The

pluggo

Plug-in Reference Guide for radiaL

Cycling '74
1186 Folsom Street
San Francisco, CA 94103 USA
tel (415) 621-5743 fax (415) 621-6563
info@cycling74.com www.cycling74.com

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Contents

Introduction

About This Manual

radiaL includes a selection of plug-ins from Cycling '74's pluggo collection—a baker's dozen of unique plug-ins you can use inside radiaL This shortened version of The *Pluggo Plug-in Reference Guide* provides an alphabetical description of all the included plug-ins.

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*.
- An *Insights* section that provides some hints about how the plug-in can be used and configured.

If you like these plug-ins, you may want to upgrade to the full pluggo collection—in addition to the plug-in you see here, there are another 75 or so plug-ins you can use with radiaL or any other VST, MAS, or RTAS host application. And pluggo 3 also includes over twenty virtual synths you can use with your other applications. For more information, visit the pluggo website (http://www.pluggo.com) for a special discount available to radiaL owners.

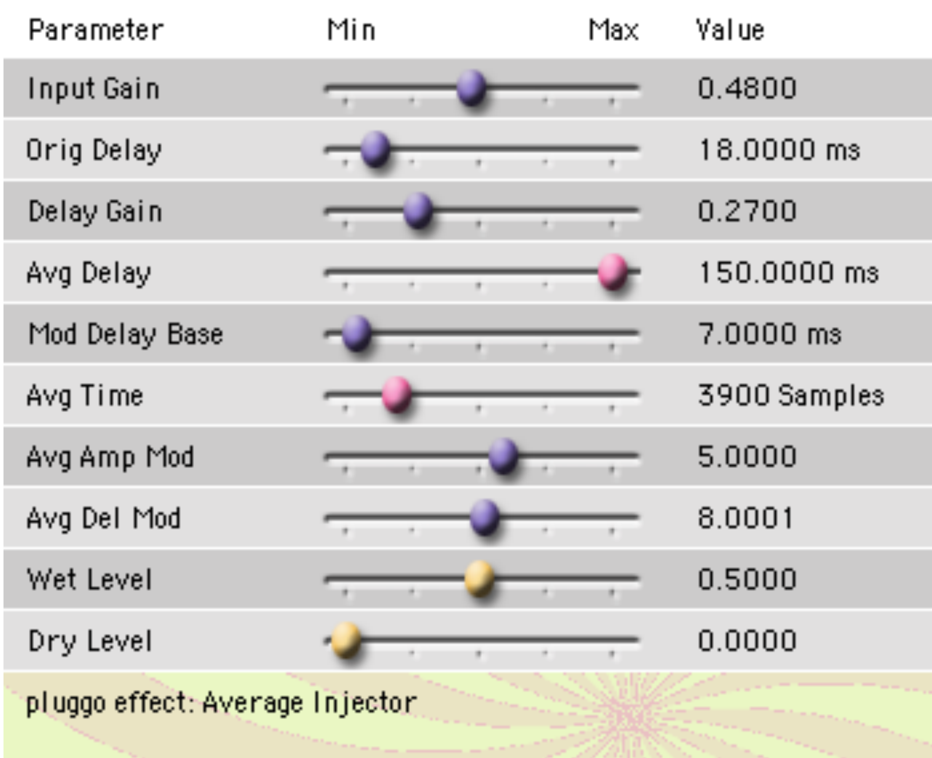
categories: delay, distortion

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.
- *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.

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.

Chamberverb

category: reverb

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

Visible Parameters

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.

