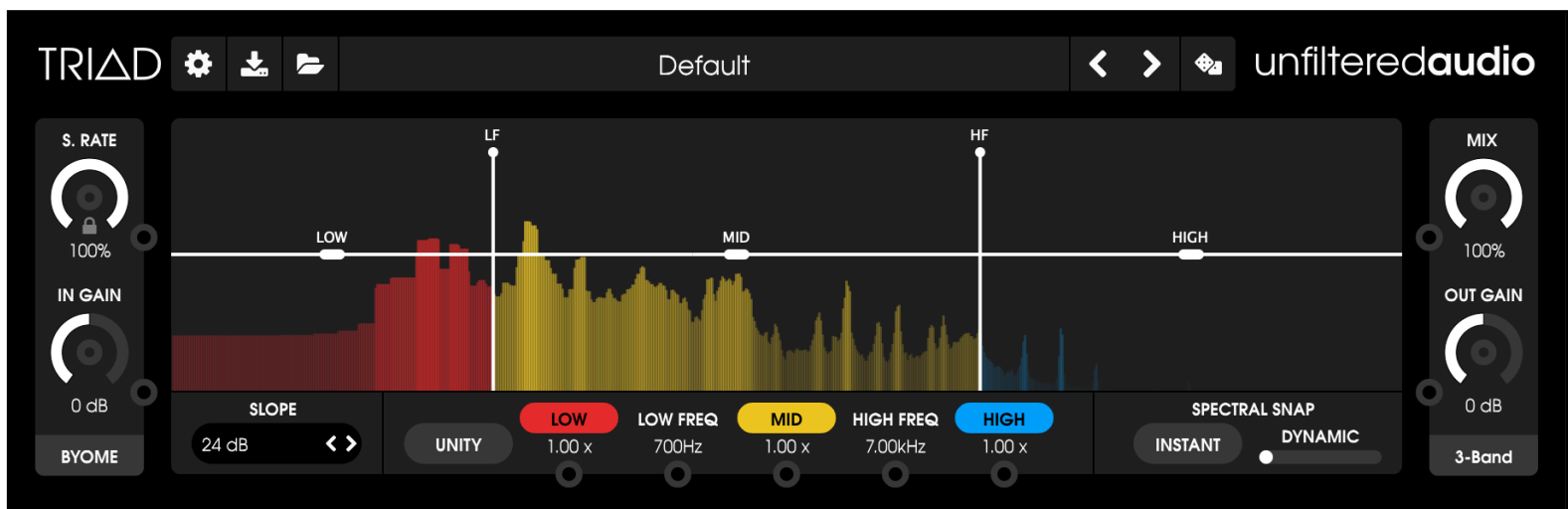# TRIAD

Manual by unfilteredaudio

Introduction

Triad is the next step in the BYOME ecosystem. It takes our celebrated BYOME technology and triples it in exciting ways.

Features

- Three simultaneous instances of BYOME with many modes to choose from.
- Auxiliary outputs, one for each band!
- Over 40 Effect Cells to choose from!
- Over 400 presets, including over 200 artist presets from Glitchmachines, Richard Devine, and ignatius.
- Many new Modulators to choose from, including Spectral Follower and Probability Gate.
- Supports external sidechain signals for Dynamics Effects and Follower Modulators.
- Presets for individual Effects and Modulators.
- Per-section, per-Effect, and per-Modulator randomization.

Getting Started

Installing Triad

Triad can be installed using the Plugin Alliance Installation Manager. If you are not familiar with how to use it, please consult the Installation Manager's manual.

Browsing Presets

Triad comes with a wide array of presets to get you started. The preset manager at the top of Triad's interface is used to load, save, or browse presets. If you want to get a good feel for the power of Triad, spend some time with these presets before browsing the manual.

