Integer

Digital-analog buffer device

AAX + **AU** + **VST** effect plugin for Mac/Windows/Linux

INTRODUCTION

Integer is a digital-analog buffer modeling effect. At its core is a delay engine based on a variable sample rate technology: just like analog "bucket bridge device" chips, this algorithm produces different delay time values by varying the speed of its internal sample clock. The plugin offers multiple buffer size and output interpolation options, as well as an ultra-wide clock range – from just hundreds of Hz up to tens of MHz, making it possible to drastically vary the audio resolution and character. A flexible sine generator is also included for modulating the delay time, with optional stereo expansion and gradual chaotic randomization.

Integer has a sophisticated bipolar feedback system with signal amplification, adjustable low-pass and high-pass filters, plus a brick-wall limiter that prevents overloads but lets the signal endlessly loop through the filters and the variable sample rate buffer. Thanks to its elastic, tape-like behavior, the pitch glide effects during delay time changes do not affect what's in the feedback loop. With all these features combined, **Integer** is capable of a huge variety of effects – ranging from delay, chorus, flanger to charismatic loops and drones, brick-wall filtering, sample rate manipulation, and anything in between.

SPECIFICATIONS

SOUND ENGINE

- Digital-analog delay with variable sample rate design, multiple buffer size options and extremely wide clock range
- Optional delay time tempo sync
- Variable input low-pass and high-pass filters
- Three output interpolation algorithms
- Bipolar feedback loop with possible endless amplification and brick-wall limiter
- Modulation generator with optional tempo sync, adjustable frequency and amplitude drift, and stereo phase offset
- One-pole lag filters on all continuous parameters for smooth, click-free adjustment
- Supports mono › mono, mono › stereo, and stereo › stereo channel configurations

GRAPHIC INTERFACE

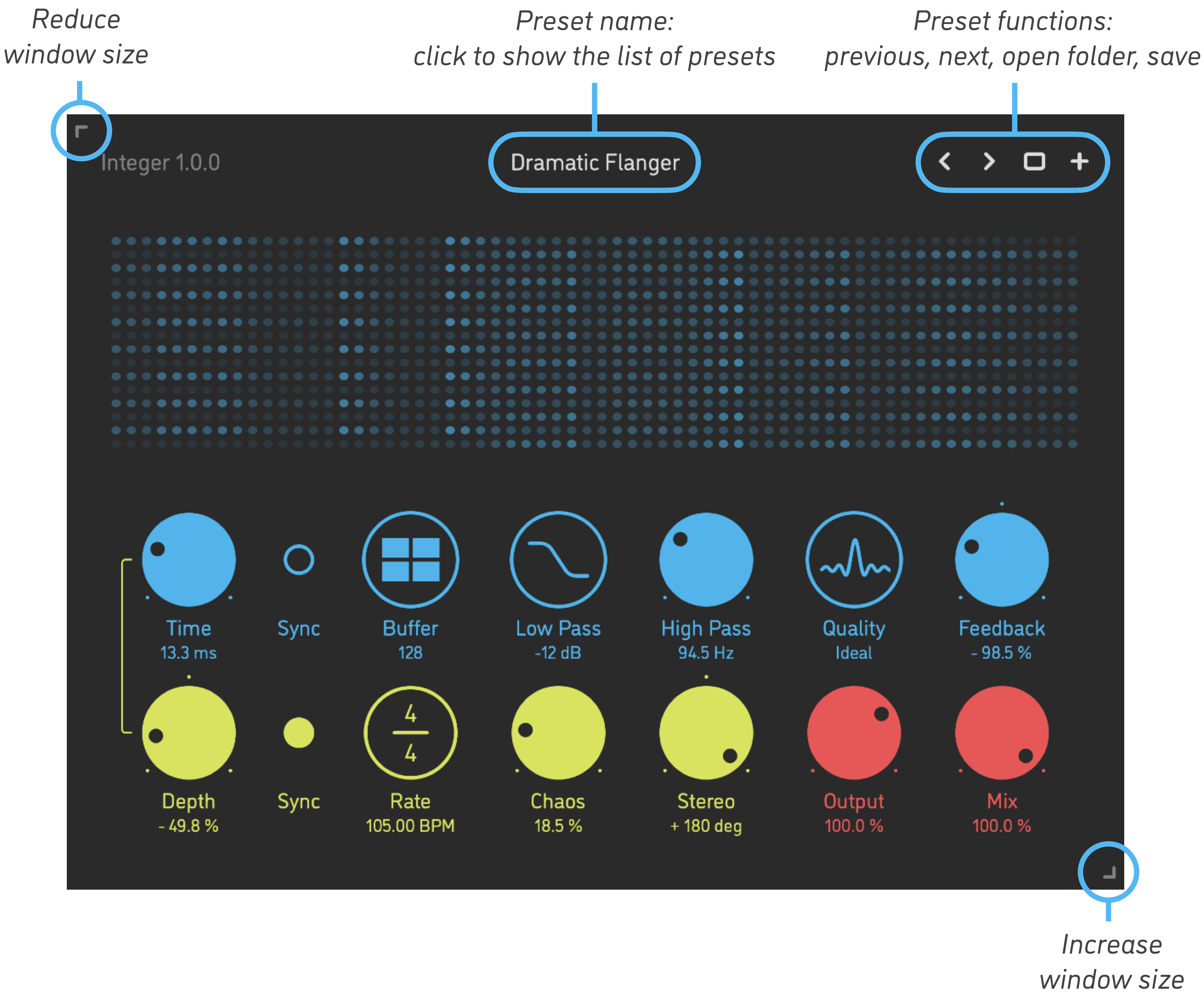
- Color-coded graphic elements
- Consistent name, mapping, value, and unit implemented for all parameters in both graphic user interface and host control/automation
- Built-in preset management functions
- Supports window size scaling up to 200%

