so that the Tempo and Note Values determine the delay times directly.

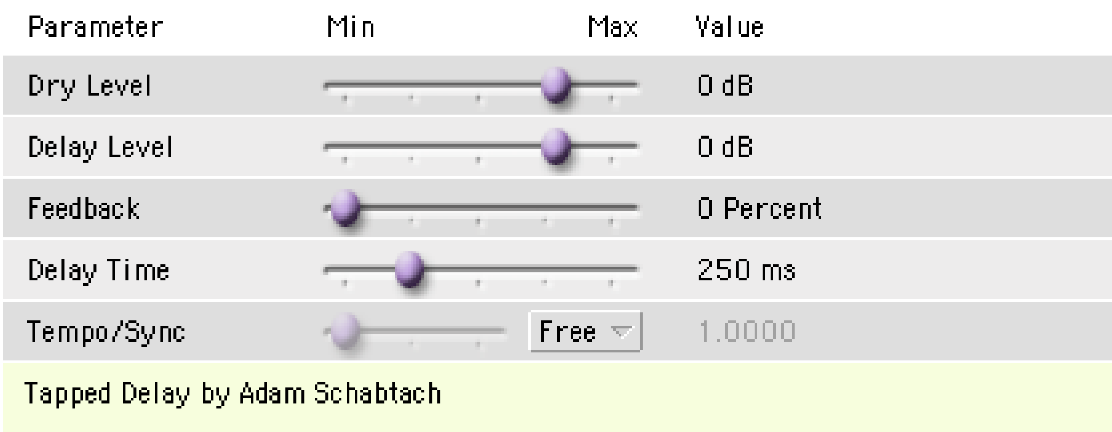
category: delay | *accepts sync*

Tapped Delay

audio input: mono | *cpu: medium* | *audio output: stereo*

What It Does

Tapped Delay is a delay line with 16 equally spaced output taps. You can adjust the output level and stereo position of each tap independently.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Dry Level	-70	+18	dB	Sets the output level of the original, undelayed signal.
Delay Level	-70	+18	dB	Sets the output level of the all the taps.
Feedback	0	100	%	Sets the amount of the signal from the last delay tap that is fed back into the beginning of the delay line.
Delay time	0	1000	ms	Sets the delay time between taps. The delay time is set in milliseconds when the Sync Mode (shown in the Tempo/Sync parameter) is set to Free. In Host, Plug, and UDT modes, the delay time parameter is specified using note units and a multiplier.
Delay Time Mult	0	1.000		Sets a multiple of the Delay Time Units currently selected that determines the delay time at the current tempo. This parameter is disabled when the Sync Mode is set to Free.
Delay Time Units	1	64t.		Sets the base note duration value used in determining the delay time in relation to the current tempo. This value is multiplied by the Delay Time to obtain the total beat value used to calculate the delay time. This parameter is disabled when the Sync Mode is set to Free.

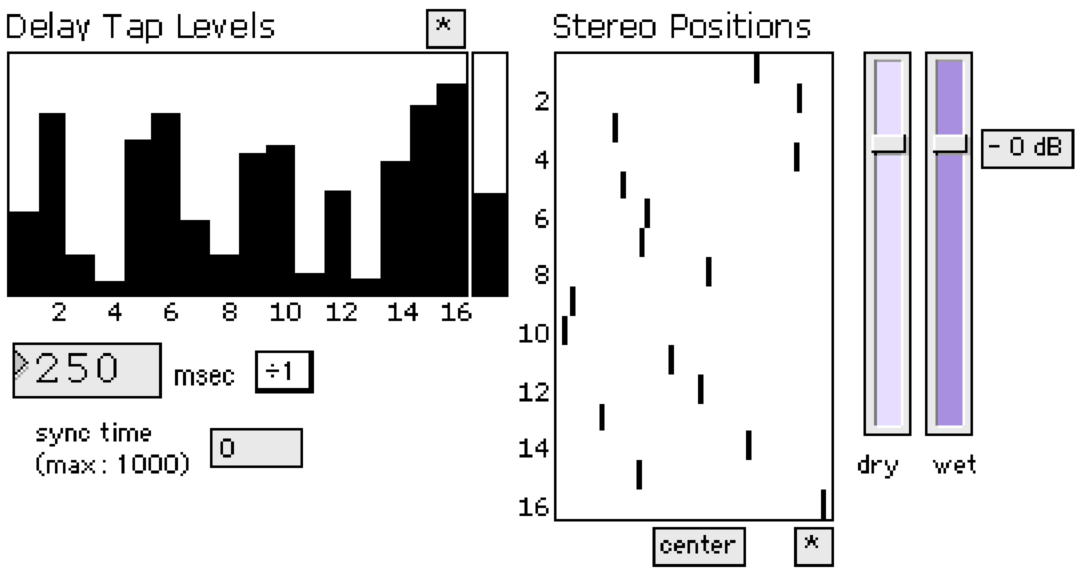
Tapped Delay

accepts sync | *category: delay*

audio input: none | *cpu: medium* | *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Tempo	1	300	BPM	Tempo is displayed as a slider in the Tempo/Sync parameter. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the delay time. When in Host and Plug modes, the tempo is an indicator that cannot be changed-it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range and Tapped Delay will calculate the delay time based on the values of the Delay Time Mult and Delay Time Units parameters. This parameter is disabled when the Sync mode is set to Free. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Sync Mode	Free	UDT		Sync Mode is a pop-up menu in the Tempo/Sync parameter. There are four modes of sync available: • Free mode lets you set the delay time independent of the host sequencer. • Host mode synchronizes the delay time to the host tempo. • Plug mode synchronizes the delay time to the beat output of PluggoSync • UDT (User-Defined Tempo) mode is equivalent to Free mode but allows you to set the delay time in terms of tempo and note unit values. Note: Host mode is only available in VST 2.0 and MAS applications that support it.

Delay View



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Delay Tap Levels (multislider)	0	100	%	Sets the relative output level of each delay tap. The numbers along the bottom indicate the tap that corresponds to each slider.
Feedback	0	100	%	Sets the amount of the signal from the last delay tap that is fed back into the beginning of the delay line.
Stereo Positions (multislider)				Sets the stereo placement of each delay tap. The numbers along the side of the multislider indicate which tap is controlled by which slider.
Delay time	1	1000	ms	Sets the delay time between each tap. Since there are 16 taps, and the delay time has a maximum value of 1000 ms, the output of the last tap is delayed by a maximum time of 16 seconds.
Delay-time divisor	1	8		Divides the delay time by this number. For example, if the delay-time numerical is set to 500 ms, and the divisor is set to 4, the delay taps will be set to intervals of 125 ms.
Dry	-70	+18	dB	Sets the output level of the original, undelayed signal.
Wet	-70	+18	dB	Sets the output level of the all the taps.
PluggoSync checkbox	Off	On		If this box is checked, the delay time is set to the current pluggoSync interval. This time is displayed in the box to the right of the checkbox.

Tapped Delay

accepts sync *category: delay*

audio input: none *cpu: medium* *audio output: stereo*

Interface Elements

- The • buttons randomize the delay tap level and stereo position multisliders.
- The **center** button sets all of the stereo position sliders to the center.
- The **0 dB** button sets both the Wet and Dry sliders to unity gain.
- Feedback can be controlled by the vertical slider to the right of the Delay Tap Levels sliders.

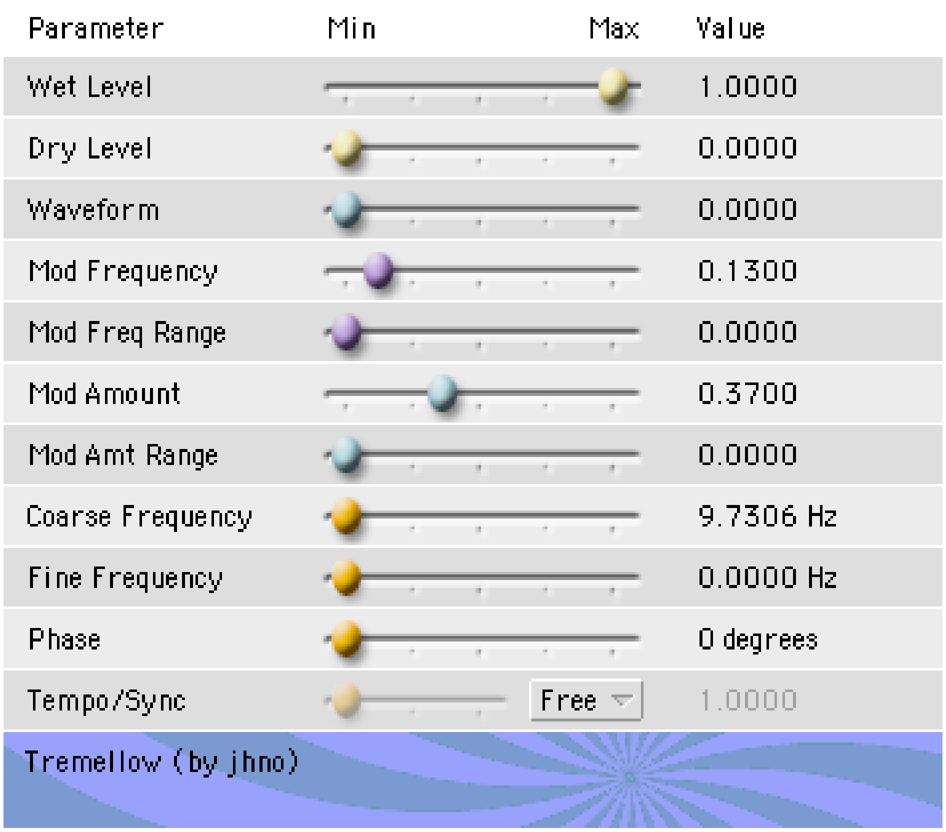
Insights

- Short delay times can create chorusing or fixed flanging effects. Long delay times can create intricate rhythmic counterpoint with short sounds (e.g., bass lines, drums & percussion, bouzouki).
- The first three presets illustrate how changing the delay-time divisor can create different rhythmic effects. Try these presets with a very simple audio signal consisting of just a short note on the first bar of each measure, played at 120 bpm. While you're at it, try clicking the randomization buttons.

category: multichannel	*accepts sync*	**Tremellow**
audio input: mono	*cpu: light*	*audio output: stereo*

What It Does

Tremellow is a stereo panning effect, similar *Audio Rate Pan*, yet different in its own way, as *Rye* is to *Wheat*.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the gain on the processed signal.
Dry Level	0	1		Sets the gain on the unprocessed signal.
Waveform	0	1		Lets you change the shape of the tremolo continuously, from a sine wave to a square wave. For AM/ring modulation effects, you probably want to keep it near 0—a pure sine wave.
Mod Frequency	0	1		Specifies the frequency of a sine wave that modulates the speed of the tremolo effect. This parameter is disabled when the Sync mode is set to Host, Plug, or UDT.

Tremellow

accepts sync *category: multichannel*

audio input: mono *cpu: light* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Mod Freq Range	0	1		Sets the maximum value of the Frequency parameter - again, letting you experiment precisely with slow ranges, or push it into audio frequencies. This parameter is disabled when the Sync mode is set to Host, Plug, or UDT.
Mod Amount	0	1		Adjusts the amount of modulation applied to the tremolo frequency. This parameter is disabled when the Sync mode is set to Host, Plug, or UDT.
Mod Amt Range	0	1		Sets the maximum value of the Mod Amount parameter. Larger values mean a greater range of possible Mod Amount values. This parameter is disabled when the Sync mode is set to Host, Plug, or UDT.
Coarse Frequency	0	10000	Hz.	Sets the frequency of the tremolo effect. This parameter has a wide (coarse) range to allow audio-rate frequencies for amplitude modulation effects. The slider is displayed differently depending on the Sync Source you have selected (see below). In Free mode, it appears as a single slider that lets you set the tremolo frequency in Hz. In Host, Plug, or UDT mode, it appears as a slider and a pop-up menu. The menu lets you select a note duration, and the slider sets a multiplication factor. These are multiplied together to set the tremolo speed according to the current tempo. For example, if you select a Sync Unit of 1/4 (a quarter note), and a Sync Multiplier of 1, Tremellow will cycle once every quarter note. At 120 BPM, this would be one cycle every 500 ms, or 2 Hz. If you change the Sync Multiplier to 2, Tremellow will cycle once every two quarter notes: twice as slow. On the other hand, a Multiplier of 0.25 will make it four times faster - one cycle every sixteenth note.
Fine Frequency	0	50	Hz	Adjusts the frequency of the tremolo effect. This parameter is added to the Coarse Frequency value to determine Tremellow's speed, allowing you to make fine adjustments. Remember, you can double-click or option-click on the slider for even more precision. This parameter is disabled when the Sync mode is set to Host, Plug, or UDT.
Phase	0	360	Degrees	Offsets the phase of the tremolo effect. This is especially useful when Tremellow is synchronized, as it lets you determine where the peak and nadir of the amplitude cycle occur, relative to the beat.

Name	*Min*	*Max*	*Units*	*Description*
Tempo/Sync	Free	UDT		Sets the tempo and synchronization mode. The pop-up menu lets you choose from four sync modes: • Free mode disables the tempo slider and lets you set the tremolo frequency in Hz with the Coarse and Fine Frequency parameters above. • Host mode sets the tremolo frequency to match the host tempo. • Plug mode sets the tremolo frequency to match the PluggoSync plug-in if it is loaded. • UDT (User-Defined Tempo) mode lets you set Tremellow's tempo with the Tempo slider. Note: Host mode is only available in VST 2.0 and MAS applications that support it.

Insights

- Try synchronizing Tremellow to a rhythmic source, such as a drum loop, and experiment with the Phase parameter to cut and expose different parts of the beat. Host sync mode is especially suited for this, since the amplitude cycles will be in sample-accurate sync with the material.

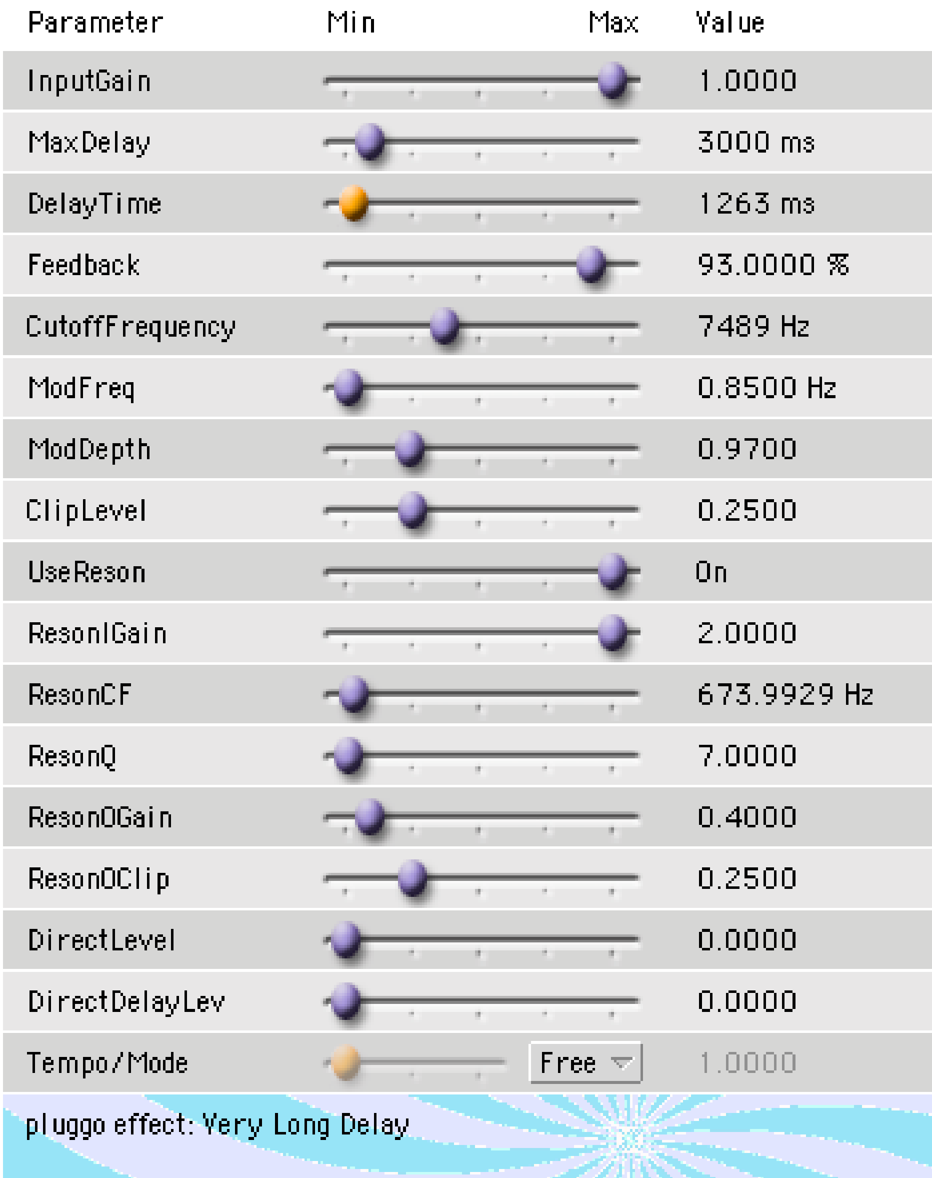
Very Long Delay

accepts sync | *category: delay*

audio input: mono | *cpu: light* | *audio output: mono*

What It Does

Very Long Delay can use a delay line of up to 30 seconds (if you have the memory). It has all the things you'd want in a delay, such as vibrato, a low-pass filter, and a resonant bandpass filter you can patch into the delay line. The bandpass filter mode is useful for creating background ambience, distortion, and feedback effects that are semi-related to the input material.