Standard Unfiltered Audio Features

CONTROLS

-All knobs and vertical sliders are controlled in a smooth up-and-down motion.
-Use the Command key (on OS X) or Control key (on Windows or OS X) to fine-tune controls.
-Double-click or Alt-click on a control to return it to a default setting.
-Double-click a value label to edit it and quickly jump to an exact value.

PRESETS

-All presets are saved with a .uap file extension. These presets are compatible across all platforms and plug-in formats. They use a simple, non-encrypted XML format for easy editing.
-Right-click on any control to Lock it. A Locked control will not change when switching or randomizing presets.

INTERFACE

-Click the Gear icon on the preset manager to bring up interface options.
-Dark and Light skins provide a comfortable view for all studio situations.
-Resizable interface with vector rendering for easy viewing on all monitors.

RANDOMIZATION

-Click the Dice icon to randomize the current preset.
-Right-click the Dice icon to bring up a more detailed randomization options menu.
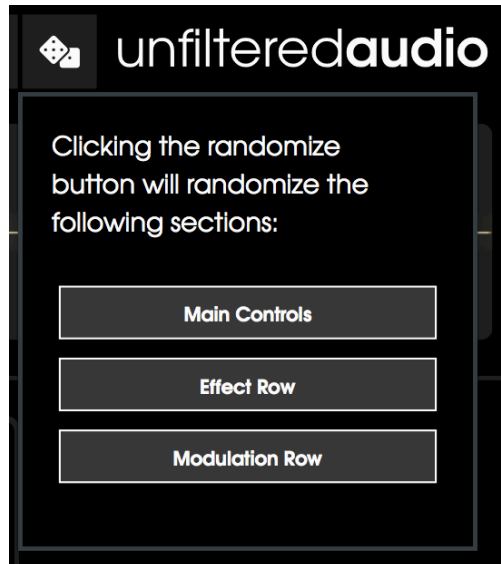
Main Controls

INPUT

IN GAIN: Controls the amount of gain applied to the incoming signal before it is sent to the Triad effect processor.

S. RATE: Controls the sampling rate of the entire plugin. Instead of simply adding a lo-fi aliasing effect to the input, this will have a massive influence over every effect. For instance, it will expand all delay buffers, change the maximum cutoff of the filters, affect internal effect modulations, and more. Note that this does not have an effect on our modulation system!

OUTPUT

MIX: Sets the balance between the dry, incoming signal and the wet (processed) output signal. At 0%, you will only hear the input signal. At 100%, you will only hear the signal that has been processed by Triad.

OUT GAIN: Controls the amount of gain applied to the signal after being processed by Triad.
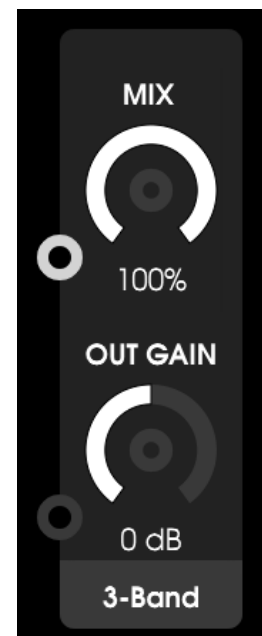
The BYOME Splitter

The heart of Triad is its BYOME splitter. This section takes your input and distributes it across Triad's BYOME rows in various ways.

Frequency Splitter

In the two frequency splitting modes ("3-Band" and "2-Band"), the incoming audio is separated into multiple frequency bands through the use of analog-style crossover filters. The individual bands are then processed on separate BYOME rows. This enables you to create extremely deep and detailed processing chains. For instance, if you want to add saturation to a signal's bass frequencies while leaving the upper registers untouched, you could add the Saturation effect to only the Low Band. On top of that, you could make the higher frequencies shine more by adding Contrast or Stereo Image.

In the 2-Band mode, one of the BYOME rows is used as a Serial processor. This Serial band can occur before the frequency split (PRE) or after it (POST), allowing for more unified processing.