SYSTEM REQUIREMENTS

- 64-bit Mac computer with Intel or Apple processor and Metal graphics support, **macOS 10.9** or later, plus a host application with support for **AAX**, **AU** or **VST3** plugins
- 64-bit PC computer with x86 processor, **Windows 8.1** or later, plus a host application with support for **AAX** or **VST3** plugins
- 64-bit computer with x86 processor, fairly recent **Linux** distribution, plus a host application with support for **VST3** plugins

INTERFACE

Integer features a fully vector-based interface, with color-coded elements for effective visual grouping. The plugin allows you to change its window size from 0.8x to 2x in 20% increments. The last size you set is stored in a preference file and is recalled the next time **Integer** is loaded.



- Hold *shift* and drag a knob to adjust the parameter with increased resolution.
- Use *option-click* (Mac) or *alt-click* (Windows, Linux), or *double-click* any knob to recall its default setting.
- To fully initialize all plugin's parameters, load the preset named *Default* from the *Factory* or the *User* bank.

PRESET FUNCTIONS

Integer features simple built-in functions for saving and loading presets, as well as for quickly switching between presets within the same bank. All these functions are accessed via the top toolbar.

Preset Name — Click the preset name at the top to show the list of presets in the current bank. Use *command-click* (Mac) or *control-click* (Windows, Linux) to reveal the actual preset file in the system file browser.