Feedback Network

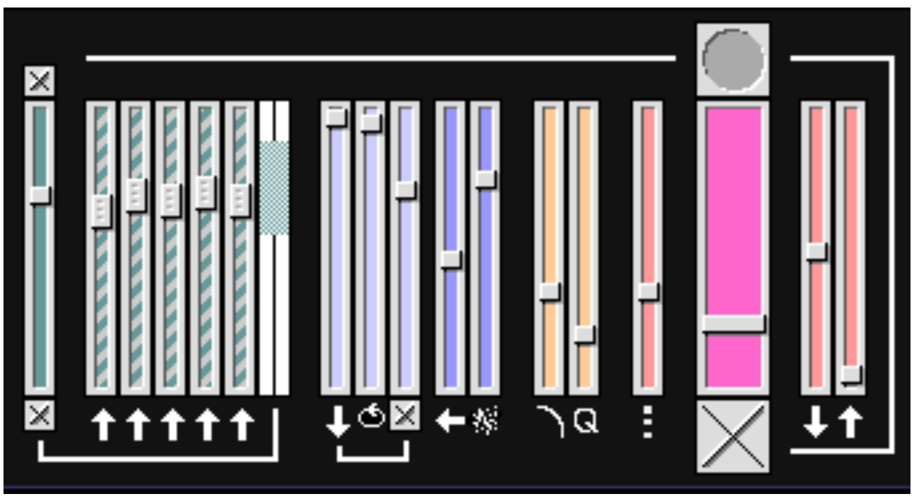
category: distortion

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

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.

category: distortion

Feedback Network

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

Name	*Min*	*Max*	*Units*	*Description*
Feedback Sensitivity	0	127		Sets the feedback volume at which the auto-feedback mechanism will start turning down the gain.
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.

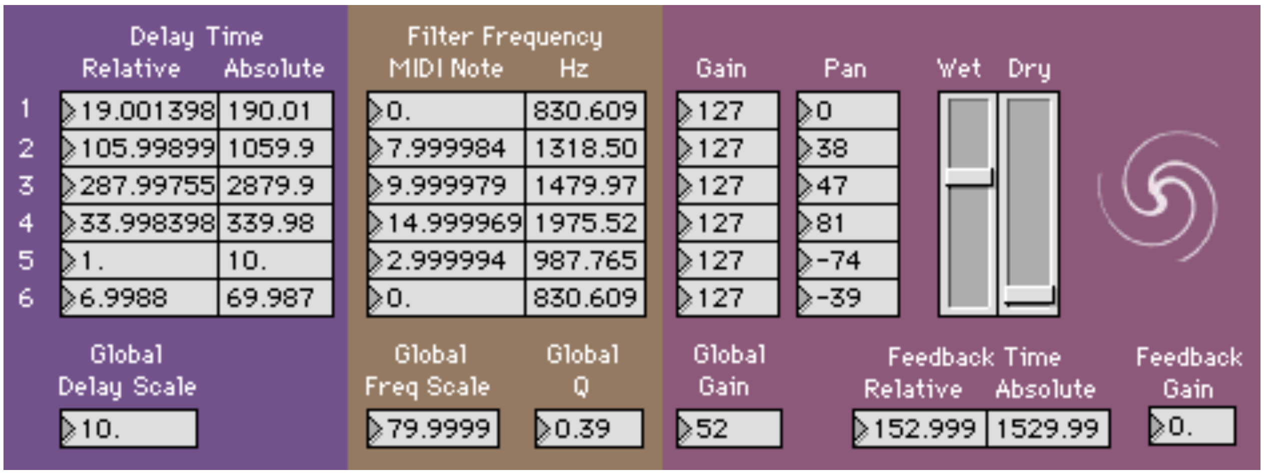
category: filter/delay

FilterTaps

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.