Very Long Delay

Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Input Gain	0	1		Sets the gain on the input to the delay.
MaxDelay	0	30000	ms	Sets the total available memory for the delay line. After changing this parameter, there is a short pause in the output as the newly allocated delay line is filled. Note that all the presets in Very Long Delay use a 3000 ms buffer, so there is not a pause when switching among them. If the delay line you attempt to allocate is too large for available memory, the effect will attempt to reallocate the delay line that existed before. In any event, you will not see an error message.
DelayTime	0	30000	ms	Sets the delay time. Note that the slider covers the full possible range, but only a subset that range will actually make sense if the MaxDelay parameter is set to a value less than the maximum possible delay time of 30 seconds. The delay time is set in milliseconds when the Sync Mode (shown in the Tempo/Sync parameter) is set to Free. In Host, Plug, and UDT modes, the delay time parameter is specified using note units and a multiplier.
Feedback	0	100	percent	Sets the percentage of the delayed signal that is fed back into the delay line.
CutoffFrequency	10	20000	Hz	Sets the cutoff frequency of a lowpass filter within the delay line's feedback loop.
ModFreq	0	50	Hz	Sets the frequency of delay time modulation.
ModDepth	0	4		Sets the amount of delay time modulation.
ClipLevel	0	1		Sets the amplitude above which values in the feedback delay line are clipped. Values less than 1 may produce distortion.
UseReson	Off	On		Sets whether the delay line includes a resonant bandpass filter.
ResonIGain	0	2		Input gain before the resonant bandpass filter. If either this value or the ResonOGain parameter is zero, there will be no feedback signal in the delay line.
ResonCF	10	18000	Hz	Sets the center frequency of the resonant bandpass filter.
ResonQ	0	500		Sets the resonance ("Q") of the bandpass filter. Q is defined as bandwidth divided by center frequency.
ResonOGain	0	4		Sets the output gain on the resonant bandpass filter.
ResonOClip	0	1		Sets the amplitude above which the output of the resonant bandpass filter is clipped before being fed back to the input of the delay line. Values less than 1 may produce distortion.
Direct Level	0	1		Sets the gain of the undelayed input signal.

Very Long Delay

accepts sync — *category: delay*

audio input: mono — *cpu: light* — *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
DirectDelayLev	0	1		Sets the gain of the output directly from the delay line. This provides an unprocessed but delayed version of the original signal to be included in the plug-in's output.
Delay Time Mult	0	10		Sets a multiple of the Delay Time Units currently selected that determines the delay time at the current tempo. For example, if the value of this parameter were 2 and the units were set to 1/4 (quarter note), the resulting delay time would be a half-note at the current tempo. This parameter is disabled when the Sync Mode is set to Free.
Delay Time Units	1	64t.		Sets the base note duration value used in determining the delay time in relation to the current tempo. This value is multiplied by the Delay Time to obtain the total beat value used to calculate the delay time. This parameter is disabled when the Sync Mode is set to Free.
Tempo	1	300	BPM	Tempo is displayed as a slider in the Tempo/Sync parameter. When the Sync mode is set to Host, Plug, or UDT, the Tempo parameter displays the current tempo used to determine the delay time. When in Host and Plug modes, the tempo is an indicator that cannot be changed--it is set by the host sequencer or PluggoSync. When in UDT mode, you can set the tempo to any desired value within its range and Very Long Delay will calculate the delay time based on the values of the Coarse Pan Freq Mult and Coarse Pan Freq Units parameters. This parameter is disabled when the Sync mode is set to Free. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Sync Mode	Free	UDT		Sync Mode is a pop-up menu in the Tempo/Sync parameter. There are four modes of sync available: • Free mode lets you set the delay time independent of the host sequencer. • Host mode synchronizes the delay time to the host tempo. • Plug mode synchronizes the delay time to the beat output of PluggoSync • UDT (User-Defined Tempo) mode is equivalent to Free mode but allows you to set the delay time in terms of tempo and note unit values. Note: Host mode is only available in VST 2.0 and MAS applications that support it.

Insights

- The extreme filtering and clipping possible in the delay line can produce self-sustaining feedback effects (although they are clearly outdistanced by the automation available in *Feedback Network*).

- *Very Long Delay* could be used as a live performance looping device, although you might want to customize your own such device using the tools available in MSP.
- You can calculate how much memory would be required for a delay line of a desired duration with the following formula:

  bytes of memory = duration (in seconds) times sampling-rate times 4

  As an example, a delay line of 5.7 seconds at a 44.1 kHz sampling rate would be 5.7 * 44100 * 4 = 1005480 bytes or approximately 982K bytes (divide by 1024 to get the size in K bytes). 982K is a little less than a megabyte.
- The ResonCF parameter of the *Background Cloud* preset would be an excellent choice for modulation by an LFO plug-in. By adjusting this parameter, you can modify the feedback ambience, leaving the foreground unchanged.
- The *Low Vague Munchkins* preset has two relatively irritating effects: a rapid vibrato and a heavily lowpassed filtered feedback. This demonstrates that you can do something to the more recent delays that eventually disappears into a formless mass after just a couple of cycles through the feedback loop.
- All of the presets have a Direct Level of 0; you may want to modify this depending on your application.

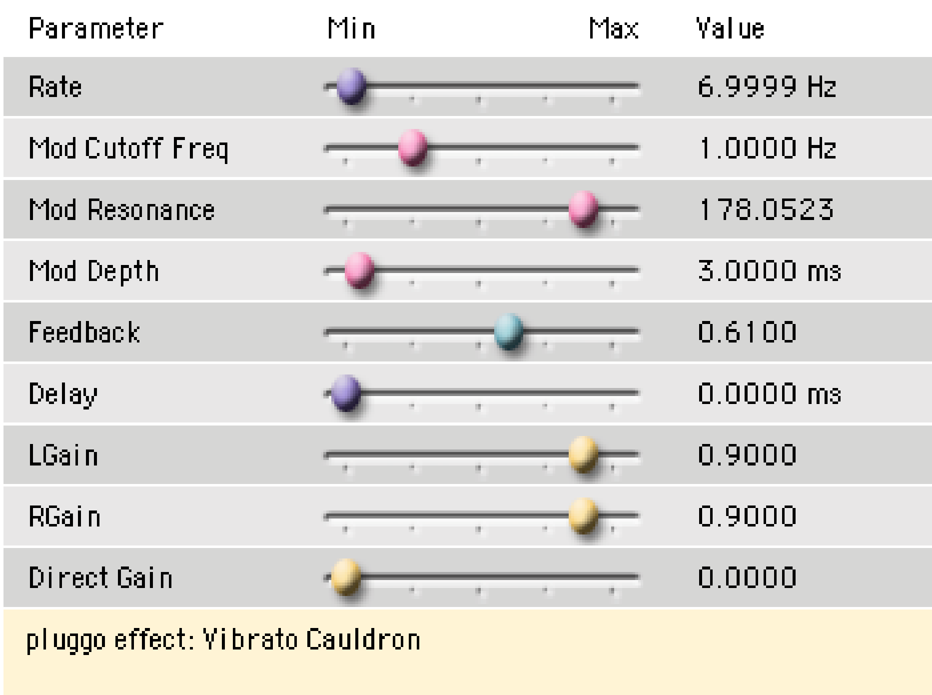
Vibrato Cauldron

category: pitch

audio input: mono | *cpu: light* | *audio output: stereo*

What It Does

Vibrato Cauldron consists of a pair of allpass filters modulated by a smoothed random process. This unusual combination creates a variety of pitch effects ranging from subtle to rather off-the-wall. The “bubbling” character of the pitch modulation found in many of the presets gives the effect its name. While *Vibrato Cauldron* has a subtle stereo character, it can also be used with a single output channel.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Rate	0	300	Hz	Sets the rate at which the noise source is sampled to create the random modulation signal. Higher sampling rates will tend to resemble modulation by white noise, while lower rates will introduce “stair step” or “sample-and-hold” changes.
Mod Cutoff Freq	0	4	Hz	Sets the cutoff frequency of a resonant lowpass filter applied to the random modulation signal. Yes, it’s quite a low frequency, required to achieve any type of smoothing of such a slow-moving signal.
Mod Resonance	0	199		Sets the amount of resonance (sharpness) of the lowpass filter applied to the random modulation signal. Higher values introduce more smoothing into the effect of the filter.
Mod Depth	0	50		Sets the amount of modulation of the allpass filters by the random modulation process.

Name	*Min*	*Max*	*Units*	*Description*
Feedback	0	.99		Sets the feedback of the allpass filters. Higher feedback levels have the effect of dissociating the filtered sound from the original and adding a mild comb-filter ambience.
Delay	0	90		Sets the delay time of the allpass filters. Higher values are more "reverberant" while lower values affect the pitch of the input signal.
LGain	0	1		Sets the gain on the left output.
RGain	0	1		Sets the gain on the right output.
Direct Gain	0	1		Sets the gain on the unprocessed input.

Insights

- The *Log In Throat* preset demonstrates the disturbing combination of an algorithmic vibrato and a relatively long delay time that blurs its effect.
- The *Repetitive Ocean* preset has an occasional outburst that exposes a difference in delay times between the left and right channels, pulling the sound to one side of your head or another. This shows that even with severe filtering, jumps in the noise process can sneak through.
- *Vocal Smoke* is an interesting preset when used in small doses on jazz guitar-type sounds. It has the effect of accentuating timbral differences from one note of a melody to another. In larger doses, it can make a vocalist sound like they need a throat operation.
- Change the Mod Resonance parameter in fine mode on the *Something is Wrong* preset to see the levels of subtlety you can get with more or less smoothing.
- The *Plucked Ambience* preset is a truly unexpected treat. High levels of feedback on the allpass filters can produce effects in which they seem completely off in their own space, having only the slightest connection to the input signal. That's certainly seems to be true here, yet there is a connection. Turn down the level of the input signal and the plucking will go away.

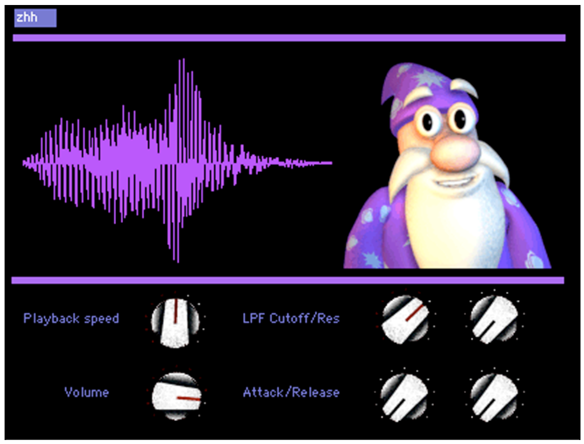
vocalese

category: synthesis

input: MIDI *cpu: light* *audio output: mono*

What It Does

vocalese is something a bit different – a phoneme-playback synth with a nifty 3-D image to tag along. Why have an image? Because we can, and because interesting interfaces can often help you create interesting results.



Visible Parameters

Name	*Min*	*Max*	*Description*
Playback Speed	0 (1x reversed)	127 (1x forward)	Sets the playback speed of the phoneme samples. Note that both positive and negative directions are supported. This control can be altered in real-time by using the Pitch Bend wheel.
Volume	0 (0 %)	127 (100%)	Adjusts the global level of the output signal.
Lowpass Filter Cutoff	0 (0 Hz)	127 (12.7 kHz)	Sets the cutoff frequency of a lowpass filter placed across the outputs of the sample playback engine. This control can be altered in real-time by using the Mod Wheel control.
Lowpass Filter Resonance	0 (0.0)	127 (0.99)	Sets the resonance value (sometimes called Q or pre-emphasis) of the filter. Resonance values range from 0.0 (no resonance) to 0.99 (nearly self-oscillation).
AR Attack	0 (0 ms)	99 (5000 ms)	When a new MIDI note is struck, the AR envelope starts running. The Attack rate determines the time for the amplitude to move from 0% to 100%.

Name	Min	Max	Description
AR Release	0 (0 ms)	99 (6000 ms)	When a MIDI note is released, the AR envelope begins its release stage. The Release rate determines the time for the amplitude to drop from 100% to 0%.

Insights

- The MIDI notes used to fire phonemes are:

36: 'eee'	43: 'uuu'	50: 'nng'	57: 'hee'	64: 'vvv'
37: 'ihh'	44: 'ooo'	51: 'ngg'	58: 'hoo'	65: 'zzz'
38: 'ehh'	45: 'aww'	52: 'fff'	59: 'hah'	66: 'thz'
39: 'aaa'	46: 'rrr'	53: 'sss'	60: 'bbb'	67: 'zhh'
40: 'ahh'	47: 'lll'	54: 'thh'	61: 'ddd'	
41: 'ohh'	48: 'mmm'	55: 'shh'	62: 'jjj'	
42: 'uhh'	49: 'nnn'	56: 'xxx'	63: 'ggg'	

- The vowel phonemes (eee – aww) are looped, but consonants are not. This is somewhat consistent with actual phoneme use, although you will find it difficult to properly articulate Valley Girl speech.
- Make sure you work the real-time controllers (pitch bend and mod wheel). These are hard-wired to the speed and filter controls, and can provide some crazed sounds when used misused.

Vocoder

category: filter

audio input: mono, pluggoBus | *cpu: heavy* | *audio output: mono*

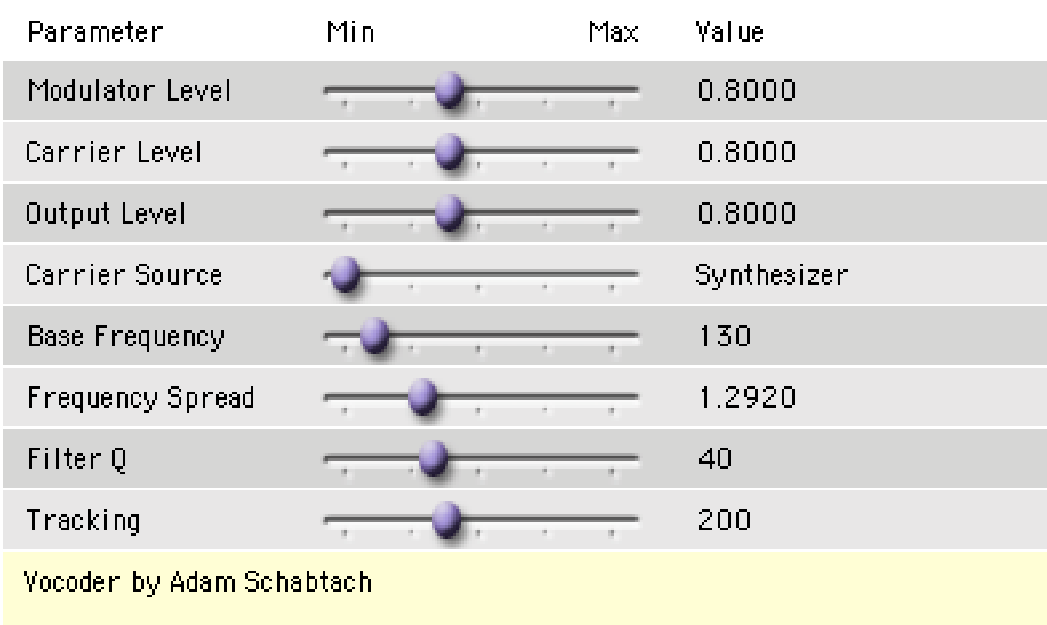
What It Does

Vocoder is a digital emulation of an analog vocoder. A vocoder consists of two banks of bandpass filters, connected in parallel. One signal, the *carrier*, is sent to one bank; another signal, the *modulator*, is sent to the second bank. The filter banks divide each signal into a number of frequency bands, one band per filter. The loudness of the frequency bands generated by the modulator signal is used to control the amplitude of the frequency bands of the carrier signal. The net effect is that the timbral and amplitude characteristics of the modulator are imparted upon the carrier.