3-BAND
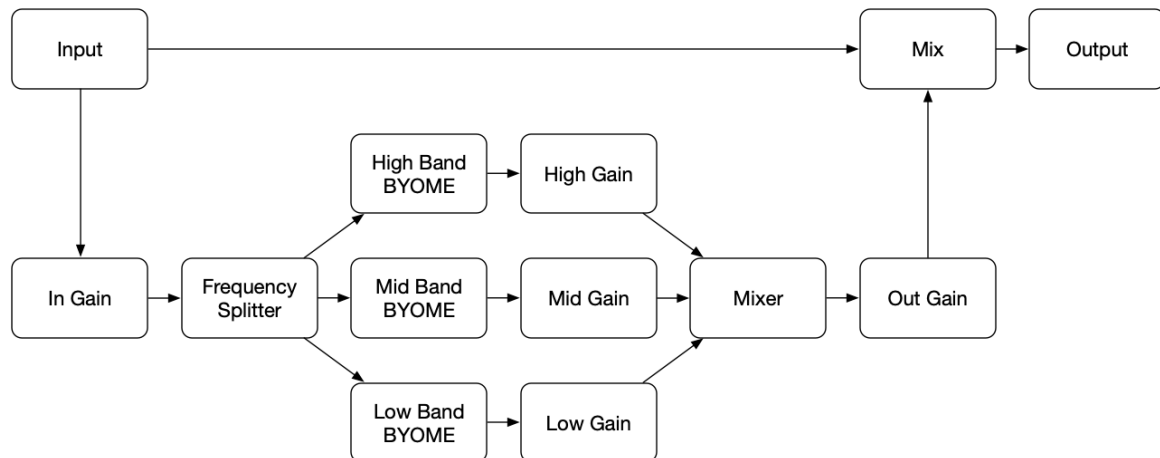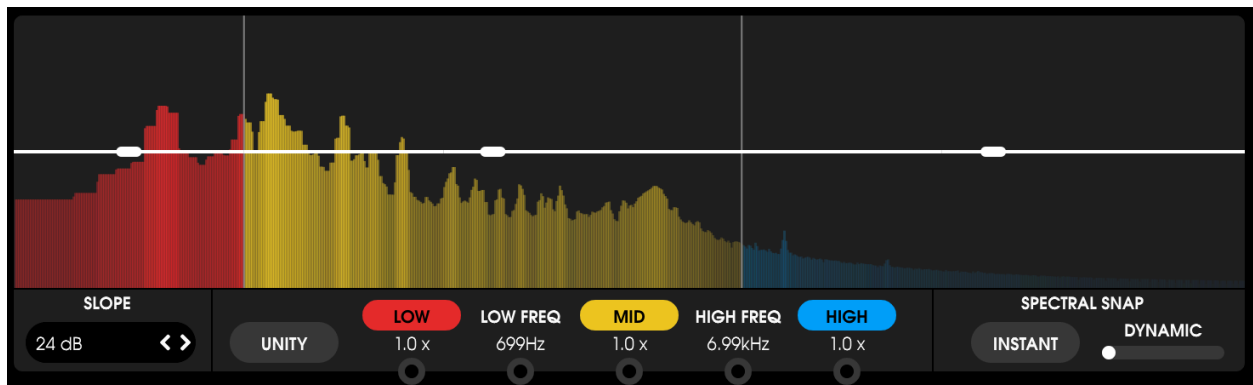
Slope: This selects the slope of the crossover filters, or how rapid the transitions are between frequency bands. Lower values will allow for more overlap between the bands, leading to less dramatic transitions between the rows.

Unity: When enabled, the individual bands' gains will always be compensated to maintain an even value when the bands are summed. Changing the individual band gain will raise/lower the others to compensate.

Low/Mid/High: These color-coded buttons (matching their frequency ranges in the visualizer) can also be clicked to mute the band. When a band is muted, it will turn grey in the visualizer.