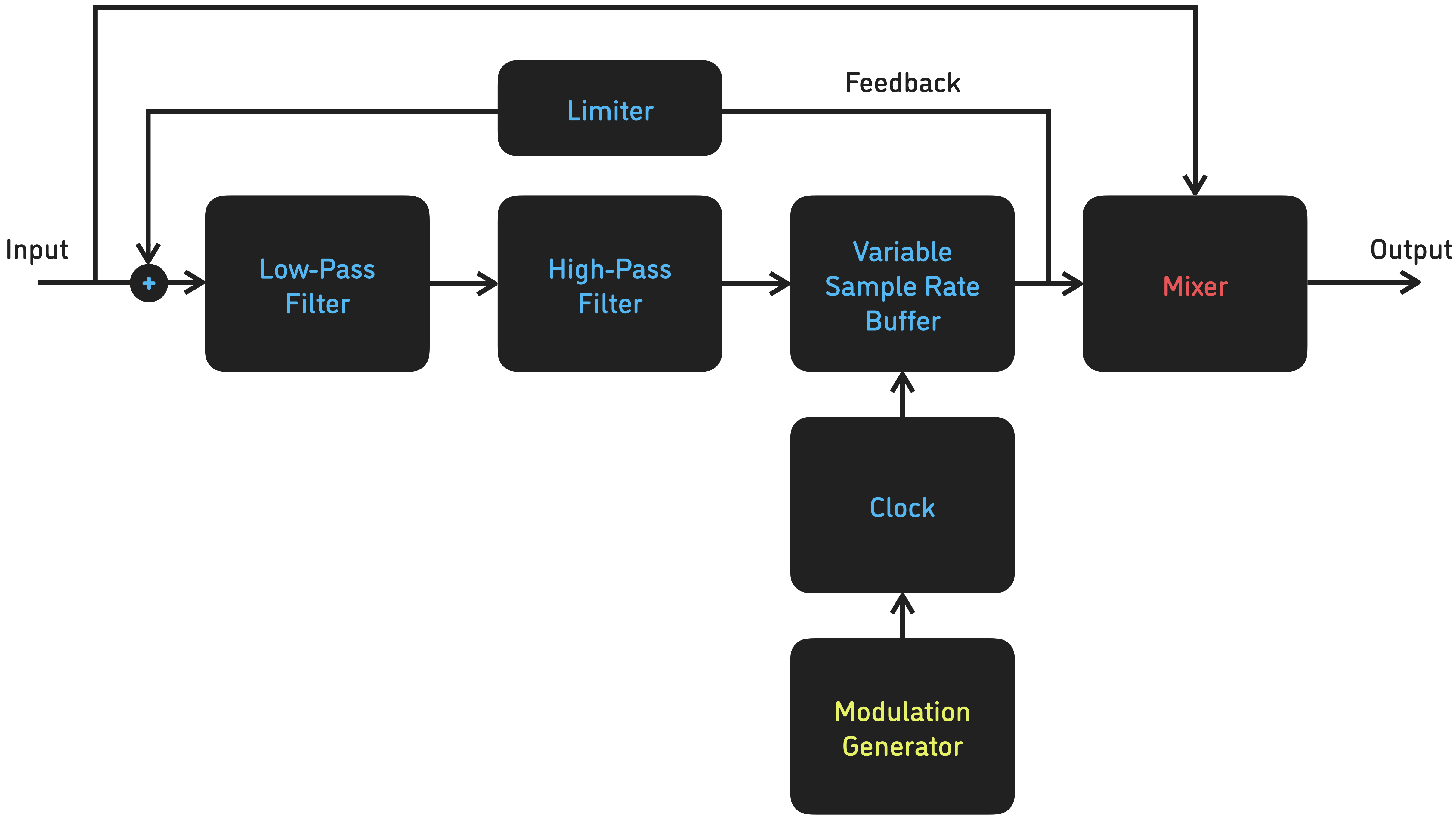
Switch to the previous preset in the current bank. The current bank is automatically set to wherever the last preset was loaded from.

Switch to the next preset in the current bank.

Show open file dialog with the list of preset banks. By default, the plugin includes two banks: *Factory* and *User*. However, you can freely create additional banks – simply by creating new subfolders.

Save current preset. Please note: due to the limitations of the typeface, you can only use the Latin alphabet when naming your presets

DSP DIAGRAM


Feedback
Limiter
Input
+
Low-Pass Filter
High-Pass Filter
Variable Sample Rate Buffer
Mixer
Output
Clock
Modulation Generator

TECHNICAL BASICS

A classic digital delay operates via a so-called circular buffer: a chunk of memory is used to store the input signal, and to change the delay time, the read head is placed closer to or further away from the write head. Such an algorithm would use less memory for shorter delay times, and more memory for longer delay times.