Another way to think about vocoders is to imagine that you have a spectrum analyzer and a graphic equalizer somehow tied together, so that the bouncing level lights on the spectrum analyzer move the sliders up and down on the equalizer. The carrier signal goes through the equalizer, and the modulator signal goes into the analyzer.

The quality of vocoders is largely determined by the number of filters or frequency bands. A larger number of bands provide greater resolution of frequency, and hence more accurate combination of the carrier and modulator signals. There are two varieties of the *Vocoder* plug-in: a 16-band deluxe model, and a 10-band standard model. The 16-band model provides better performance, but places greater demands on your computer. Note that many hardware vocoders have 10 or fewer bands, so you may find that the standard model works fine for your needs.

Typically vocoders are used for talking-synthesizer effects, by using a voice for the modulator signal and a synthesizer as the carrier signal. *Vocoder* includes an internal synthesizer so that you can easily create this classic effect. You can also route any other signal to the carrier input via a PluggoBus connection.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Modulator Level	0	2		Sets the input level of the modulator signal. If you have a full-scale input signal, putting this control at about 1.0 is usually right. Move it higher if you have a weak input signal.

category: filter

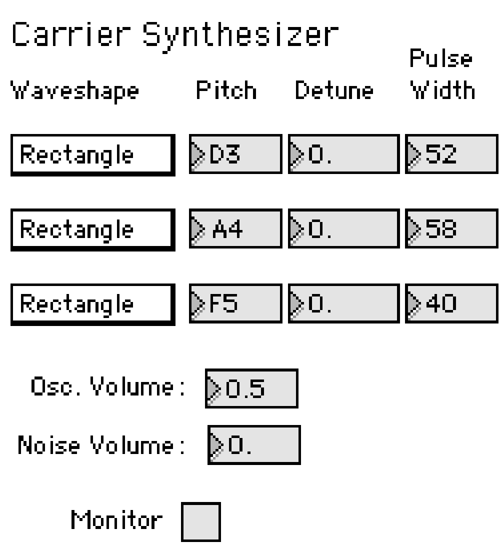
Vocoder

audio input: mono *cpu: heavy* *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
Carrier Level	0	2		Sets the input level of the carrier signal. See comments above re signal levels.
Output Level	0	2		Sets the gain of the carrier-bank filters, and hence the overall output level.
Carrier Source	Synthesizer	Pluggo Bus 4R		Selects the source of the carrier signal; either the internal synthesizer (see below) or one of the PluggoBus signals. (The input signal for the plug-in is always the modulator.)
Base Frequency	20	1000	Hz	Sets the frequency of the lowest-pitched filter in the filter banks.
Frequency Spread	1	2		Sets the ratio between frequencies of adjacent filters in the filter banks. A value of 1 means that all filters have the same frequency, which isn't particularly useful. A value of 2 means that the filters are tuned in octaves.
Filter Q	10	100		Sets the resonance ("Q") of the filters. Q is defined as bandwidth divided by center frequency. Low Q values will make the vocoding effect seem more diffuse; high values will make it seem more tight or accurate, but may also sound thin.
Tracking	10	500		Adjusts the rate at which the vocoder responds to changes in the modulator. Lower values produce a more accurate response, but may create audible artifacts.

Synthesizer View Parameters

The synthesizer consists of three fixed oscillators and a white-noise source. Each oscillator has independent waveshape, pitch, and pulse-width controls:



Name	*Min*	*Max*	*Units*	*Description*

Vocoder

category: filter

audio input: mono, pluggoBus *cpu: heavy* *audio output: mono*

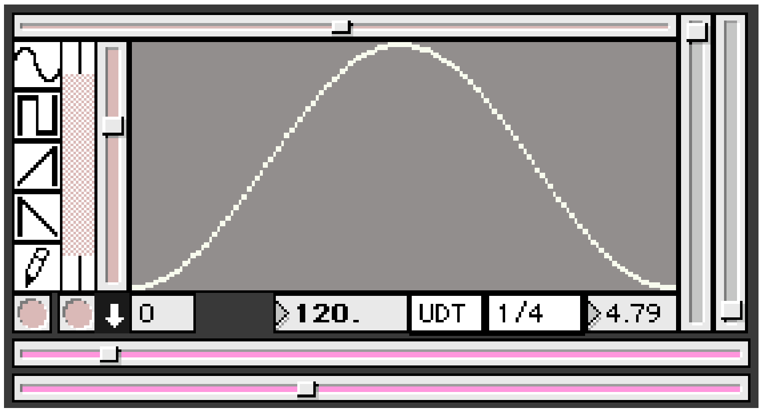
Name	*Min*	*Max*	*Units*	*Description*
Waveshape	Off	Rectangle		Sets the waveshapes of the oscillator to a sawtooth or rectangular wave, or turns the oscillator off altogether.
Pitch	C-2	E6	MIDI notes	Sets the pitch of the oscillator.
Detune	-10	+10	percent	Adjusts the pitch of the oscillator by a small amount. Detuning the oscillators can create a richer vocoding effect.
Pulse Width	1	99	percent	Sets the duty cycle, or pulse width, of the rectangle wave. A pulse width of other than 50% adds more harmonic content to the signal, which can increase the fullness of the vocoder's output. This control has no effect on the sawtooth wave.
Osc. Volume	0	1		Sets the overall level of the three oscillators. Lower this if you hear clipping.
Noise Volume	0	1		Sets the loudness of the white-noise source. Adding some noise to the carrier signal can improve the intelligibility of vocoded voices, since many consonant sounds consist largely of noise.
Monitor	(off)	(on)		Check this box to hear the carrier signal before it passes into the filter bank. Handy for tuning the other synthesizer parameters.

Insights

- Vocoders are, by nature, somewhat fussy devices. The results depend heavily on both the carrier and modulator signals and their resemblance to each other; and on the settings of all of the parameters. You will need to tweak all of the parameters to achieve the best results with any given pair of signals.
- The Base Frequency and Frequency Spread parameters are unique to *Vocoder*. The default values (130 Hz and 1.292, respectively) provided by the presets tune the filters to roughly match the frequencies of an expensive hardware vocoder of European origin. You might notice that changing the Frequency parameters creates some interesting sweeping effects. If you like these effects, try the *Resonation* plug-in. *Resonation* is sort of like half of a vocoder.
- The outputs from the carrier filters are sent alternately to the left and right output channels of the plug-in. This creates a synthetic stereo effect. If this effect isn't useful in your musical context, use the vocoder as a mono-in/mono-out plug-in.
- The vocoder was invented by a man named Homer Dudley. Yes, really.

What It Does

Warble changes the playback speed of its input signal, resulting in effects that range from subtle vibrato to record scratching emulation to complete mayhem. The playback speed is modulated by a built-in LFO module. The LFO can be retriggered in real time for envelope effects.



LFO Module Parameters

Name	*Min*	*Max*	*Units*	*Description*
Warble Range	0	1		Sets the overall scaling of the Warble effect. Lower settings produce subtle shifting of playback speed, while higher values result in more extreme changes.
LFO Waveform	Sine	Draw		Sets the current shape of the LFO. By clicking on the buttons to on the left side of the LFO module, you can select a Sine, Square, Saw Up, Saw Down, or User wave shape. When the User wave shape is selected, you can draw in the LFO Waveform display to set the LFO shape.
LFO Min	0	127		Sets the minimum value that the LFO will produce (when the LFO Waveform display shows its lowest value).
LFO Max	0	127		Sets the maximum value that the LFO will produce (when the LFO Waveform display shows its highest value).
LFO Phase	0	360	degrees	Offsets the start position of the LFO, in other words, where it starts in its waveform when it is retriggered. A value of 0 means the LFO starts at the beginning, and a value of 360 means it starts at the end. This parameter can be useful to employ when LFO modules are synchronized

Warble

accepts sync | *category: pitch*

audio input: mono | *cpu: light* | *audio output: mono*

Name	*Min*	*Max*	*Units*	*Description*
LFO Sync Source	UDT	Plug		This pop-up menu lets you select the synchronization source for Warbler's LFO. The pop-up menu lets you choose from three sync modes: • UDT (User-Defined Tempo) mode lets you set the LFO Tempo in BPM. • Host mode sets the LFO Tempo to match the host tempo. • Plug mode sets the LFO Tempo to match the PluggoSync plug-in if it is loaded. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
LFO Tempo	1	300	BPM	Sets the LFO tempo in beats per minute. This parameter, along with the Note and Multiplier parameters below, determines the rate of the LFO cycle.
LFO Sync Multiplier	0.01	75		Sets the multiplication factor that scales the LFO speed, as described below.
LFO Sync Note Value	1	1/64t		Sets the base note duration for a single LFO cycle. This note duration is multiplied by the LFO Sync Multiplier to set the LFO speed, according to the current LFO Tempo. For example, if you select a Sync Unit of 1/4 (a quarter note), and a Sync Multiplier of 1, the LFO will cycle once every quarter note. With an LFO Tempo of 120 BPM, this would be one cycle every 500 ms, or 2 Hz. If you change the Sync Multiplier to 2, the LFO will cycle once every two quarter notes: twice as slow. On the other hand, a Multiplier of 0.25 will make it four times faster - one cycle every sixteenth note.

Visible Warble Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the output gain of the processed signal.
Dry Level	0	1		Sets the output gain of the unprocessed signal.
Warble Range	0	1		Sets the degree to which the LFO affects the playback speed of the input signal.
Lag	0	511		Sets an amount of time over which changes in pitch are averaged. It's something like a portamento control.

Other Parameters

Note: A Warble parameter labeled "142. Warble Sync" will appear in any PlugMod menu when the Warble plug-in is loaded. Modulating this parameter will have no effect.

category: pitch	*accepts sync*	**Warble**
audio input: mono	*cpu: light*	*audio output: mono*

Insights

- We strongly recommend the use of *Warble* in conjunction with jazz recordings. But it works with anything. Depending on your definition of "works."

Warpoon

category: delay

audio input: stereo | *cpu: medium* | *audio output: stereo*

What It Does

Warpoon is an ambient chorus effect that can also create clusters of pitches around the original signal. It uses four stereo taps that can be individually modulated.

Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	2		Sets the level of the processed signal. 1 should be reasonably close to the level of the input signal.
Dry Level	0	1		Sets the output level of the unprocessed input signal.

Warpoon

audio input: stereo *cpu: medium* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Waveform	Sine	Triangle		Selects the waveform used to modulate the delay times of the taps. At certain combinations of parameter settings, the Triangle waveform will produce constant pitch shifts, while the Sine waveform will produce moving pitch shifts.
Tap Delay	0	100	ms	Sets the base delay time for the specified delay line tap. The modulation occurs around this time value.
Tap Mod Freq	0	2	Hz	Sets the frequency of the delay time modulation of the specified tap.
Tap Mod Depth	0	100		Sets the amount of delay time modulation of the specified tap. Typically, larger delay times require larger modulation depths to get a perceptible effect.

Insights

- Using the Triangle as the modulation waveform can create constant pitches that alternate being above and below the pitch of the original signal. However, the exact pitch offset from the original depends on two factors: the modulation frequency and the modulation depth. The *Dark Clusters* preset contains some settings that create this pitch shifting effect that you can use as a starting point.
- The *Mod Cut Out* preset demonstrates the effect of increasing the Mod Depth parameter above the Delay parameter for each tap. Since the delay time can't be negative (at least in the known universe), it is clipped to zero, resulting in short times in which the original signal makes a return among the chaos of munchkin voices.

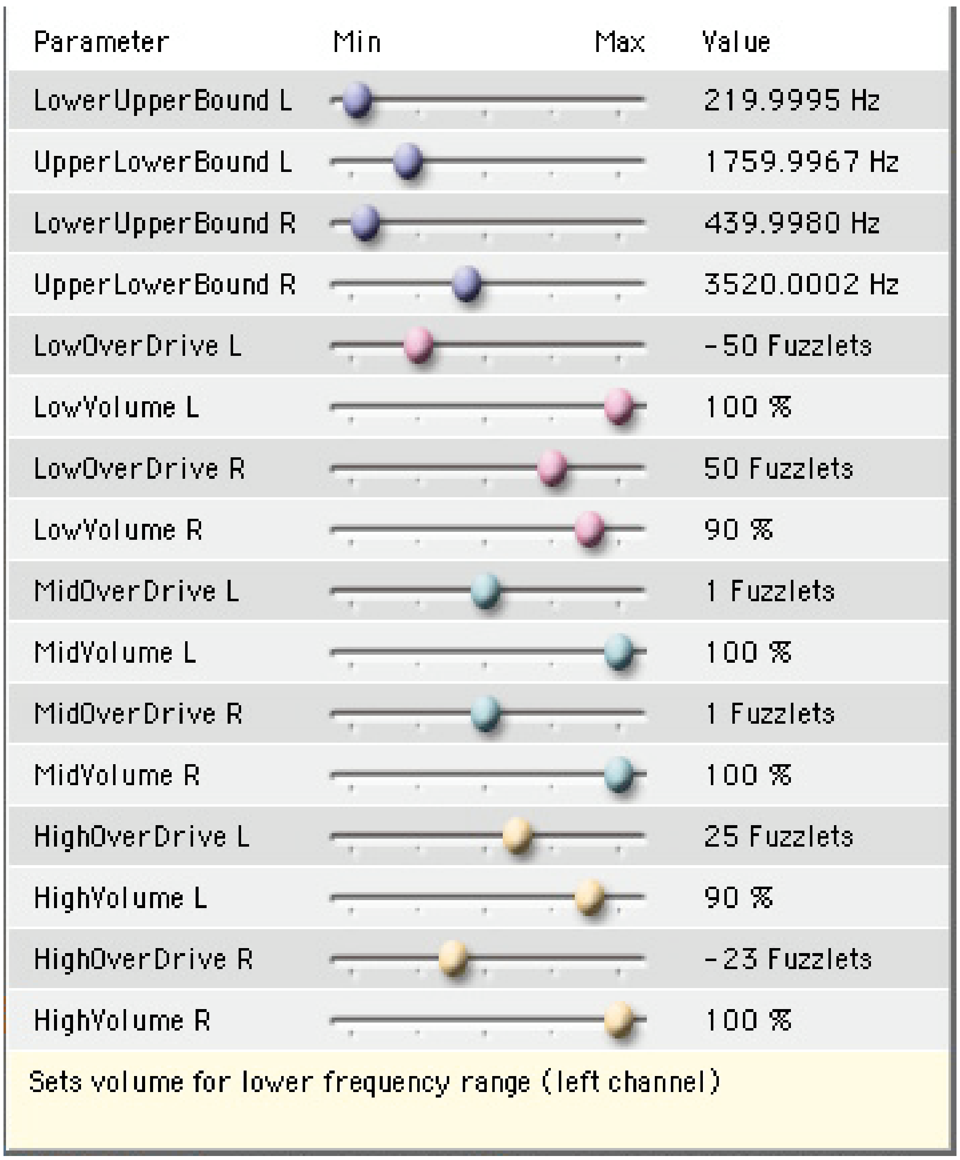
WasteBand

category: delay

audio input: stereo | *cpu: light* | *audio output: stereo*

What It Does

WasteBand splits each channel of your stereo file into three frequency ranges and allows you to mute, pass, or overdrive each band individually. You can use it to obliterate entire frequency ranges, create narrow bands of warm overdrive, or simply turn your entire low end into a fizzy mess.