FilterTaps

category: filter/delay

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

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

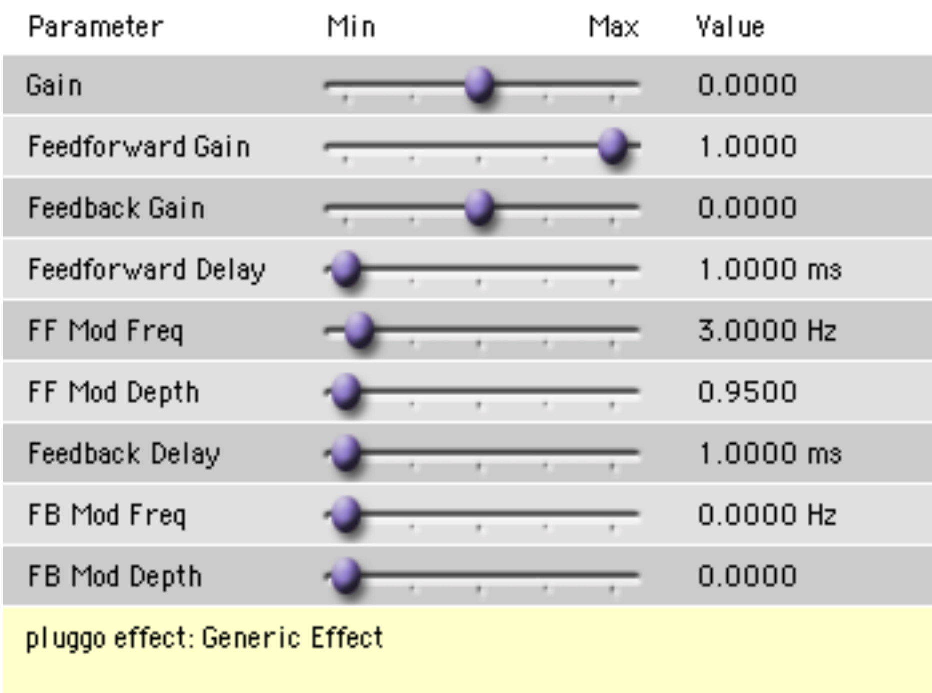
category: delay

Generic Effect

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

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.

Generic Effect

category: delay

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

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.

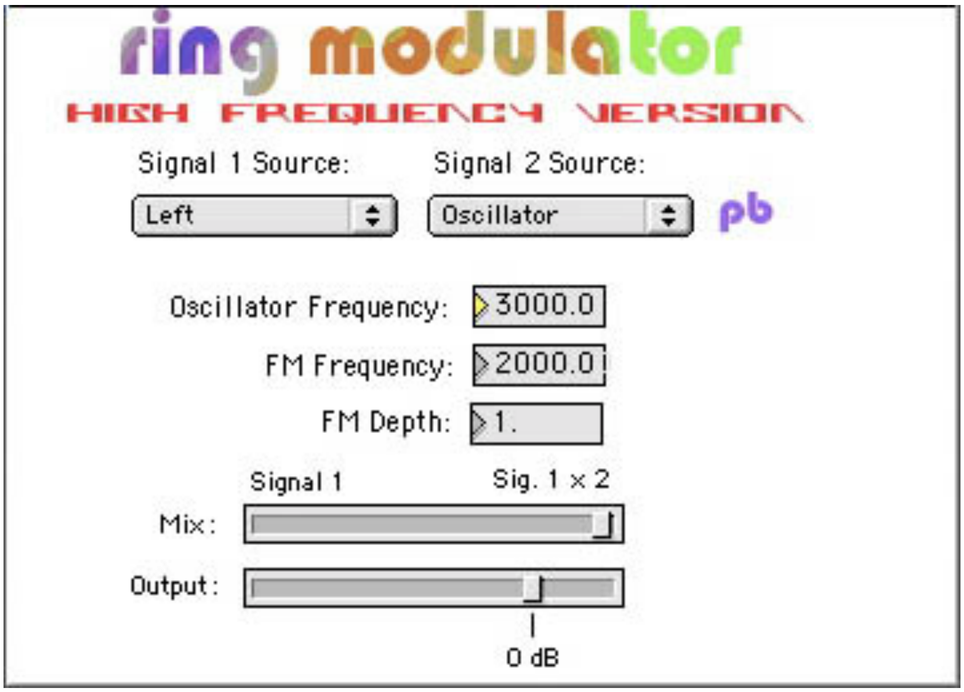
HF Ring Mod

category: distortion

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

What It Does

HF Ring Mod 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. The plug-in's oscillators have a wide 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 high range too, so the modulating signal can have a wider range of frequency components.



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.

category: *delay* | **Jet**

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

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 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

- 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.