Low/Mid/High Gain: The horizontal sliders on the visualizer can be used to change the gain of each frequency band. Even without using the BYOME rows, the 3-band frequency splitter can be used as a very powerful 3-band EQ.

Low/High Freq: The vertical bars between the frequency bands can be dragged to change the cutoff frequencies between the bands.

Instant Spectral Snap: When this button is clicked, Triad will analyze the current spectral frame of audio and automatically change the cutoffs between the frequency bands. It will attempt to set the cutoffs to values that create frequency regions that are roughly equal in "power".

Dynamic Spectral Snap: This is similar to Instant Spectral Snap, but it will adjust the frequency bands in real-time. The depth of this parameter will determine the range of deviation from the user-selected cutoff points.
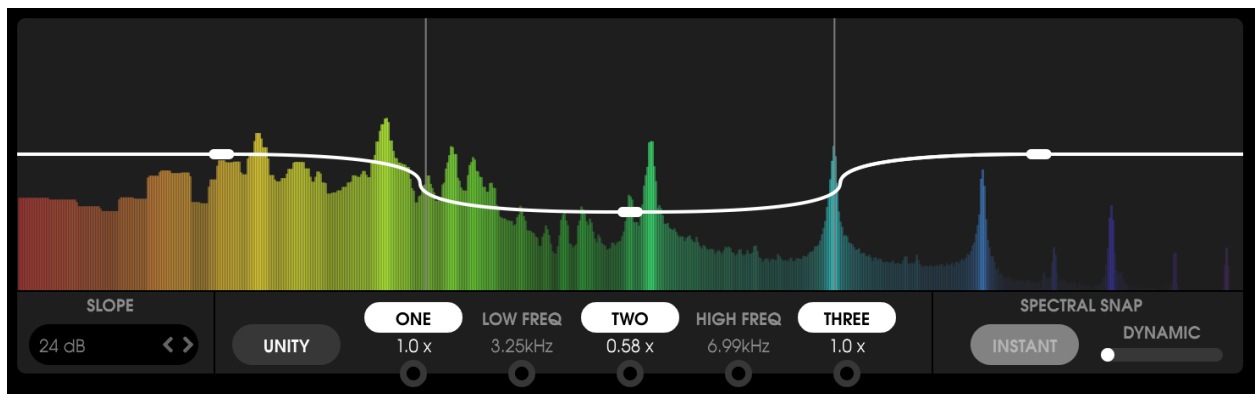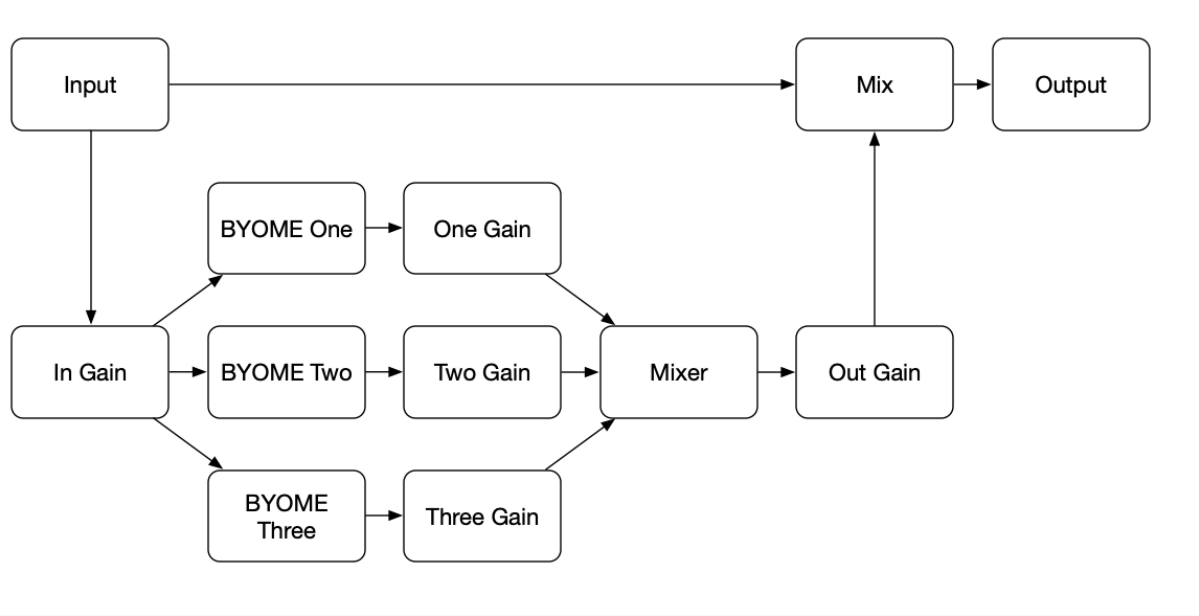
2-BAND
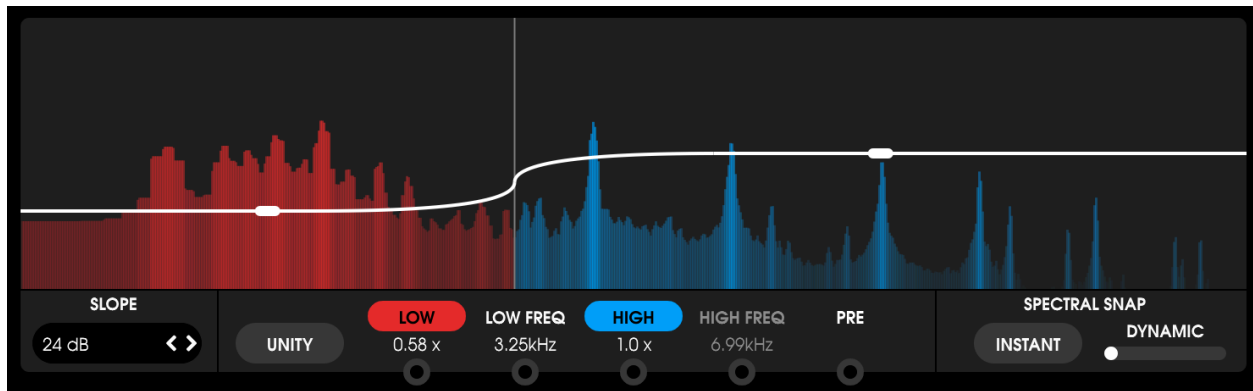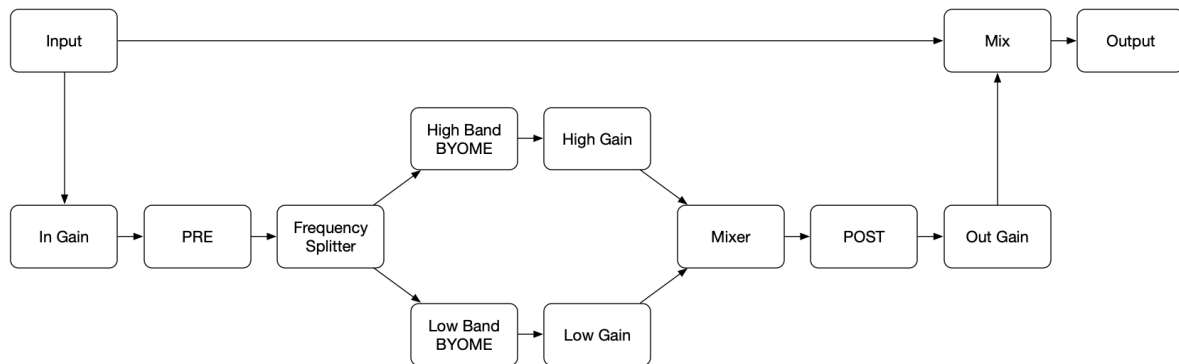
The control set for the 2-Band mode is very similar to the 3-Band mode. Instead of Low/Mid/High, there are only Low and High bands. The biggest change is the PRE button. Clicking this buttons will determine whether the Serial row appears before or after the frequency split.