Visible Parameters

Name	Min	Max	Units	Description
LowerUpperBound L	40	8000	Hz	Sets the dividing frequency between the lower and middle ranges for the left channel. Any audio content to the input signal below the dividing frequency will be overdriven and have its amplitude set by the LowOverDrive L and LowVollume L parameters, and any audio content above the dividing frequency will use the MidOverDrive L and MidVolume L parameters.
UpperLowerBound L	40	8000	Hz	Sets the dividing frequency between the middle and high ranges for the left channel. Any audio content to the input signal below the dividing frequency will be overdriven and have its amplitude set by the MidOverDrive L and MidVolume L parameters, and any audio content above the dividing frequency will use the HighOverDrive L and HighVolume L parameters.
LowerUpperBound R	40	8000	Hz	Sets the dividing frequency between the lower and middle ranges for the right channel. Any audio content to the input signal below the dividing frequency will be overdriven and have its amplitude set by the LowOverDrive R and LowVollume R parameters, and any audio content above the dividing frequency will use the MidOverDrive R and MidVolume R parameters.
UpperLowerBound R	40	8000	Hz	Sets the dividing frequency between the middle and high ranges for the right channel. Any audio content to the input signal below the dividing frequency will be overdriven and have its amplitude set by the MidOverDrive R and MidVolume R parameters, and any audio content above the dividing frequency will use the HighOverDrive R and HighVolume R parameters.
LowOverDrive L	-100	100	Fuzzlets	Sets the amount of overdrive for left channel audio frequencies below the frequency set by the UpperLowerBound L parameter
LowVolume L	0	100	Percent	Sets the percentage amplitude of the left channel audio signal; in the frequency range below that set by the LowerUpperBound L parameter to pass through to the output. If overdrive is used, this parameter controls the amplitude of the overdriven signal.
LowOverDrive R	-100	100	Fuzzlets	Sets the amount of overdrive for right channel audio frequencies below the frequency set by the UpperLowerBound R parameter
LowVolume R	0	100	Percent	Sets the percentage amplitude of the right channel audio signal; in the frequency range below that set by the LowerUpperBound R parameter to pass through to the output. If overdrive is used, this parameter controls the amplitude of the overdriven signal.
MidOverDrive L	-100	100	Fuzzlets	Sets the amount of overdrive for left channel audio frequencies below the frequency set by the UpperLowerBound L parameter

WasteBand

category: delay

audio input: stereo | *cpu: light* | *audio output: stereo*

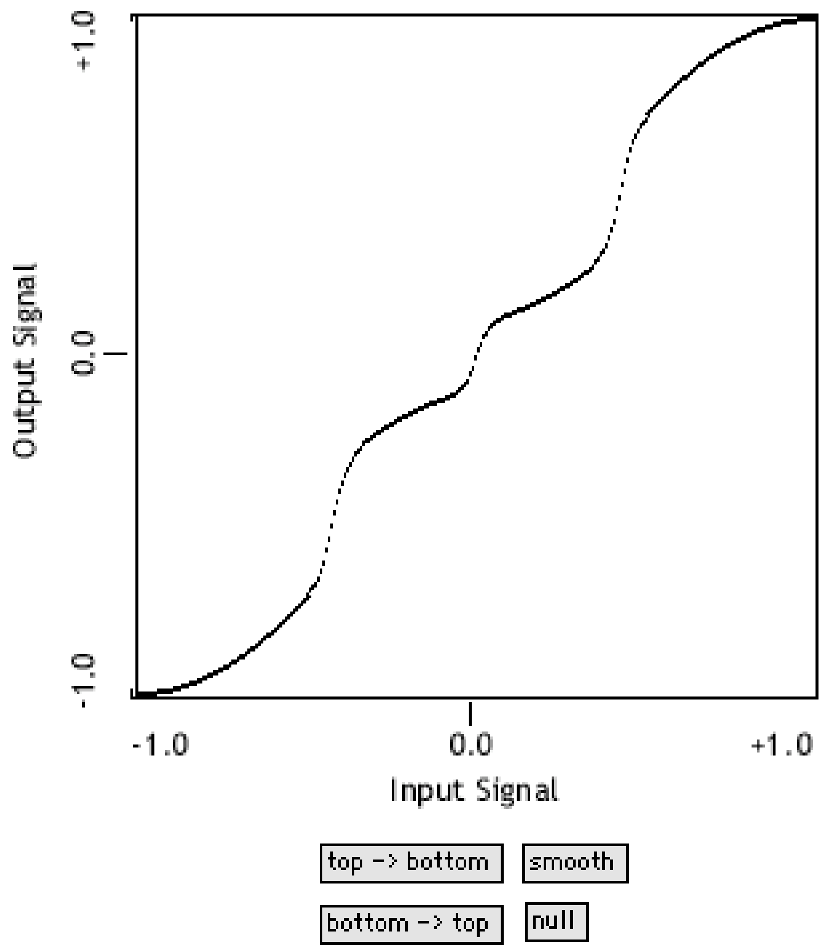
Name	*Min*	*Max*	*Units*	*Description*
MidVolume L	0	100	Percent	Sets the percentage amplitude of the left channel audio signal; in the frequency range between that set by the LowerUpperBound L and UpperLowerBound L parameters to pass through to the output. If overdrive is used, this parameter controls the amplitude of the overdriven signal.
MidOverDrive R	-100	100	Fuzzlets	Sets the amount of overdrive for right channel audio frequencies below the frequency set by the UpperLowerBound L parameter
MidVolume R	0	100	Percent	Sets the percentage amplitude of the right channel audio signal; in the frequency between that set by the LowerUpperBound R and UpperLowerBound R parameters to pass through to the output. If overdrive is used, this parameter controls the amplitude of the overdriven signal.
HighOverDrive L	-100	100	Fuzzlets	Sets the amount of overdrive for left channel audio frequencies above the frequency set by the LowerUpperBound L parameter
HighVolume L	0	100	Percent	Sets the percentage amplitude of the audio signal; in the frequency range above that set by the LowerUpperBound L parameter to pass through to the output. If overdrive is used, this parameter controls the amplitude of the overdriven signal.
HighOverDrive R	-100	100	Fuzzlets	Sets the amount of overdrive for right channel audio frequencies above the frequency set by the LowerUpperBound R parameter
HighVolume R	0	100	Percent	Sets the percentage amplitude of the audio signal; in the frequency range above that set by the LowerUpperBound L parameter to pass through to the output. If overdrive is used, this parameter controls the amplitude of the overdriven signal.

Insights

- Negative fuzzlet values may produce extreme effects that may or may not be exactly what you're looking for.
- The extreme effects of *WasteBand* may be further enhanced by combining it with the host-synchronized gating and accenting capabilities of the *Xformer* plug-in.
- *WasteBand* is the first known plug-in to implement the recent ISO-established International Distortion Metric (IDM) unit known as the fuzzlet.

What It Does

Waveshaper transforms signals by using a *transfer curve* to map the input signal to a new signal. The transfer curve shows the relationship between the input signal, which is plotted on the horizontal axis, and the output signal, which is plotted on the vertical axis. Hence a diagonal line running from the lower left to the upper right, at a 45° angle, creates an output signal that is identical to the input. Any other curve will alter the signal in some manner.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Transfer curve	-1	1		The transfer curve is the square graph in the editing window. Click and drag in it to change the curve.

Interface Elements

- The **top->bottom** button copies the top half of the transfer curve to the bottom half.
- The **bottom->top** button copies the bottom half of the transfer curve to the top half.

- The **smooth** button averages each point on the curve with its neighbors, which smoothes out sharp changes in the curve.
- The **null** button sets the curve to a straight, diagonal line, which passes the signal through unchanged. Useful as a starting point.

Insights

- The **bottom->top** and **top->bottom** buttons are handy because typically you want the waveshaping function to be symmetric, i.e. affect the positive and negative sides of the signal equally.
- If the transfer curve has sharp corners or discontinuities, it will add substantial amounts of inharmonic content (i.e. noise) to the signal. Use the **smooth** button to remove corners and discontinuities—unless, of course, substantial amounts of inharmonic content are useful to you.
- Waveshaping is probably most useful when applied to monophonic signals with relatively simple harmonic content such as sine waves.
- The *Soft Clip* and *Hard Clip* presets illustrate one of the differences between transistor- and tube-based circuits. Transistor amplifiers abruptly truncate signals when they exceed the circuit's capacity, whereas tube amps provide a smooth transition into distortion as they overload.
- The Randomize All command in the parameter change pop-up menu will indeed randomize the entire waveshaping table, turning any signal into full-volume, quasi-pitched noise. Don't do this while monitoring with headphones.

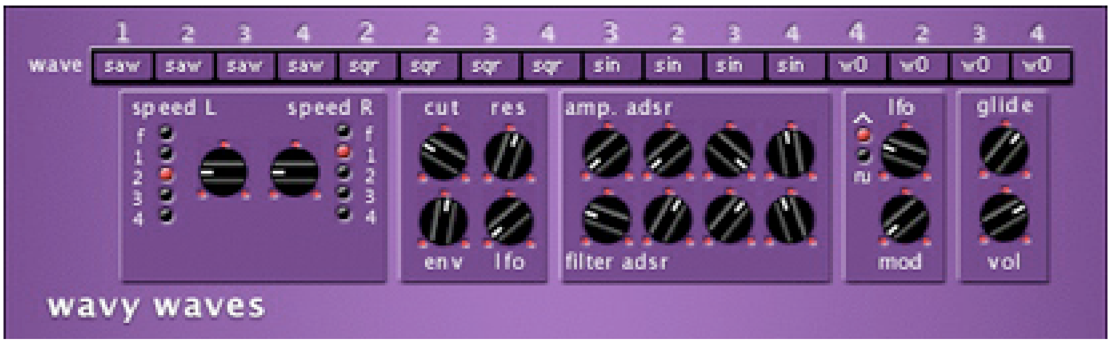
category: synthesis

wavy waves

input: MIDI | *cpu: medium/heavy* | *audio output: mono*

What It Does

wavy waves is a wave-sequencing synthesizer - it steps through a number of waveforms, either synced to a sequencer's clock, or in free-running mode.



Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Wave Sequence Steps	0 (saw)	33 (w27)	14-29 (1/1 – 4/4)	Sets the waveform used for each of the 16 wave steps. The options include sawtooth, square, triangle, pulse, sine and 28 digital waveforms (w0-w27).
Step Sync (Left)	0 (free)	4 (4x)	30	Determines the step sequence sync of the left-side oscillator. The *free* setting allows the step rate to free-run (at the rate setting speed), where 1/2/3/4 will multiply incoming sync rate by the selected rate.
Step Speed (Left)	0 (0 Hz)	127 (100 Hz)	31	If the left-side oscillator is set to free-run, the speed setting will determine the wave step rate.
Step Speed (Right)	0 (0 Hz)	127 (100 Hz)	32	If the right-side oscillator is set to free-run, the speed setting will determine the wave step rate.
Step Sync (Right)	0 (free)	4 (4x)	33	Determines the step sequence sync of the right-side oscillator. The *free* setting allows the step rate to free run (at the rate setting speed), where 1/2/3/4 will multiply incoming sync rate by the selected rate.
Filter Cutoff	0 (0 Hz)	127 (10 kHz)	35	Sets the cutoff frequency of the filter.
Filter Resonance	0 (0.0)	127 (1.0)	36	Sets the resonance value (sometimes called Q or pre-emphasis) of the filter. Resonance values range from 0.0 (no resonance) to 1.0 (self-oscillation).

Name	*Min*	*Max*	*CC#*	*Description*
Filter Envelope Depth	0 (0%)	127 (100%)	37	Sets the amount that the Filter Envelope modulates the filter cutoff.
Filter LFO Depth	0 (0%)	127 (100%)	38	Sets the amount that the LFO modulates the filter cutoff.
Amplitude Envelope Attack	0 (0 ms)	127 (5000 ms)	42	Sets the amplitude envelope attack rate. When a note is played on the synthesizer, the amplitude envelope is triggered, and the attack time determines how long it takes the amplitude to move from zero to its maximum value.
Amplitude Envelope Decay	0 (0 ms)	127 (4000 ms)	43	Sets the amplitude envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the amplitude to move to its sustain level.
Amplitude Envelope Sustain	0 (0%)	127 (100%)	44	Sets the amplitude envelope sustain level. The amplitude will remain at this level until the note is released.
Amplitude Envelope Release	0 (0 ms)	127 (6000 ms)	45	Sets the amplitude envelope release rate. This is the amount of time it takes for the amplitude to move back to zero after a note has been released.
Filter Envelope Attack	0 (0 ms)	127 (5000 ms)	46	Sets the filter envelope attack rate. When a note is played on the synthesizer, the filter envelope is triggered, and the attack time determines how long it takes the modulation output to move from zero to its maximum value.
Filter Envelope Decay	0 (0 ms)	127 (4000 ms)	47	Sets the filter envelope decay rate. After the envelope has reached its maximum value (based on the Attack parameter), the decay time determines how long it takes for the modulation output to move to its sustain level.
Filter Envelope Sustain	0 (0%)	127 (100%)	48	Sets the filter envelope sustain level. The modulation output will remain at this level until the note is released.
Filter Envelope Release	0 (0 ms)	127 (6000 ms)	49	Sets the filter envelope release rate. This is the amount of time it takes for the modulation output to move to zero after the note has been released.
LFO Shape	0 (tri)	2 (square)	40	Sets the waveform of the LFO.
LFO Speed (Rate)	0 (0 Hz)	127 (20 Hz)	41	Sets the rate of the LFO.
LFO Depth	0 (0%)	127 (100%)	34	Sets the amount of LFO modulation applied to the wave-sequencing step selection.

Name	*Min*	*Max*	*CC#*	*Description*
Glide	0	127	39	Sets the amount of time for the pitch to reach a MIDI note value.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer's pitch.
Mod Wheel Range	-100	100	percent	Sets the amount that the modulation wheel will affect the harmonic setting of the second voice.

Insights

- To get the whole picture of wavy waves, you need to be testing it in a sequencer with host sync capability. Start the sequencer, and all the presets will make perfect sense…
- The LFO Depth controls changes the wave stepping rate – continuously. The effect is at add a "hitch" to the wave sequence stepping. The preset *Jalopy* is a good example of LFO step rate modulation.

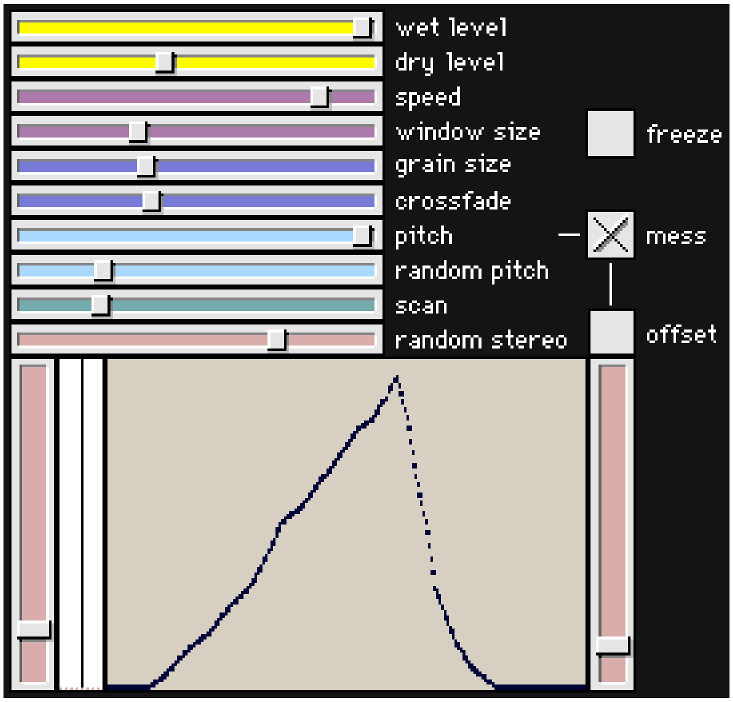
Wheat

category: granular

audio input: mono | *cpu: light* | *audio output: stereo*

What It Does

Wheat is part of the Pluggo Signature Series Granular Synthesizers. *Wheat* features the usual granular synthesis controls for grain speed, length, crossfade, pitch etc. However, it also includes a pitch envelope that can be applied to the grains. This additional capability can create effects that are even more extreme than you might expect from granular synthesis and the other PSSGS effects.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Wet Level	0	1		Sets the output level of the granular synthesis.
Dry Level	0	1		Sets the output gain of the original input signal.
Speed	0	512		Sets the rate at which grains are played.
Window Size	0	512		Determines the length of the audio loop that the grains play from. The plug-in's signal input is continuously recorded into this loop. The maximum window size is 1024 ms.
Grain Size	0	500	ms	Sets the length of the audio grains.
Grain Crossfade	0	500	ms	Sets the length of the crossfade between subsequent grains.
Pitch	-3	5		Sets the playback speed of the grains.
Pitch Random	0	4		Determines the amount of randomization applied to the playback speed of the grains.

Scan	0	512	Sets the rate at which the grains progress through the input audio loop.
Stereo	0	512	Adjusts the stereo image of the granular synthesis output.
ModSpeed	0	512	Sets the rate of pitch modulation.
ModMin	0	127	Sets the minimum pitch modulation.
ModMax	0	127	Sets the maximum pitch modulation.
ModAmount	0	512	Sets the amount of pitch modulation.
Freeze	Off	On	Stops audio input recording into the input loop, holding the current sound.
Mess	Off	On	Enables the pitch envelope modulation.
Offset	Off	On	Applies the modulation amount parameter.