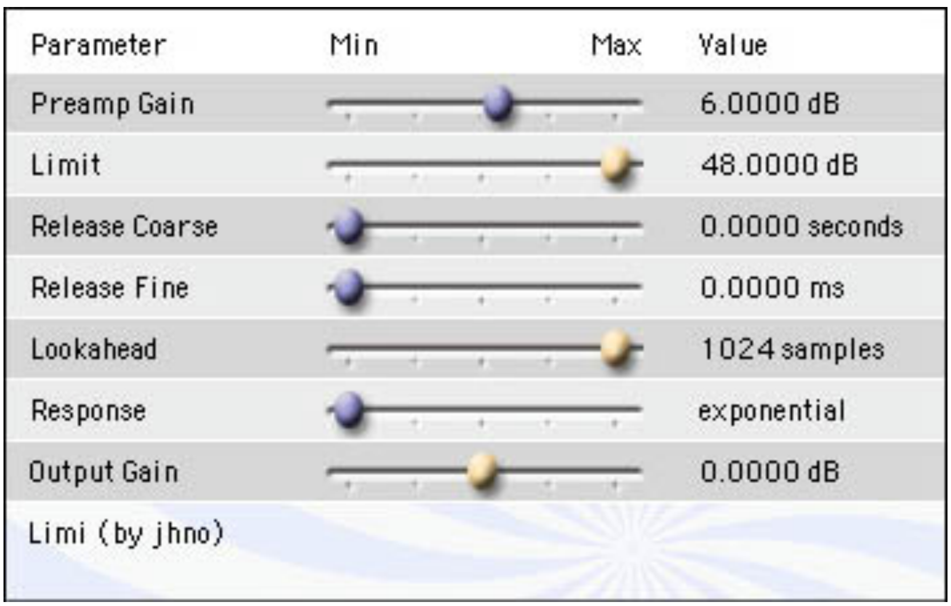
- To create classic flanging effects, set the Mix knob to its center position (12:00 o' clock) to mix the raw and delayed signals in equal amounts.

category: reverb/dynamics | Limi

audio input: mono | *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 millieseconds
Release Fine	1	1000	msec	Used in conjunction with the Coarse Recovery Time parameter to fine-tune the recovery time of the limiter in millieseconds
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.

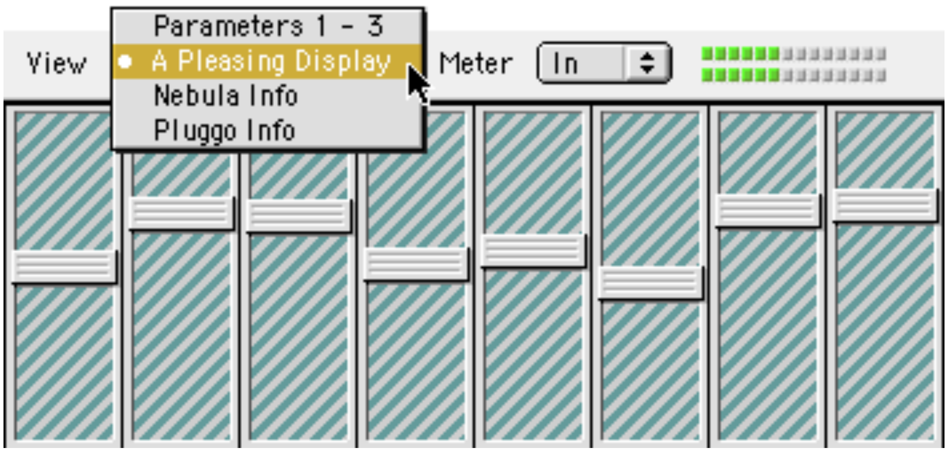
Nebula

category: multichannel

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

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.