Parallel Processor

In the next two modes ("3x Parallel and 2x Parallel"), Triad's BYOME rows are reconfigured for parallel processing. This means that the signal is duplicated and the identical copies are processed separately before being summed together. Like the 2-Band frequency splitting mode, the 2x Parallel mode has a Serial band that can be used before or after the parallel processing.

3X PARALLEL

In this mode, the Slope, Spectral Snap, and Frequency Cutoff controls are disabled. The only active controls are the gain controls for each parallel row.

2X PARALLEL
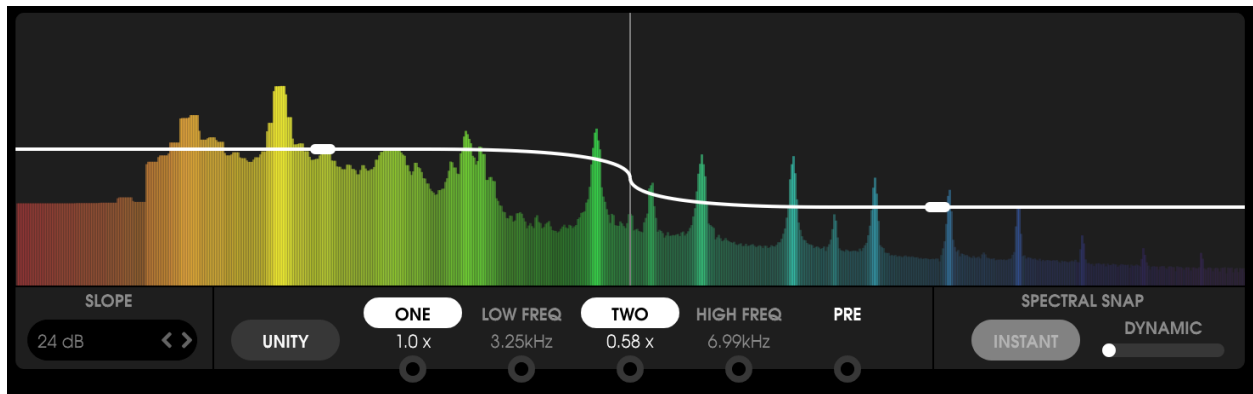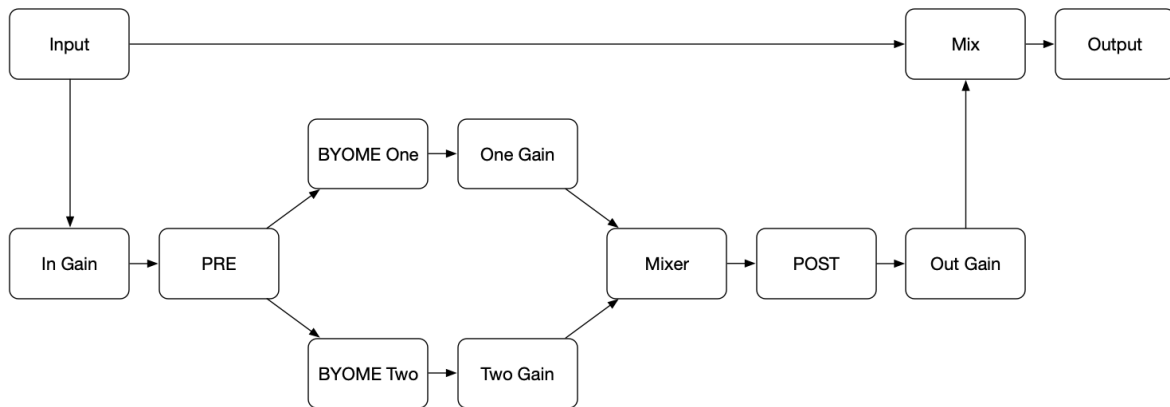
Like the 2-Band mode, the third band is replaced with a Serial band with a PRE/POST button to select where the band appears in the processing chain.