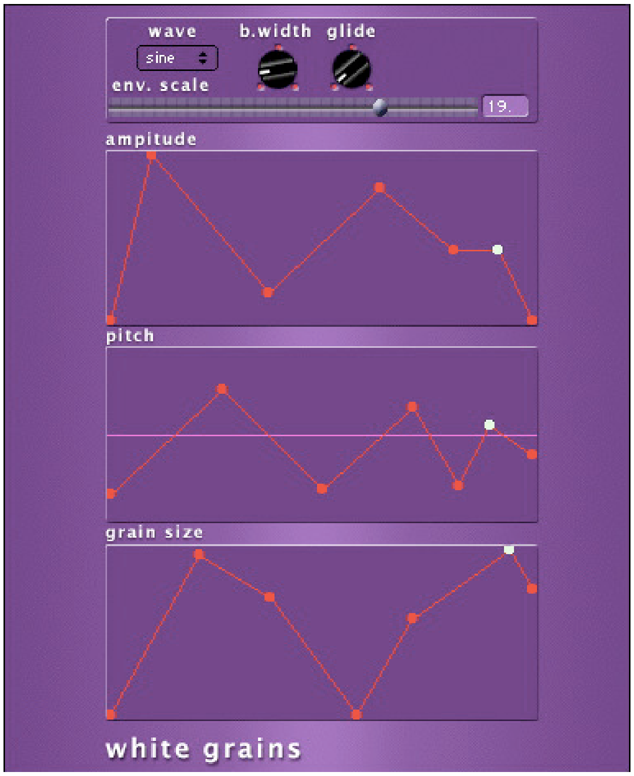
white grains

category: synthesis

input: MIDI *cpu: medium/heavy* *audio output: mono*

What It Does

white grains is a granular synthesizer, with individual seven-point envelopes for amplitude, pitch and grain size. Granular synthesis can provide deeply complex tones, and the variations provided through the envelopes range from simple robotic phrases to full whisk-broom wonder.



Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Waveform	0 (sine)	8	14	Sets the base waveform used for the granular process. The options include sine, triangle, square, saw, noise, pulse and three digital waveforms.
Bandwidth	0 (0 Hz)	127 (12.7 kHz)	15	Determines the base frequency of the granular process. As the rate increases, the waveform becomes more “random” sounding.
Glide	0 (2 ms)	127 (1272 ms)	16	Sets the amount of time to move from an existing pitch to a new one.

Name	*Min*	*Max*	*CC#*	*Description*
Envelope Scale	0 (0.0)	127 (25.4)	17	Sets the duration represented by the envelope editing displays. Since the envelope can be shorter than the full length of the display, the Envelope Scale setting provides a relative adjustment to envelope duration.
Amplitude Envelope			N/A	This seven-segment envelope editor provides on-screen manipulation of the amplitude output of the synth engine. The "white dot" represents the "sustain" level. The envelopes are not accessible with MIDI Controllers or external modifiers.
Pitch Envelope			N/A	This seven-segment envelope editor provides on-screen manipulation of the base pitch modulation of the incoming MIDI notes. The "white dot" represents the "sustain" position. The envelopes are not accessible with MIDI Controllers or external modifiers.
Grain Size Envelope			N/A	The seven-segment envelope editor provides on-screen manipulation of the grain size used by the granular voices. Each grain is modified in pitch and level in the creation of a new waveform. The "white dot" represents the "sustain" level. The envelopes are not accessible with MIDI Controllers or external modifiers.

Insights

- Very small grain sizes can provide a more traditional synth sound - an example would be Preset 2 ("What Computers Hear").
- Don't expect to get highly tonal sounds from the *white grains* synthesizer. The point of granular processing is to chop up a basic sound and create a complex waveform as a result. While not the place to find Moog emulations, *white grains* can make a variety of pad, percussive and glitchy sounds.

Xformer

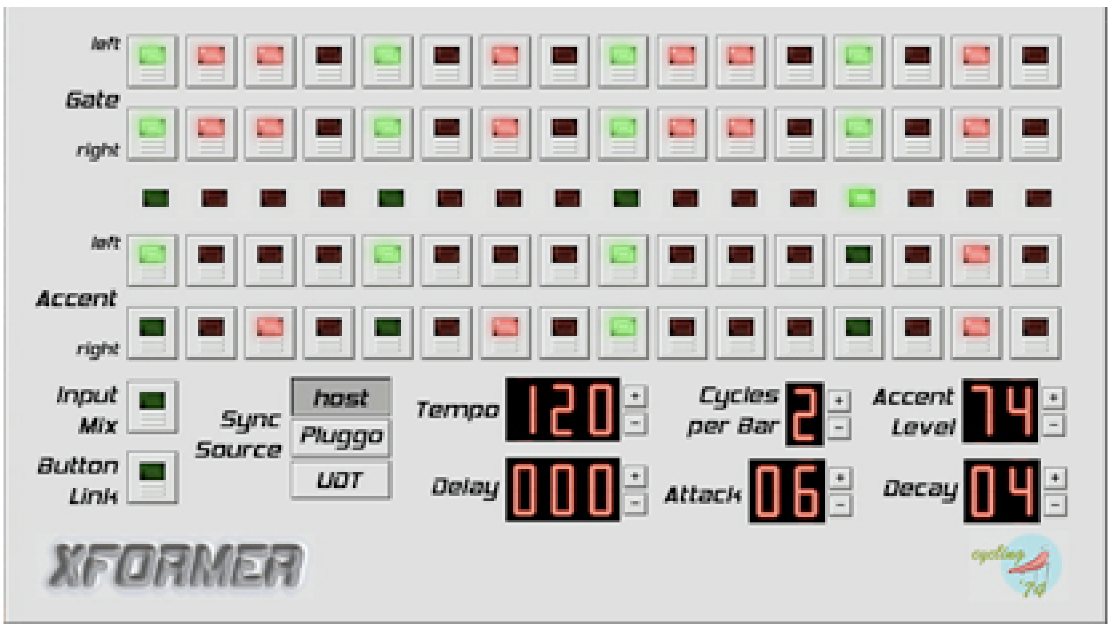
category: multichannel

input: mono/stereo *cpu: light* *audio output: stereo*

What It Does

Xformer is a groove-oriented plug-in that rhythmically mutes and accents your audio, transforming even a simple sine wave into a toe-tapping (or booty-shaking) riff. If your application supports host sync, *Xformer's* tempo can lock to your sequencer, providing perfectly synced effects. The delay parameter offsets *Xformer* from your sequencer's clock, allowing you to slide it to match the feel of your drum loops, bass lines, or bouzouki riffs. The attack and decay parameters let you smooth *Xformer's* extreme gating into softly rounded envelopes.

In addition, *Xformer* is a stereo plug-in. You can use the Input Mix button to send a mono signal to both channels, or combine the left and right channels of a stereo signal and build different patterns on the left and right button banks to create stereo panning effects.



Visible Parameters

Name	*Min*	*Max*	*Units*	*Description*
Gate left/right	Off	On		If the gate button is on for a given step, the input signal is passed to the output when *Xformer*'s internal sequencer reaches that step. Note that the left and right channels have independent gate controls.
Accent left/right	Off	On		If the accent button is on for a given step, and the gate button is also on for this step, the input signal is boosted by the amount set by the Accent control, and passed to the output when *Xformer*'s internal sequencer reaches that step. Note that the left and right channels have independent accent controls.

Name	*Min*	*Max*	*Units*	*Description*
Button Link	Off	On		If selected, the left and right channel settings for both signal gating and accent are linked; clicking on a gate or accent button in either channel will set the gate or accent for both channels.
Sync Source	Free	UDT		Three buttons let you choose the sync source for gating and accenting the input audio signal. There are three modes of sync: • Host mode synchronizes the gating and accenting to the host tempo. • Plug mode synchronizes the gating and accenting time to the beat output of PluggoSync • UDT (User-Defined Tempo) mode lets you set the tempo manually using the Tempo buttons. Note: Host mode is only available in VST 2.0 and MAS applications that support it.
Tempo	10	240	BPM	Tempo is displayed as an LED readout. If you have selected UDT mode as a sync source, clicking on the + or – buttons will raise or lover the tempo, or rate at which gating and accenting of the input signal occurs. This parameter is disabled when the Sync Source is set to Pluggo or host.
Delay	0	220		Delays the gating of the input signals relative to the Sync Source clock. Adjust this parameter to align *Xformer*'s gating with the beat of your music.
Cycles per Bar	1	9		Sets the number of times the Gate and Accent pattern you set will occur relative to the Tempo. Setting the Cycles per Bar to 2 will play through the gate and accent pattern at twice the speed
Attack	0	99		Sets an attack envelope for each gated portion of the input signal to be processed
Accent Level	0	99		If the accent button is on for a given step, and the gate button is also on for this step, the input signal is boosted by the amount set by the Accent control, and passed to the output when Xformer's internal sequencer reaches that step. Note that the left and right channels have independent accent controls.

Xformer

category: multichannel

input: mono/stereo *cpu: light* *audio output: stereo*

Name	*Min*	*Max*	*Units*	*Description*
Decay Level	0	99		Sets a decay envelope for each gated portion of the input signal to be processed. With a value of zero, *Xformer's* audio gates switch from on to off instantly. Higher values cause the gate to close more slowly.

Insights

- *Xformer* gets its name from the word transformer; "transforming" is the term for the effect popular with DJs and dance-music artists in which part or all of the music is turned off and on rapidly in sync with the song's rhythm. Now you can create this effect even if you have no innate sense of rhythm.
- If you are annoyed by slight clicks in Xformer1s output when its gates open and close, try increasing the Attack and Decay parameters from zero to 1 or 2.
- To create more exotic gated polyrhythms, bus your track to two separate channels and put an *Xformer* on each channel. Select only the first Gate position for each of the two effects, and use the Cycles per Bar setting to set the polyrhythm values (for example, 5 against 7). You can adjust the Decay values for each channel until they sound about the same.

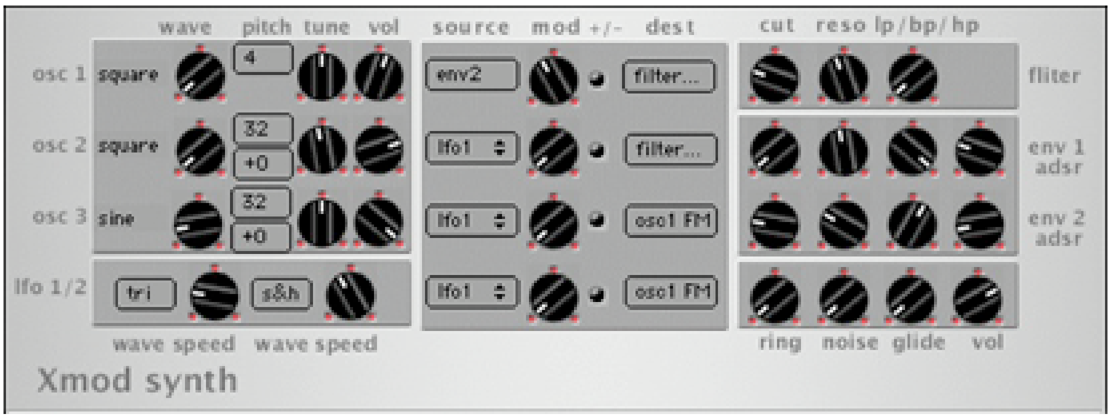
category: synthesis

Xmod synth

input: MIDI *cpu: medium/heavy* *audio output: mono*

What It Does

Xmod synth is analog-modeled synth with three oscillators, two LFOs, and a multi-mode filter and modulation matrix that gives this otherwise modest configuration its considerable expressive range.



Visible Parameters

Name	*Min*	*Max*	*CC#*	*Description*
Waveform	0 (saw)	32 (wav27)	14,18,22 (1,2,3)	Sets the waveforms used by the three oscillators. Options include sawtooth, square, triangle, pulse, sine and digital waveforms 0-27.
Oscillator Pitch	0 (32')	4 (2')	15,19,23 (1,2,3)	Sets the octave of each oscillator. Uses "organ-style" octave markings, where 32' is the lowest octave, and 2' is the highest.
Oscillator Pitch Offset (osc 2 & 3)	0 (-6 semi)	12 (+6 semi)	26,27 (2,3)	Adjusts the tuning offset of each oscillator. This adjustment is only available on the second and third oscillators.
Oscillator Fine Tuning	0 (-1 semi)	127 (+1 semi)	16,20,24 (1,2,3)	Sets the fine tuning of each oscillator.
Oscillator Volume	0 (0%)	127 (100%)	17,21,25 (1,2,3)	Sets the volume of each oscillator.
LFO Shape	0 (tri)	3 (s&h)	28,30 (1,2)	Sets the waveform of the two LFOs. Options include triangle, square, saw and s&h (sample & hold – a digital modulator based on a noise waveform).
LFO Rate	0 (.8 Hz)	127 (20 Hz)	29,31 (1,2)	Sets the LFO rate.

Xmod synth

category: synthesis

input: MIDI | *cpu: medium/heavy* | *audio output: mono*

Name	*Min*	*Max*	*CC#*	*Description*
Modulation Source	0 (lfo 1)	7 (noise)	46,50,54,58 (1,2,3,4)	Sets the source of the modifier for each of the four modulation matrices. Options include LFO 1/2, Envelope 1/2, Osc 1/2/3 and Noise.
Modulation Depth	0 (0%)	127 (100%)	47,51,55,59 (1,2,3,4)	Adjusts the amount of destination modulation by the selected source.
Modulation Direction	0 (off)	1 (on)	49,53,57,61 (1,2,3,4)	Determines whether the modulation amount will be positive or negative.
Modulation Destination	0 (osc 1 fm)	7 (filter fm)	48,52,56,60 (1,2,3,4)	Options include Osc 1 FM (Frequency Modulation)/PM (Pulse Width Modulation), Osc 2 FM/PM, Osc 3 FM/PM and Filter FM.
Filter Cutoff	0 (0 Hz)	127 (10 kHz)	41	Sets the cutoff frequency of the filter.
Filter Resonance	0 (0.0)	127 (1.0)	42	Sets the resonance value (sometimes called Q or pre-emphasis) of the filter. Resonance values range from 0.0 (no resonance) to 1.0 (self-oscillation).
Filter Type	0 (low-pass)	127 (high-pass)	43	Sets the filter type of the resonant filter. Rather than switching between the three types, this control continuously “morphs” from lowpass, through bandpass and into highpass.
Envelope 1 Attack	0 (0 ms)	127 (5000 ms)	32	Sets the first envelope attack rate. When a note is played on the synthesizer, the envelope is triggered, and the attack time determines how long it takes the output level to move from zero to its maximum value.
Envelope 1 Decay	0 (0 ms)	127 (4000 ms)	33	Sets the first envelope decay rate. After the envelope has reached its maximum value (according to the Attack parameter), the decay time determines how long it takes for the output level to move to its sustain level.
Envelope 1 Sustain	0 (0%)	127 (100%)	34	Sets the first envelope sustain level. The output will remain at this level until the note is released.
Envelope 1 Release	0 (0 ms)	127 (6000 ms)	35	Sets the first envelope release rate. This is the amount of time it takes for the output level to move back to zero after a note has been released.

Name	*Min*	*Max*	*CC#*	*Description*
Envelope 2 Attack	0 (0 ms)	127 (5000 ms)	36	Sets the second envelope attack rate. When a note is played on the synthesizer, the envelope is triggered, and the attack time determines how long it takes the modulation output to move from zero to its maximum value.
Envelope 2 Decay	0 (0 ms)	127 (4000 ms)	37	Sets the second envelope decay rate. After the envelope has reached its maximum value (based on the Attack parameter), the decay time determines how long it takes for the modulation output to move to its sustain level.
Envelope 2 Sustain	0 (0%)	127 (100%)	38	Sets the second envelope sustain level. The modulation output will remain at this level until the note is released.
Envelope 2 Release	0 (0 ms)	127 (6000 ms)	39	Sets the second envelope release rate. This is the amount of time it takes for the modulation output to move to zero after the note has been released.
Ring Modulation Depth	0 (0%)	127 (100%)	40	Sets the amount of ring modulation applied to the output of the synth voices.
Noise Mix Level	0 (0%)	127 (100%)	45	Adjusts the level of noise mixed into the output of the synth voice.
Glide	0 (2 ms)	127 (637 ms)	44	Sets the amount of time for the pitch to reach a MIDI note value.
Volume	0 (0%)	127 (100%)	7	Adjusts the global level of the output signal.

Global Parameters

Name	*Min*	*Max*	*Units*	*Description*
Pitch Bend Range	2	12	semitones	Adjusts the number of semitones that the MIDI pitchbend control will affect the synthesizer’s pitch.
Mod Wheel Range 1	-100	100	percent	Sets the amount that the modulation wheel will affect the depth of the first modulator.
Mod Wheel Range 2	-100	100	percent	Sets the amount that the modulation wheel will affect the depth of the second modulator.
Mod Wheel Range 3	-100	100	percent	Sets the amount that the modulation wheel will affect the depth of the third modulator.
Mod Wheel Range 4	-100	100	percent	Sets the amount that the modulation wheel will affect the depth of the fourth modulator.