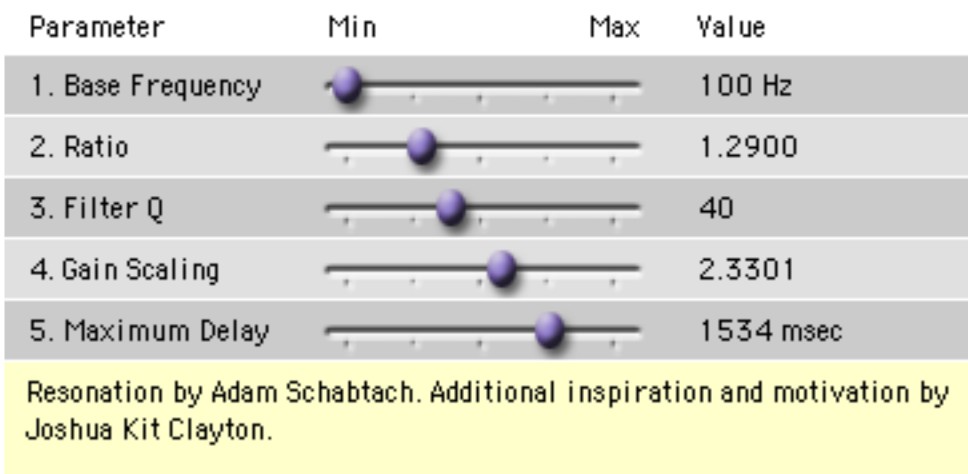
category: filter/delay

Resonation

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.



Visible Global Parameters

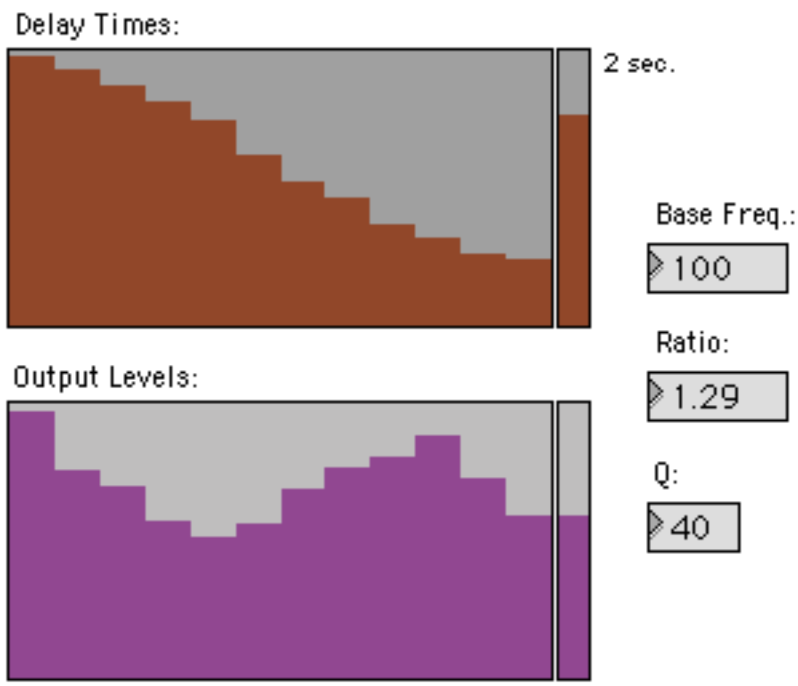
Name	*Min*	*Max*	*Units*	*Description*
Base Frequency	20	10000	Hz	Center frequency of the lowest-pitched filter.
Ratio	1	2		Ratio between successive filter frequencies. If this parameter is 1.0, all filters have the same frequency. If it is 2.0, the filters are tuned in octaves.
Filter Q	1	100		Sets 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.

Resonation

category: filter/delay

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

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.

Resonation

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

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 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.