Integer operates differently from such digital delays, and it's more similar to analog BBD (bucket brigade device) delays: utilizing a relatively small buffer, the position of the read head in relation to the write head is fixed, and to change the delay time the clock that controls the read-write process simply runs at a variable rate. This can also be called a "variable sample rate" delay – to achieve different time ranges it literally shrinks or stretches its buffer, and regardless of the delay time and the clock rate, the exact same amount of memory is used. Thus, the audio resolution changes with delay time, and the smaller the buffer, the more severely the sound quality will degrade with longer delay time values. At the most extreme settings – smallest buffer size with longest delay time – the effect will reproduce only the lowest of the low frequencies.

The quality and overall character of the effect also changes with two other important settings in **Integer**: input low-pass filter slope and output interpolation type. The low-pass filter has its cutoff frequency locked to half of the buffer clock rate, to fight the aliasing of the higher frequencies; reducing or increasing the filter slope will, respectively, increase or reduce the aliasing as the signal is being written into the buffer. Additionally, three settings are offered for interpolation during reading from the buffer, with a dramatically varying degree of restoration precision.

It is important to consider that the buffer clock rate depends on three variables: buffer size, delay time, and the plugin host's sample rate. Due to an extremely wide range of buffer size and delay time parameters offered in **Integer**, the clock rate can reach both extremely high and extremely low values – and at the high end, not only it can exceed 40 MHz (0.1 ms delay time with 4096 sample buffer at 48000 KHz host sample rate), but it will also drastically increase the amount of computing power required to operate at such high values. Likewise, the most high-quality output interpolation option available in **Integer** (33-point sinc) is going to further increase the computational load as well.

PARAMETERS 1/2

Time	0.1 .. 1000 ms 1/64 .. 8/8	Delay time in milliseconds (when the Sync switch is off) Delay time as a host tempo fraction (when the Sync switch is on)
Sync	On / Off	Defines whether the delay time is set as a continuous value in milliseconds, or as a fraction of the current host tempo
Buffer	128 .. 4096	Size of the delay buffer: this setting, together with the Time parameter, defines the buffer clock rate and the spectrum resolution across the entire Time range
Low Pass	-6 .. -48 dB	Slope of the input low-pass filter (the cutoff frequency is set to half of the clock rate); this parameter affects the amount of aliasing produced when writing the incoming signal into the buffer
High Pass	20 .. 800 Hz	Cutoff frequency of the -6 dB/octave input high-pass filter
Quality	...	Buffer readout interpolation type: Vintage (2-point linear), Modern (4-point spline) or Ideal (33-point sinc); this parameter affects the resolution of the reconstruction process when reading from the buffer
Feedback	-200 .. +200%	Amount of the delay buffer’s output signal being fed back into the input

PARAMETERS 2/2

Frequency	0.01 .. 10.0 Hz	Modulation generator speed in Hertz (when the Sync switch is off)
Rate	1/32 .. 32/1	Modulation generator speed as a host tempo fraction (when the Sync switch is on)
Depth	-100 .. +100%	Amount of modulation applied onto the delay Time parameter (modulator signal is a bipolar sine wave, with the maximum modulation range of +/- 25 milliseconds)
Sync	On / Off	Defines whether the modulation generator speed is set in Hertz or as a fraction of the current host tempo; when this switch is on, the modulator phase is also continuously synchronized to the host transport location
Chaos	0 .. 100%	Amount of smooth randomization applied onto the modulator frequency and amplitude via a separate random triangle generator (when the Sync switch is on, only the modulator's amplitude gets randomized)
Stereo	-180 .. +180 deg	Modulation phase offset between the left and the right channels
Output	0 .. 200%	Wet processed signal level (with up to 2x amplification)
Mix	0 .. 100%	Crossfade-style balance between the dry input signal and the wet processed signal: – 0 .. 50%: dry signal level is at max, while the wet signal is gradually faded in – 50%: the dry and the wet signal levels are equal – 50 .. 100%: wet signal level is at max, while the dry signal is gradually faded out

SINEVIBES

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