Xmod synth

category: synthesis

input: MIDI *cpu: medium/heavy* *audio output: mono*

Insights

- With four-way modulation, you can create a highly-animated patch affected by the mod wheel.
- The modulation matrix (for which this synth is named) is most useful in creating modified oscillator sounds. Since the destinations exist for both oscillator frequency and pulse width, the oscillators can be modified to create complex (and often atonal) waveforms.
- Frequency modulation of a filter can have a variety of effects. When modulated at a slow rate, you will get the filter sweeps that are common in many types of music. However, if you use a faster modulation (say, from an oscillator instead of an LFO), you can get a wide range of gritty aliasing tones that are very different from FM oscillators.

Appendix A – Plug-in Summary

Plug-in	*Category*	*Input*	*Output*	*CPU*	*Sync*
additive heaven	synthesis	MIDI	mono	light/medium	
analogue drums	synthesis	MIDI	mono	light	
analogue percussion	synthesis	MIDI	mono	light	
Audio Rate Pan	multichannel	mono or stereo	stereo	light	host, plug
Audio2Control	modulator	mono	none	light	
Average Injector	delay, distortion	mono	mono	light	
bassline	synthesis	MIDI	mono	medium	
beatN	display	none	none	light	host
big ben bell	synthesis	MIDI	mono	light	
Breakpoints	volume, modulator	mono	mono, pluggoBus, none	light	host, plug
Center Channel	volume	stereo	mono	light	
Chamberverb	reverb	mono	stereo	medium	
Chorus x2	delay	stereo	stereo	light	
Comber	delay	stereo	stereo	light	
Control2Audio	signal generator	none	mono, pluggoBus	light	
Convolver	spectral domain	mono, stereo, pluggoBus	mono	heavy	
Cyclotron	filter	mono	mono	light	host, plug
deep bass	synthesis	MIDI	mono	light/medium	
Degrader	distortion	stereo	stereo	light	
D-Meter	visual display	stereo	stereo	light	
Dynamical	dynamics	mono	mono	medium	
easy sampler	synthesis	MIDI	mono	medium	
Env Follower	modulator	mono	mono	light	
Feedback Network	distortion	stereo	stereo	medium	
filtered drums	synthesis	MIDI	mono/stereo	light	
FilterTaps	filter/delay	mono	stereo	medium	
Flange-o-tron	delay	mono	mono	light	host, plug
flying waves	synthesis	MIDI	mono	light	
fm 4-op	synthesis	MIDI	mono	medium/heavy	
Fragulator	distortion	mono	mono	light	
Frequency Shift	distortion	mono	mono	light	
Generic Effect	delay	mono	mono	light	
Granular-to-Go	granular	mono	stereo	medium	
harmonic dreamz	synthesis	MIDI	mono	medium	
Harmonic Filter	filter	mono	stereo	medium - heavy (adjustable)	
HF Ring Mod	Distortion	mono, pluggoBus	Mono	light	
Jet	delay	mono	stereo	light	
Key Triggers	modulator	none	none	light	
KnaveStories	modulator	none	none	light	
Laverne	synthesis	MIDI	mono	light	
LFO	modulator	none	none	medium	host, plug
Light Organ	visual display	stereo	none	light	

Limi	reverb/dynamics	mono	mono	light	
lo-fi drums	synthesis	MIDI	mono/stereo	light	
Long Stereo Delay	delay	stereo	stereo	medium	host
M2M	modulator	MIDI	none	light	
Mangle Filter	distortion	mono	mono	light	
Monstercrunch	distortion	mono	mono	light	
Mouse Mod	modulator	none	none	light	
Moving Filters	filter	mono/stereo	stereo	light	host, plug
moving waves	synthesis	MIDI	mono	heavy	
Multi-Filter	filter	mono	mono	light	
Nebula	multichannel	stereo	stereo	light	
Noyzckippr	distortion	mono	mono	light	
OneByEight	multichannel	stereo	stereo	light	
Pendulum	granular	mono, stereo	stereo	light	
pgs-1	synthesis	MIDI	stereo	heavy	
Phase Scope	visual display	stereo	none	light	
Phase Shifter	filter	mono	mono	light	host, plug
Phone Filter	filter	mono	mono	light	
PluggoBus Rcv	audio routing	pluggoBus	mono, stereo	light	
PluggoBus Send	audio routing	mono, stereo	pluggoBus	light	
PluggoFuzz	distortion	mono	mono	light	
PluggoSync	synchronization	mono	thru	light	
PluggoSynth	wild card	MIDI	determined by plug-in	light	
PlugLogic	modulator	none	thru	light	
PlugLoop	filter/delay	mono, stereo	stereo	medium	host, plug
qsynth	synthesis	MIDI	mono	medium/heavy	
quick drums	synthesis	MIDI	mono/stereo	light	
Raindrops	filter/delay	mono	mono	light	
Randomizer	modulator	none	thru	light	
Resonation	filter/delay	mono	stereo	medium	
Resosweep	filter/delay	mono, stereo	stereo	medium	
Ring Modulator	distortion	mono, pluggoBus	mono	light	
Rough Reverb	reverb	stereo	stereo	medium	
Rye	granular	mono, stereo	stereo	medium	
shape synth	synthesis	MIDI	mono	medium/heavy	
Shepard Tones	synthesis	none	mono	light	
Shuffler	granular	mono	mono	Medium	host, plug
Sine Bank	synthesis	none	mono	medium	
Sizzle Delays	filter/delay	stereo	stereo	light	
Slice-n-Dice	granular	mono	mono	light	host, plug
Space Echo	delay	mono	mono	medium	host, plug
Spectral Filter	spectral domain	mono	mono	heavy	
Speed Shifter	delay	mono	stereo	light	
Squirrel Parade	granular	mono	stereo	medium	
Step Sequencer	modulator	none	thru	light	host, plug
Stereo Adjuster	multichannel	stereo	stereo	light	
Stereo Faker	multichannel	mono	stereo	light	

Stutterer	delay	mono, stereo	stereo	light	
Swirl	delay	mono	stereo	light	
Swish	filter	stereo	stereo	medium	host, plug
TapNet	delay	mono	stereo	medium	
Tapped Delay	delay	mono	stereo	medium	host, plug
Tremellow	multichannel	mono	stereo	light	host, plug
Very Long Delay	delay	mono	mono	light	host, plug
Vibrato Cauldron	pitch	mono	stereo	light	
vocalese	synth	MIDI	mono	light	
Vocoder	filter	mono, pluggoBus	mono	heavy	
Warble	pitch	mono	mono	light	host, plug
Warpoon	delay	stereo	stereo	medium	
WasteBand	filter	stereo	stereo	light	
Waveshaper	distortion	mono	mono	light	
wavy waves	synthesis	MIDI	mono	medium/heavy	
Wheat	granular	mono	stereo	light	
white grains	synthesis	MIDI	mono	medium/heavy	
Xformer	multichannel	mono/stereo	stereo	light	
Xmod synth	synthesis	MIDI	mono	medium/heavy	

Appendix B - Presets

Sometimes the creative process can use a little help, and you find yourself turning to unusual sources of inspiration. In the process of putting pluggo together, we discovered that the presets had some rather evocative names. We thought that you might like to shop for an effect by using these names rather than by plug-in name.

Following the modest truth that what delights the senses may be found rather than made, the following table contains a listing of every single preset for all plug-ins.

(In A) Big Country	xmod synth
...and another thing...	TapNet
1 kHz Modulator	Ring Modulator
1 Step Forward	Warble
1.5 Sec Buildup	Very Long Delay
1.5 Sec Buildup	LongStereoDelay
10 Seconds	Rough Reverb
11 Steps	Step Sequencer
11kHz	Degrader
12 bits	Degrader
120BPM 16ths	Tapped Delay
120BPM 32nds	Tapped Delay
120BPM Eighths	Tapped Delay
16 bits	Degrader
16 mm	xmod synth
20 Seconds	Rough Reverb
20 Thousand Leeks?	shape synth
200 MHz modulator	Ring Modulator
22kHz	Degrader
2400 Baud	Fragulator
24-hr Plumbing	harmonic dreamz
2x @ 2x speed	Slice-n-Dice
3 One Sec Loops	Stutterer
3 Slow 1 Fast	Shuffler
4 bits	Degrader
4 kHz Modulator	Ring Modulator
400 Hz Sub (sandwich)	additive heavens
4-note Self Oscillation	Cyclotron
4x Swung About	fm 4-op
5 Seconds	Rough Reverb
5.5kHz	Degrader
50's Design	Light Organ
8 bits	Degrader
A 440	Sine Bank
A Bit More Agressive	filtered drums
A Few Spikes	Randomizer
a kind of swing	Monstercrunch
A Lease Is Hot Rodded...	filtered drums
A Little Punchy	analogue percussion
A little sluggish	Pendulum
A Rink Factor	additive heavens
a sheen of rain	TapNet
A Splash of Vodka	white grains
A Steady Clip	Speed Shifter
Accent Pulse Tremolo	Xformer
Accordians & Organs	moving waves

According to Plan	Vibrato Cauldron
Acornaphobia	Squirrel Parade
Acoustic Guitar	Very Long Delay
Acoustic Guitar	LongStereoDelay
Across the Boards	analogue percussion
Active Bands	Swish
Added Low Junk	Very Long Delay
Added Low Junk	LongStereoDelay
Additive Straits	fm 4-op
Aftertaste	analogue percussion
Ahhh	Audio Rate Pan
Air of Tension	Chamberverb
Airy Ambience + Dry	Center Channel
Alien Abduction	moving waves
Alien Kittens	Vibrato Cauldron
Alien Squaredance	Speed Shifter
Alien Terrain	Randomizer
Alien Torch Song	Warpoon
Alienation	additive heavens
All .74!	Very Long Delay
All .74!	LongStereoDelay
All Over the Place	Space Echo
All Reverse	Stutterer
almost the same	TapNet
AM 2	Tremellow
AM 3	Tremellow
AM on Datsum 510	Resosweep
Ambient FB delay	Chamberverb
Amish Cribs	additive heavens
Amplitude Modulation	Tremellow
An Untidy Admission	harmonic dreamz
analogue drums	additive heavens
Anothr Cluster	Warpoon
Ant-Man	pgs-1
Aquaman	pgs-1
Arc Welder	shape synth
Armstrong's First Steps	additive heavens
As the Wizard Speaks...	vocalese
As You'd Expect	white grains
Ascending Loops	Fragulator
Asteroid Belt	qsynth
At the Ballpark	Harmonic Filter
At the Beach	Very Long Delay
At the Beach	LongStereoDelay
At-At Prep	qsynth
AtmoPad 1	easy sampler

AtmoPad 2	easy sampler
AtmoPad 3	easy sampler
Atom Ant	pgs-1
Attack by Sharpie	Resosweep
Audio	PluggoSync
Auto Comp	xmod synth
AutoBahnd	big ben bell
Auto-Feedback	Feedback Network
Automatik	Pendulum
Auto-Scratch	Warble
Back and Forth	Audio Rate Pan
Back Groove	deep bass
Backdraft	shape synth
Background Cloud	Very Long Delay
Background Cloud	LongStereoDelay
Back'n'Forth	moving waves
Backward	Slice-n-Dice
Backward Sludge	PlugLoop
Bad Aftertaste	fm 4-op
Bad Craziness	Feedback Network
Bad Machine Loop	easy sampler
Balaerophone	analogue percussion
Bam! Pow! Zap! Check?	Average Injector
Bandit (j.q.)	pgs-1
Bang Your Head	Stutterer
Banshee	Spectral Filter
Basic Filter Sweep	Cyclotron
Basic Flanger	Flange-o-tron
Basic Phaser	Phase Shifter
Bass Chorus	chorusX2
Bass Gates	Swish
Bass Phases	Swish
Bass Spasm	Space Echo
Bathtub Fun	Raindrops
Batman (movie)	pgs-1
Beautiful Breakup	Noyzckippr
Beep Beep	moving waves
Beginning the Run	xmod synth
Belly Full	additive heavens
Bi-Directional	xmod synth
Big Comb	Comber
Big Foot	bassline
Bing-Bong	big ben bell
Bipolar	Resonation
Bipolar	Sine Bank
Bipolar	Slice-n-Dice
Bipolar 2	Sine Bank
Bips	Spectral Filter
Birdman	pgs-1
Black Eyes	additive heavens
Black Sheep	Step Sequencer
Blind Gravity	additive heavens
Blind Terror	harmonic dreamz

Blips	Raindrops
Blotter Plotter	Light Organ
Blown Bottle	Laverne
Blown Suitcase	shape synth
Blurt	Noyzckippr
Boom Chick Parang	additive heavens
Boomerang	fm 4-op
Booming in the Back	Very Long Delay
Booming in the Back	LongStereoDelay
Boom-Tschick	additive heavens
Boop Pik Pom Pim	additive heavens
Bouncing Sizzles	Sizzle Delays
Boxer Rebellion	additive heavens
Breakout Loop	easy sampler
Brick Wall	Spectral Filter
Brighter Than 12	Noyzckippr
Brillo	Rough Reverb
Broken Flanger	Average Injector
Broken Vee	Waveshaper
Bubbles	Moving Filters
Bubbling Ambience	Vibrato Cauldron
Bubbling Over	Vibrato Cauldron
Bug Zapper	PluggoFuzz
Bug Zapper	additive heavens
Bulbous	Generic Effect
Busted Speaker	Mangle Filter
C maj	Vocoder 10-band
C maj	Vocoder 16-band
C min	Vocoder 10-band
C min	Vocoder 16-band
C'74??? C-64!!!	bassline
Calling All Monsters	shape synth
Calling All Monsters...	deep bass
Canard	Stereo Faker
Candy Crunch	Noyzckippr
Can't Get Started	Vibrato Cauldron
Captain Marvel	pgs-1
Car Show Proto	shape synth
Chaotic	Frequency Shift
Chasers	Moving Filters
Chatterbox	PlugLoop
Cheap Organism	xmod synth
Cheap Spring	Generic Effect
Cheetara	pgs-1
Chipmunk Perc	analogue percussion
Chipmunk Skips	PlugLoop
Chippy Bass	bassline
Choeps Deluxe	shape synth
Chopstick Chunks	harmonic dreamz
Choral	Pendulum
Chordifier	xmod synth
Chorus	Generic Effect
chorus dropout	TapNet

Appendix B - Presets

Chorus of Flappers	Average Injector
Chorus of Laughter	Warpoon
Chromalodics	deep bass
Chromatic Resonators	Resonation
Chuckee Cheeze	analogue percussion
Church Bells (MW)	big ben bell
Churchtops	additive heavens
Classic FM Organ	fm 4-op
Classic FM Piano	fm 4-op
Classic FM Steel Drum	fm 4-op
Classical Gas	easy sampler
Clock-sickle	shape synth
Coconuts (for the King)	additive heavens
Color Me Funky	fm 4-op
Comb	Tapped Delay
Commentary 7	Very Long Delay
Commentary 7	LongStereoDelay
Communion	fm 4-op
Complimentary Q	Cyclotron
Con Artist	Stereo Faker
Construction Site	analogue percussion
Convergence	Shuffler
Convergence	Step Sequencer
Cosmic Rotation	xmod synth
Could Lash Out	Vibrato Cauldron
Cowbell Loop 140bpm	easy sampler
Crampons	additive heavens
Crank the Q	Cyclotron
Creepy Crawly	Noyzckippr
Crossties	wavy waves
Crunchy Topping	WasteBand
Crystal Clear	wavy waves
Crystal Clone	Rye
Crystalline	Speed Shifter
Cubic Zirconium	Stereo Faker
Cuisinart	Pendulum
Cultural Exchange 96bpm	easy sampler
Current Mac Audio	Degrader
Current PC Audio :-)	Degrader
Daftly Speaking	xmod synth
dah dit dit	Xformer
Dark Cluster	Warpoon
Dark Murk	chorusX2
Darkwing Duck	pgs-1
Deathsquad Rally	Noyzckippr
Decaying Bub Choir	Average Injector
Dee Tune	big ben bell
DeeEx Whatever	fm 4-op
Default	Convolver
Default	PluggoBus Rcv
Default	PluggoBus Send
Default	Rough Reverb
Default	Space Echo
Default LFO	LFO
Default Settings	Env Follower
Default Values	Vocoder 10-band
Default Values	Vocoder 16-band
DefCon Five	deep bass
Delays x2	chorusX2
Descending Loops	Fragulator
Different Stripes	WasteBand
Diffuse	Phase Shifter
Digital Brook	white grains
Digital Error #1	Slice-n-Dice
Digital Error #2	Slice-n-Dice
Digital Major	harmonic dreamz
Dinner Bell	additive heavens
Dinner Call	deep bass
Dinoboy	pgs-1
Dippity-Doo	white grains
Dirty Heads	Space Echo
Disoriented	Rye
Distant	Wheat
Distinct Possibility	moving waves
Distorted Flange	Generic Effect
Distorto	Wheat
Disturbed Context	fm 4-op
Disturbing	analogue percussion
dit dit dah	Xformer
Ditto-head	bassline
Don't Break the Tape	Space Echo
Doodles	Wheat
DoomDays Sale	deep bass
Double	Generic Effect
Double Band-Pass	Multi-Filter
Double Curve	Waveshaper
Double Highs	chorusX2
Double Notch	Multi-Filter
DoubleDip Filter	deep bass
Double-wide Chorus	chorusX2
Down	Flange-o-tron
Down	Resonation
down by 5's and 8's	TapNet
Down the Chute	Step Sequencer
Down the Drain	Chamberverb
Dr. Detuna	shape synth
Drag Race Case	analogue percussion
Draw Example	LFO
Dreamworks	additive heavens
Dribble Glass	bassline
Drop That Bass! Loop	easy sampler
Drum Loop Weed-Out	Dynamical
Drunken	Swirl
Drunken Sailor	xmod synth
Dual Orbitz	xmod synth
DuneScape	qsynth