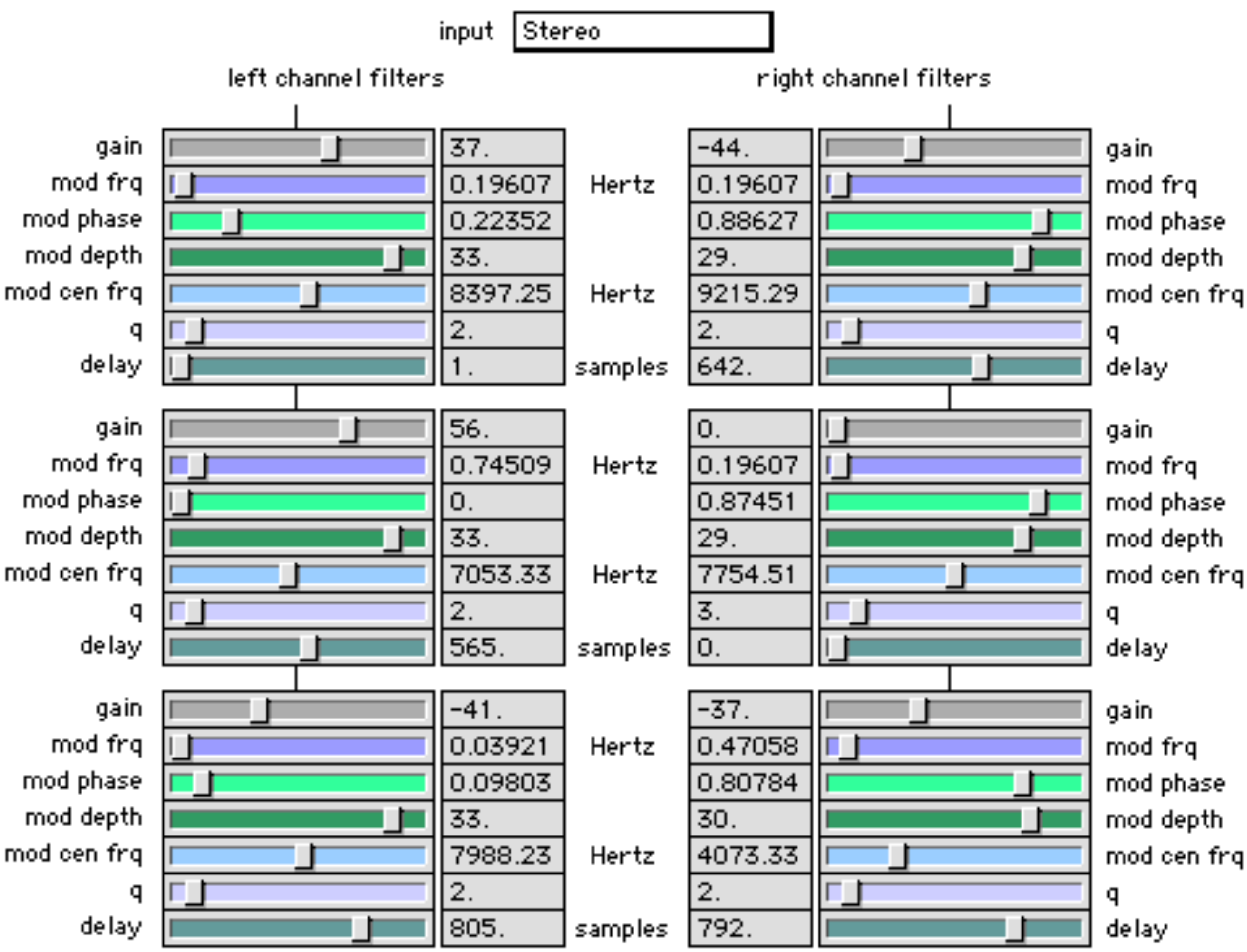
Resosweep

category: filter/delay

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

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.

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."

Spectral Filter

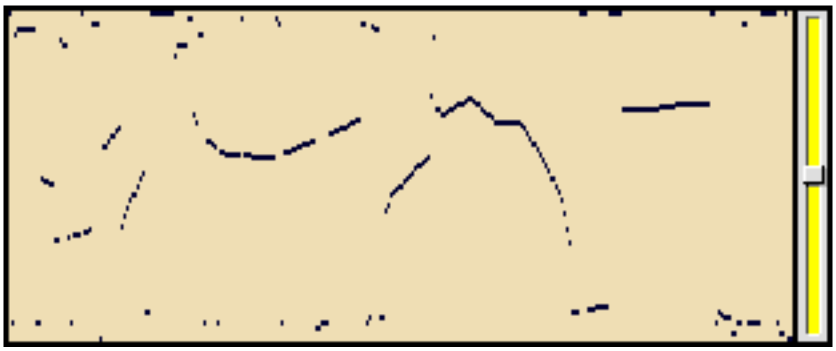
category: spectral

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

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