Durge	Shuffler
Dust Storm	PluggoFuzz
Dusty	Rough Reverb
DX-100 Standard	easy sampler
EA Bass, Inc.	easy sampler
Early Mac Audio	Degrader
Early Release (Please)	white grains
Earth	qsynth
Ebb and Flow	Swish
EchoFlecks	Space Echo
edge of disaster	TapNet
EeeKayGee	deep bass
Elastic	Fragulator
Electrified	Sizzle Delays
Elevator Up	Harmonic Filter
Empty Bottle	Laverne
Empty Fullness	Resosweep
Empty Jazz Club	Warpoon
End of Broadcast	white grains
End of the Rope	Wheat
End of Transmission	xmod synth
Enhancement	Warpoon
Envelopes	Dynamical
Ess Haitch One Oh One	bassline
ethnic rolled r	Monstercrunch
Even More	Cyclotron
Even Wave Dance	wavy waves
Eventual Sputnik	fm 4-op
Eventually	xmod synth
Evil Genius	bassline
Exciter	Raindrops
Exciter 2	Raindrops
Expand & Contract	Tapped Delay
Extreme Prejudice	fm 4-op
Facade	Stereo Faker
Fade	Tapped Delay
Fade Out	Noyzckippr
Fake Stereoizer	Moving Filters
Fast and Jerky	Randomizer
Fast Polyrhythm	PlugLoop
Fast Skyline	Step Sequencer
Fast Stir	Nebula
Fast Twitch	Comber
Faster	Phase Shifter
Feedback Delay	Chamberverb
Feedback on Rye	shape synth
feedbackless taps	TapNet
Feel Every Bump	Env Follower
Ferris Weel	Frequency Shift
Fiasco 16	Chamberverb
Fifths	Vocoder 10-band
Fifths	Vocoder 16-band
Filmstrip Testtone	fm 4-op

Filter Echo	deep bass
Finger Walking	white grains
Finis le sommeil	Resosweep
Flange	Generic Effect
Flange	Swirl
Flapstack	shape synth
Flash Gordon (movie)	pgs-1
flat chorus	TapNet
Flatulent Destruction	Moving Filters
Flight of the Filter	deep bass
Flipped Crater	shape synth
Florida Fresh Squeezed	shape synth
FM Sunrise	fm 4-op
Foghorn Leghorn	deep bass
Folger's (in your cup)	big ben bell
Folger's Cuppa	additive heavens
Follow the Leader	Env Follower
For Your Convenience	PluggoBus Rcv
For Your Convenience	PluggoBus Send
Forgery	Stereo Faker
Forklift Fun	Laverne
Forth and Back	Slice-n-Dice
Forward	Slice-n-Dice
Four is Enough?	Shuffler
Fragulation Station	Squirrel Parade
Frah Ken Steen	wavy waves
Frayed Silk	Chamberverb
Free with subscription	Phone Filter
Freerun Sync	moving waves
Frenzied Lather	Vibrato Cauldron
FrightNight	harmonic dreamz
Froggy	Resonation
Froggy Backmasking	Slice-n-Dice
from the planet Xorx	Generic Effect
Full Rectify	Waveshaper
Funeral Parlor	harmonic dreamz
Funk-doobie	harmonic dreamz
Funkified Chicken	harmonic dreamz
Funky Pings	Vibrato Cauldron
Galactic Swirl	white grains
Gaslit Sign	moving waves
Gated Rumble	Chamberverb
Gated Walls	additive heavens
Geiger's Best	shape synth
Gentle Punch	Dynamical
Gentle Shifts	OneByEight
Gershwin In The Trees?	Squirrel Parade
Gettin' the Grease	deep bass
Ghost Rider	pgs-1
Ghostie Bass	easy sampler
Ghostie Treats	analogue percussion
Giddy-Yup	deep bass
Glitches	OneByEight

Appendix B - Presets

Glitchy Bass	easy sampler
Goddard's Best	qsynth
Go-Fight-Win!	deep bass
Grade B Martians	Ring Modulator
Gravel Pit	Granular-to-go
Gravel Quick	Granular-to-go
Gravity	Shuffler
Gravity	additive heavens
Green Hornet	pgs-1
Green Lantern	pgs-1
Grind	Fragulator
Groan to Smile	Resosweep
Groovy Clipping	Very Long Delay
Groovy Clipping	LongStereoDelay
Growling Strings	xmod synth
Growling Synth	easy sampler
Grumbling Guyz	deep bass
Guarded Optimism	ShepardTones
Hailstorm	Raindrops
Half Orig & Half Diff	Center Channel
Half Rectify	Waveshaper
Happy Buildup	Very Long Delay
Happy Buildup	LongStereoDelay
Hard Clip	Waveshaper
Hard Gate	Dynamical
Harley Set	analogue percussion
Harmonica 10	Warpoon
Harpsichord	harmonic dreamz
Harsh Tremolo	Noyzckippr
Hart Kore	analogue percussion
Hat Loop 105bpm	easy sampler
Hat Loop 90bpm	easy sampler
headpound sizzle	Monstercrunch
Heavenly Chord	easy sampler
Heavenly Doorbell	big ben bell
Heavy Necking	Laverne
Heavy Seas	Vibrato Cauldron
Heifer Bass	easy sampler
heliport	Monstercrunch
Helium	Fragulator
Hesitant Slapback	Vibrato Cauldron
High A octaves	Vocoder 10-band
High A octaves	Vocoder 16-band
High Activity	Very Long Delay
High Activity	LongStereoDelay
High Delays	Sizzle Delays
High Lonesome	xmod synth
High School Football	Average Injector
High Slices	Rye
Hight Beating	Sine Bank
Hi-Pass w/Separation	Multi-Filter
Hi-Q!	Resosweep
Hiss-N-Spit	Mangle Filter

Hoax	Stereo Faker
Hockey Puck #14	harmonic dreamz
Hold On Now...	white grains
Hollow Core Door	harmonic dreamz
Hollow Echo	harmonic dreamz
Hollow Roboto	fm 4-op
Hollow Tree Home	wavy waves
Hooves and Paws	analogue percussion
House Fantastique	bassline
Hovercraft	additive heavens
How Now...	shape synth
Human Torch	pgs-1
Hump	Tapped Delay
Humpty Dumpty	deep bass
I Don't Feel So Good	Granular-to-go
I Drank Windex	Vibrato Cauldron
I Hab a Code	Laverne
I-Beam Chord	big ben bell
Immortality B	additive heavens
Implication	moving waves
Implied Rhythm	analogue percussion
Important Announcement	Very Long Delay
Important Announcement	LongStereoDelay
Impostor	Stereo Faker
In Extremis	WasteBand
In Search Of...	deep bass
In your head	Pendulum
Indie LPFs	Multi-Filter
indra's net	TapNet
Industrial Closer Loop	easy sampler
Inharmonicity	Ring Modulator
Init (10)	Key Triggers
Init (10)	Mouse Mod
Init (8)	PlugLogic
insecticide	analogue percussion
Insectile	chorusX2
Insects	Moving Filters
Insinkorator	Granular-to-go
Insta-Type	Granular-to-go
Interference	Granular-to-go
Interference X4	Warpoon
Internal	PluggoSync
Intimate Fluid Space	Chamberverb
Introduction	shape synth
Ironman	pgs-1
Irregular Pan	Audio Rate Pan
Jackhammer	Shuffler
Jacob's Ladder	moving waves
Jalopy	wavy waves
Jay Oh Par Dee	fm 4-op
Jazz on the Titanic	Warpoon
JazzMaster D.	additive heavens
Jingle Bells	Xformer

Joker's Wild	deep bass
Jumping Beans	Harmonic Filter
Jupiter	qsynth
just 2 delays	TapNet
just 4 delays	TapNet
Just a Bad Idea	analogue percussion
Just a Few Drops	Raindrops
Just a Little Bit	Nebula
Just Hanging Around	harmonic dreamz
Just noise	Vocoder 10-band
Just noise	Vocoder 16-band
Just Off Center	Randomizer
Key Sorta Follow	deep bass
Kick Combo	analogue percussion
Kick Loop 115bpm	easy sampler
Kick Loop 75bpm	easy sampler
Kick-Snare-Hat-Hat	analogue percussion
Kinda Goofy	Flange-o-tron
Kinda Queasy	Flange-o-tron
Kitten Vibrato	Vibrato Cauldron
K-k-k-Ken!	Slice-n-Dice
Kneeling at the Wall	Resosweep
Knives in Wall	Resosweep
Komputor	Granular-to-go
Kraftbot	Fragulator
Krikkets	wavy waves
Kristin	vocalese
L No Mid R No Ends	WasteBand
Landing Party	moving waves
Large Bathroom	Generic Effect
large unpleasant chord	TapNet
Lazy Combs	Comber
Leading The Band	Squirrel Parade
Leaning Left	Resosweep
Left & Right Strum	Xformer
Left = Left – Right	Center Channel
Left Hand Consideration	bassline
Left Leg First	deep bass
Left Minus Left	Center Channel
Left Times Right	Ring Modulator
Left With Itself	Convolver
Leisurely	Swirl
Let Them Ring	moving waves
Lethargic	Nebula
Liberty Nebula	big ben bell
Light Speed	Granular-to-go
Lightning	Waveshaper
Likable Vibrato	Chamberverb
Like Butter...	moving waves
Linked Low-Pass	Multi-Filter
Lioness	wavy waves
Lite	Phase Shifter
Loaded With Power	PluggoFuzz

Local Call	Phone Filter
Local Weather	Light Organ
LockedGroove	Fragulator
Log in Throat	Vibrato Cauldron
Long Decay	Space Echo
Long Distance	Phone Filter
Loopify	PlugLoop
lopsided	TapNet
Lopsided Mirror Fizz	WasteBand
Lopsided Peaks	Randomizer
Lots of Stuff	Audio Rate Pan
lovely	Monstercrunch
Low and Slow	Env Follower
Low and Slow	Feedback Network
Low and Vague	Very Long Delay
Low and Vague	LongStereoDelay
Low Beating	Sine Bank
Low F octaves	Vocoder 10-band
Low F octaves	Vocoder 16-band
Low Pass	Spectral Filter
Low Reverse	Rye
Low Vague Munchkins	Very Long Delay
Low Vague Munchkins	LongStereoDelay
Lowexplosion	Very Long Delay
Lowexplosion	LongStereoDelay
Lush Faced	harmonic dreamz
Lymphonic	Swirl
Machine Gun Vibrato	Generic Effect
Madness	Step Sequencer
Magic Flute	Resosweep
Majestic Walk	bassline
Major Malfunction	additive heavens
Make It Stop	chorusX2
Mall-o-rama	additive heavens
Man - Or...	qsynth
Man from K.L.A.M.P.	shape synth
Mangle	Mangle Filter
Mangy Chorus	Rye
Mario Hits Barrel	big ben bell
Mars	qsynth
Mastique Impress	wavy waves
Matter or Not-Matter	qsynth
Max Whatshisname	Slice-n-Dice
Mayday! Mayday!	Mangle Filter
Mayflower Funk	harmonic dreamz
Me Worry?	moving waves
Mechanismo	Flange-o-tron
Medium Edge	Swirl
Mellow Tunnel	Chamberverb
Mellow Xorx	Generic Effect
Mellowtone	Sine Bank
Mercury	qsynth
Metal Filings	Fragulator

metal insects	TapNet
Metal Machine	Feedback Network
Metallic Glow	analogue percussion
Meteor Shower	Nebula
Metric	Stutterer
Mid Heavy	Spectral Filter
Midas Muffler	filtered drums
Middle Ground	Nebula
Miller Triller	Warpoon
Milwaukee Car Horn	Laverne
Mind of its Own	Generic Effect
Minor Seventh	xmod synth
Minor Shift	Frequency Shift
Mirror	Step Sequencer
Mister Roboto	xmod synth
Mistuned AM Radio	Generic Effect
Mixexpressed Anxiety	Average Injector
Mod Cut Out	Warpoon
Mod Wheel Mambo	shape synth
Mod Wheel Squeedgy	lofi drums
Modularity 1	filtered drums
Modularity 2	filtered drums
Modulated Modulator	Ring Modulator
Modulated Rubber	Chamberverb
Modulation Fisticuffs	shape synth
Monks and Stars	moving waves
mono delay	TapNet
Monorail Route	wavy waves
Moog-a-licious	bassline
Moom-Mick Mak	additive heavens
Moonrise	qsynth
More Arbitrary Settings	Env Follower
More Filings	Fragulator
more from Xorx	Generic Effect
More Mangle	Mangle Filter
More Nasality	Noyzckippr
More Stepped Filtering	Cyclotron
Morphology	fm 4-op
Most of the Bass	Center Channel
Mountains	Dynamical
Mountains	Waveshaper
Move Too Quick Loop	easy sampler
Movie Set	Warpoon
moving chorus	TapNet
Moving Filters	Moving Filters
Moving Nasality	Warpoon
Moving Walkway	Harmonic Filter
Mr. Fantastic	pgs-1
Mr. French & Buffie	wavy waves
Multiplication Tables	harmonic dreamz
Murky	Moving Filters
Murky	Speed Shifter
Murky 2	Speed Shifter

Mystery Excitement	Noyzckippr
Narrow Band	WasteBand
Nasal Chorus	Average Injector
Neptune	qsynth
Nervous 71	Audio Rate Pan
Nervous Laughter	Resosweep
Nervousness	Very Long Delay
Nervousness	LongStereoDelay
net of bugs	TapNet
net of jewels	TapNet
net of tremolo	TapNet
nicely	Monstercrunch
Niporto Kwah	Spectral Filter
No Bass	Center Channel
No Effect	Waveshaper
No Effect	Xformer
No Es Musica	additive heavens
No Explanation Necessary	bassline
no feedback	TapNet
No More Nuts	Squirrel Parade
No Shift	Frequency Shift
No Vertical Adjustment	easy sampler
Noise Mod	Noyzckippr
Noise Never Sleeps	shape synth
Noise Plus Ultra	analogue percussion
Noisy Neighbors	Resosweep
Nom De Plume	xmod synth
Non-Bass Squiggle	bassline
Nonsensical	fm 4-op
Non-sequitar	bassline
Norelco Transmission	harmonic dreamz
Not So Rough	PluggoFuzz
Number Crunch	Granular-to-go
Nyet Nein No	bassline
Oblique Digitalis	fm 4-op
Ocean Eleven	shape synth
Oct. + flatted 5th	Vocoder 10-band
Oct. + flatted 5th	Vocoder 16-band
Octave Drop	Fragulator
Octaves	PlugLoop
Octaves	Resonation
Odd Turn Of Events	wavy waves
Off In The Trees	Squirrel Parade
Off in the Weeds	Warble
Off-center Spindle	wavy waves
Oh Carmine	Laverne
Ohm Sweet Ohm	bassline
Okay Cellular	Phone Filter
Old Fuzzbottom	WasteBand
One Clowns Nose	wavy waves
One Second Echoes	Very Long Delay
One Second Echoes	LongStereoDelay
One Sparrow's Voice	additive heavens

One Time Too Many	harmonic dreamz
Original	Chamberverb
Oscillation	additive heavens
Out of Sync	Wheat
Out of the Box 909	lofi drums
Out of the Fog	fm 4-op
Out of Tune	Sine Bank
Outdoor Acid Trip	Very Long Delay
Outdoor Acid Trip	LongStereoDelay
Outdoor Summer Night	moving waves
Outside Windbells	additive heavens
Outta Steam	harmonic dreamz
over and under	TapNet
Over to the Left	Convolver
Over to the Right	Convolver
Overloaded Wah	Resosweep
Overseas	Phone Filter
Oxidize	Fragulator
Pad Orpheus	moving waves
Paint Can Lid	big ben bell
Pan via Filtering	Moving Filters
Pan Way	Moving Filters
Panning	Resonation
Pantonal	Harmonic Filter
Pantone	Pendulum
Paradox	harmonic dreamz
Parallelism	deep bass
Parallelism	xmod synth
Past Your Limit?	Very Long Delay
Past Your Limit?	LongStereoDelay
Penny Loafer	wavy waves
Peoow Shring	additive heavens
Pepto Bismoid	Frequency Shift
Percussive Xorx	Generic Effect
Peter Quincy Taggert	qsynth
Phantom 2040	pgs-1
Phase Invert	Waveshaper
Phasy Breakup	Average Injector
Photon Torpedoes	fm 4-op
Pickup Sticks	Light Organ
Ping Pong	Tapped Delay
Pinkerton Bass	bassline
pitch down	TapNet
pitch scrunch	TapNet
Planet X	qsynth
Plastic 24	Swirl
Plastic Man	pgs-1
Plink Central	additive heavens
Plodding	Speed Shifter
Plucked Ambience	Vibrato Cauldron
Plummeting	Fragulator
Plunky	big ben bell
Polite Twister	Average Injector

Popcorn	Harmonic Filter
Popcorn	lofi drums
Power Drain	Dynamical
Power Takeoff Shaft	wavy waves
ppppp-p-p--p--- Pan	Audio Rate Pan
prettily	Monstercrunch
Pretty Dinger	big ben bell
Pretty Slow	Swirl
Prime Directive	moving waves
Pro-Aliasing	bassline
Program (1)	Phase Scope
Program (6)	LFO
Program (64)	Stereo Adjuster
Program (8)	Audio2Control
Program (8)	Breakpoints
Propeller Signature	wavy waves
Pseudo	Stereo Faker
Pulcifer's Pad	moving waves
PulsAmbient	moving waves
Pulsar Twins	qsynth
Pulse	flying waves
Pulsing Bands	Swish
Punch 47	Audio Rate Pan
Punchy Zzzzz	Average Injector
Quad Filter Sweep	deep bass
Quantize	Waveshaper
Quasi Ring Mod	Audio Rate Pan
Queasy Pan	Audio Rate Pan
Question Marx	deep bass
Quick & Dirty 808	quick drums
Quick & DIrty 909	quick drums
Quick 707	quick drums
Quick 909	quick drums
Quick Echoes	Space Echo
Quick HR-16	quick drums
Quick Modular 1	quick drums
Quick Modular 2	quick drums
Radar Alert	OneByEight
Radar Dub	deep bass
Radar Sweep	moving waves
Raft-bound	shape synth
Ragged Tails	shape synth
Raindrops	Raindrops
Raining Nuts	Squirrel Parade
Ramp Down	Randomizer
Ramp Up	Randomizer
Random Acts	xmod synth
Random Walk	moving waves
Randomizer	Randomizer
Rattling	Rough Reverb
Ray's Organ	harmonic dreamz
Really High	Frequency Shift
Really High	Spectral Filter

Appendix B - Presets

Really Low	Spectral Filter
Really Slow	Phase Shifter
Really Sparse	Granular-to-go
receeding into	TapNet
Recess	additive heavens
Rectification (Halfway)	harmonic dreamz
Red Cardinal	harmonic dreamz
Redwood Hills	wavy waves
Repetitive Ocean	Vibrato Cauldron
Retlif Elgnam	Mangle Filter
Retrouvailles	Resosweep
Reversal	Speed Shifter
Reversal of Fortune	big ben bell
Reverse Echos	Rye
Reverse Matrix	analogue percussion
Reverse Tape Slices	moving waves
Reversed SinAire	analogue percussion
Rhodes	Tremellow
Rhythmic Insects	Sizzle Delays
Right Minus Right	Center Channel
Rings	Raindrops
Ringy	Tapped Delay
Ringy Steps	Flange-o-tron
rising pitch	TapNet
Rising Sun	deep bass
RiverSplash	wavy waves
Road Work	Granular-to-go
Robbie Has Somthing ...	harmonic dreamz
Robin (tv)	pgs-1
RoboCop	pgs-1
RoboDuck	Fragulator
Robot Scratch	Warble
Robotz Chat	analogue percussion
roken Pickup	additive heavens
Rolling Thunder Bass	easy sampler
Romantic Descent	harmonic dreamz
Rotational Aspect	moving waves
Rotational Axis	qsynth
Rounder	deep bass
Rounders	xmod synth
Roundy's Goodness	analogue percussion
Rubber Band Man	bassline
Rubber Room	Chamberverb
Rubber Stadium	Average Injector
Rudeness	Phase Shifter
RuffHouse	shape synth
Rumble	PluggoFuzz
RumbleBass	xmod synth
Sad Vibrato	Swirl
Sample & Hold	Cyclotron
Saturn	qsynth
Saturnalian	moving waves
Saw	flying waves
Sawdust Symphony	analogue percussion
Scattered	Shuffler
Scattered Waves	Feedback Network
Scoiattolo Futuristo	Squirrel Parade
Scraping	PlugLoop
Scratched Cornea	Light Organ
Scratchy Seizure	OneByEight
scrunch stretch	TapNet
Seasick	Pendulum
Seasickness Deluxe	deep bass
Secret Steps	additive heavens
Seizure	Light Organ
Self-Oscillating	Feedback Network
Sequenced Filtering	Cyclotron
Set On Stun	Comber
Set Phasers on Stun	Phase Shifter
Shaky Tail	bassline
Shamen Crunch Chord	easy sampler
Shanty Clear	moving waves
Shave and a Haircut	Harmonic Filter
Shimmer Change	fm 4-op
Shimmering Light	moving waves
Shirl	Laverne
Short Circuit	Granular-to-go
SinAire	analogue percussion
Sine	flying waves
Sine Spread	Sine Bank
Sines of the Times	analogue percussion
Single Band-Pass	Multi-Filter
Single High Shelf	Multi-Filter
Single Low Shelf	Multi-Filter
Sinusitis	harmonic dreamz
Sizzle	Sizzle Delays
Sizzlepuss	Vibrato Cauldron
Skip To My Loupes	wavy waves
Skipped Steps	deep bass
Skipping CDs	Granular-to-go
Skylab	qsynth
Slave to the Past	easy sampler
Slide Chord	xmod synth
Slide into Home	xmod synth
Slouch	Wheat
Slow Bad Trip	Audio Rate Pan
Slow Bendit	Vibrato Cauldron
Slow Complexity	Audio Rate Pan
Slow Freq Mod	Audio Rate Pan
Slow Notch	OneByEight
Slow Phases	Swirl
Slow Scratchinance	Vibrato Cauldron
Slow Shifter	Rye
Slow Skyline	Step Sequencer
Slow Stir	Nebula
Slow Trill	Warpoon

Slow Vibrato X	Vibrato Cauldron
Slow Warble	Warble
Sluggo	vocalese
Slurred Notes	Vibrato Cauldron
small net	TapNet
Smee-ap Froom	shape synth
Smooth on Square	xmod synth
Smooth Seizure	OneByEight
Snare Loop 135bpm	easy sampler
Snare Loop 80bpm	easy sampler
Snipe Hunt	analogue percussion
Snorkeling	Swirl
So useless	Swirl
Soaking Wet	Sizzle Delays
Soft Clip	Waveshaper
Soft Distortion	Noyzckippr
Soft FB Ambience	Very Long Delay
Soft FB Ambience	LongStereoDelay
Soft Something	Noyzckippr
Solitary Confinement	moving waves
Some Movement	Center Channel
Some Short Echoes	Average Injector
Someone Will Use It...	shape synth
Something is Wrong	Vibrato Cauldron
Somonex	additive heavens
Sore Throat	wavy waves
Sorta Hoppy Loop	easy sampler
Space Delay	Comber
Space fill	Monstercrunch
Space Ghost	pgs-1
Spacicle	bassline
Spare	Laverne
Sparse Loops	Granular-to-go
Spawn	pgs-1
Spectral Flavoring	bassline
Spectrum 1	flying waves
Spectrum 2	flying waves
Spectrum 3	flying waves
Speech Teacher	moving waves
Speed Racer	lofi drums
Speed Skaters	wavy waves
Speeder Escape	qsynth
Spiderman	pgs-1
Spiral	Nebula
Spirograph	OneByEight
Spittle Horn	fm 4-op
Spittoon Cartoon	wavy waves
Splatterpuss	Feedback Network
'Splosion	xmod synth
Sputnik	qsynth
Square	flying waves
Squawk	PluggoFuzz
Squeaky Cleaner	filtered drums

Squeeze Flange	Generic Effect
Squeezen Boxen	shape synth
Squidge	xmod synth
Squid's Ink	moving waves
Squiggles	Pendulum
Squirbulence	Squirrel Parade
Squirrel And Hold	Squirrel Parade
Squirrel Trouble	Squirrel Parade
Squirreley Squiggles	Squirrel Parade
Squirrels In Space	Squirrel Parade
Staccato Glide	bassline
Stage 2	qsynth
Starter Set	fm 4-op
Starter Stock	xmod synth
Starting Point	WasteBand
Steel Wool	Fragulator
Steely Span	big ben bell
Stepped Flanging	Flange-o-tron
Stepping	Shuffler
Stereo Shudder	Audio Rate Pan
Stereo Subtractor	Warpoon
Stirred Scoops	Swish
Stock 808	lofi drums
Stock 909	filtered drums
Stock Portfolio	analogue percussion
Stompin' On The Savoy	fm 4-op
Stones in Black Holes	qsynth
Stories	additive heavens
Straight 707	lofi drums
Strange Days Mix	quick drums
Strike	Laverne
Strobe	Light Organ
Strolling	Swirl
Strummer Joe	easy sampler
Stuck	Shuffler
Stutterer	Stutterer
Subtle Combs	Comber
Subtle Depression	Generic Effect
Subtle Exciter	Sizzle Delays
Subtle Panning	Xformer
Sudden Expander	Dynamical
Sunday Hangover	big ben bell
Superman (comix)	pgs-1
Supple and Subtle	Env Follower
Surface Impact	qsynth
Swaggart Picnic	Audio Rate Pan
Sweeper	Tremellow
Swell	Tapped Delay
Swell Flange	Flange-o-tron
Swish	Swish
Syncopation	Granular-to-go
Take 'em Down	bassline
Take the Cure (Please)	additive heavens

Take Us To Your Acorn	Squirrel Parade
Takeoff	Generic Effect
Talk To The Animals	bassline
Talking Drum Chorus	Chamberverb
Tape Detune	Speed Shifter
Tasty Waves	moving waves
Tea Kettle	white grains
Technics Power-Down	Warble
Telephone Bill	easy sampler
TelStar	xmod synth
tempo patter	TapNet
tempo pitches	TapNet
Tensile Strength	xmod synth
Terminator Too	wavy waves
Test Tube Baby	moving waves
That Cut-In-And-Out Effect	Warble
That Same Ol' Chord	fm 4-op
The Alarm	bassline
The Alderson Point	qsynth
The Amazing Chan	pgs-1
The Brain Bug	qsynth
The Brittle Edge	shape synth
The Daily Grind	moving waves
The Dog Star	qsynth
The Don Rickles	shape synth
The Flash	pgs-1
The Flywheel Effect	fm 4-op
The Incredible Hulk	pgs-1
The Invisible Boy	pgs-1
The Last Waltz	wavy waves
The Little Dipper	qsynth
The Long Run	lofi drums
The Long Sweep Home	bassline
The Meanest Average	harmonic dreamz
The Metal Prism	fm 4-op
The Newts Have It!	harmonic dreamz
The Nuthouse	Squirrel Parade
The Octopus	white grains
The Phasing Effect	Average Injector
The Silver Surfer	pgs-1
The Swarm	Harmonic Filter
The Warm Jets	shape synth
The Wind-up	wavy waves
The X-Men	deep bass
Thermian's Revenge	qsynth
Thin and Nasty	Noyzckippr
Thin Delay	chorusX2
Thin Soup	deep bass
Third Eye	Light Organ
this is not granular synthesis	TapNet
This Isn't Acid Jazz	Resosweep
Thrashing Machine	wavy waves
Three Dimensional	bassline

Thriftshop sounds	additive heavens
Thru the Floor	additive heavens
Tight	Stutterer
Time Pusher	Dynamical
Tiny Baby Rattle	Noyzckippr
Top & Tails	harmonic dreamz
Touch of Rasp	Audio Rate Pan
Toyland	analogue percussion
Track Larger Changes	Env Follower
Training Wheels	additive heavens
Transformer Rhythms	Swish
Transgalactic Radio	Frequency Shift
Transmission Problems	analogue percussion
Travelling	Swirl
Tremellow	Tremellow
Tremulous Tremolo	Ring Modulator
Triangle	flying waves
Triple Delay	PlugLoop
Triple Dip	white grains
Trippy Loop	easy sampler
Trmellow Cauldron	Tremellow
Trolly	additive heavens
Tuna Salad	Wheat
Tune	Stutterer
Tuned4Success	additive heavens
Tuning Standard	white grains
Turned Into a Cube	fm 4-op
TV on in the other room	Resosweep
Tweezers	shape synth
Tweezerverb	Vibrato Cauldron
TwentySomething	wavy waves
Twin Oars	xmod synth
Twitchy	Vibrato Cauldron
Two-Face	additive heavens
TwoStep	wavy waves
UFOlogy	white grains
Underdog	pgs-1
Unfortunate Settings	Audio Rate Pan
UniSomber	xmod synth
Unmanned Beacon	qsynth
Unnatural Echos	Shuffler
Unstable Friends	Average Injector
Unsubtle Panning	Xformer
Up	Flange-o-tron
Up	Resonation
Up WIth People	additive heavens
Up-n-Down	Stutterer
Upper Mod	Env Follower
Uranus	qsynth
User Control	OneByEight
Velocity Flutter	fm 4-op
Venus	qsynth
Venus de Pluggo	Waveshaper

Verb Overtones	harmonic dreamz
Verbish	Sizzle Delays
VerFas Scratch	Warble
verfas tremolo	TapNet
Vertical Blinds	wavy waves
Very Simple	Tremellow
Very Slow	Swirl
Vibrato	Comber
Vibrato	Generic Effect
Vocal Smoke	Vibrato Cauldron
Vocal Surprise	Noyzckippr
Voices in my Head	chorusX2
Voyager	qsynth
Wah Wah Echoes	Very Long Delay
Wah Wah Echoes	LongStereoDelay
Walk the Walk	deep bass
Walking Around	Swirl
Walks Like One	bassline
Warble	Frequency Shift
Warm In Winter	Squirrel Parade
Wash Behind Your Ears	Mangle Filter
Watery	Resonation
Watery Flange	Average Injector
Waverider Bass	wavy waves
Wavetable	Fragulator
Wax and Wane	Nebula
Waxless	Rough Reverb
We Love Glenda	Rye
Weed Whacker	wavy waves
What Computers Hear	white grains
What to do	Shuffler
What's the Score?	lofi drums
What's Wrong?	Granular-to-go
When Worlds Collide	vocalese
Where's the Love?	PluggoFuzz
Which Blair?	additive heavens
Whirlibirds	Comber
Whispers in my Head	chorusX2
Whistler's Mom	bassline
White Chorus	Generic Effect
White Shag Carpeting	Warpoon
Who Knows Where?	bassline
Whoom Thang	additive heavens
Why the Punch?	Chamberverb
Wider	Phase Shifter
Wild Chorus	Wheat
Wind Generator	fm 4-op
Windy Chorus	Chamberverb
Wined Up	white grains
Wino-Synth	analogue percussion
Wobbly Pins	bassline
Wonder Woman	pgs-1
Woodcock Drums	filtered drums
Wooly Mammoth	shape synth
Wooo Are You	big ben bell
Xformer Simple On/Off	Xformer
Xorx Fabric Softener	Generic Effect
Yeah Yah	Average Injector
Yet Another Option	harmonic dreamz
Yet More Mangling	Mangle Filter
Your Little Brother	Feedback Network
Your Shepard and Flock Here	ShepardTones
Zipper	PluggoFuzz
Zipper Toys	deep bass
zoob toobify	TapNet
Zzzzz	Swirl

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