Channel Processor

In the final two modes ("Mid-Side" and "Left-Right"), Triad's BYOME rows act on the input's individual channels. These two modes are only available when Triad's outputs are stereo.

MID-SIDE
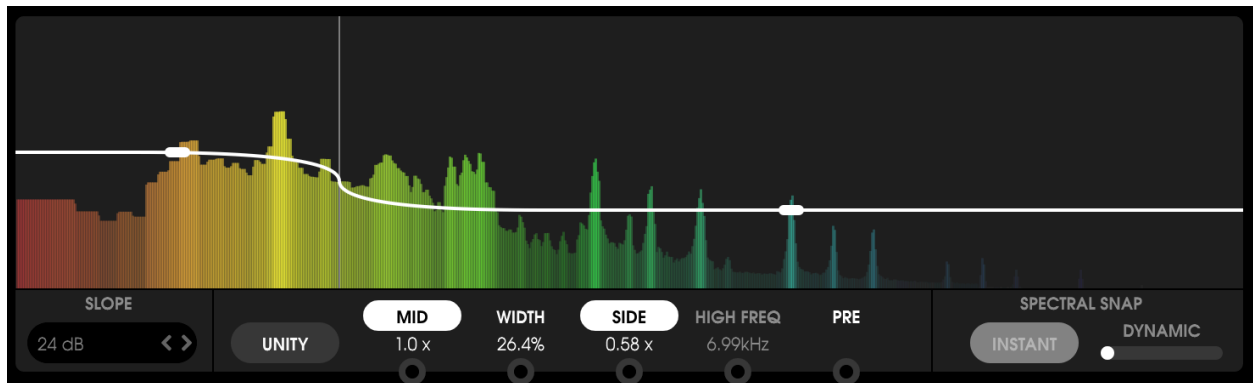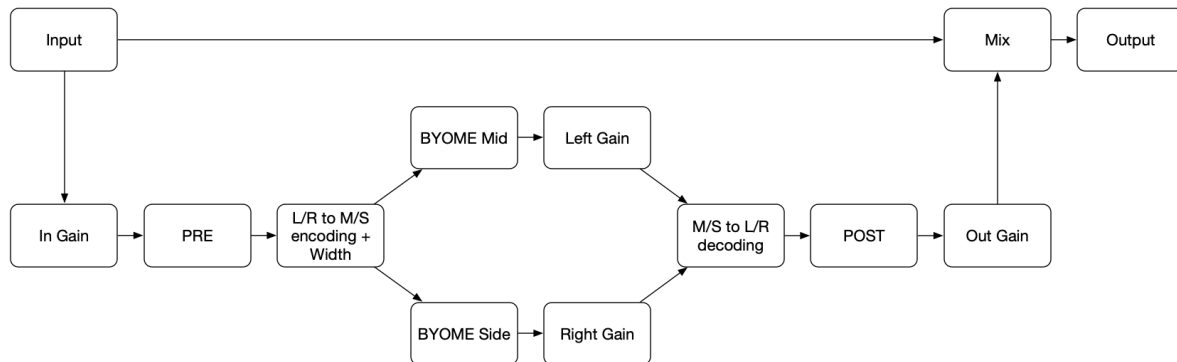
In this mode, the vertical visualizer bar becomes a WIDTH control. The WIDTH control determines how much spatialization is applied to the input at the point of Mid-Side encoding. At 0%, the input is made mono, leading to everything being routed to the Mid channel. At 200%, the audio is artificially widened, creating a "super-stereo" effect.

LEFT-RIGHT
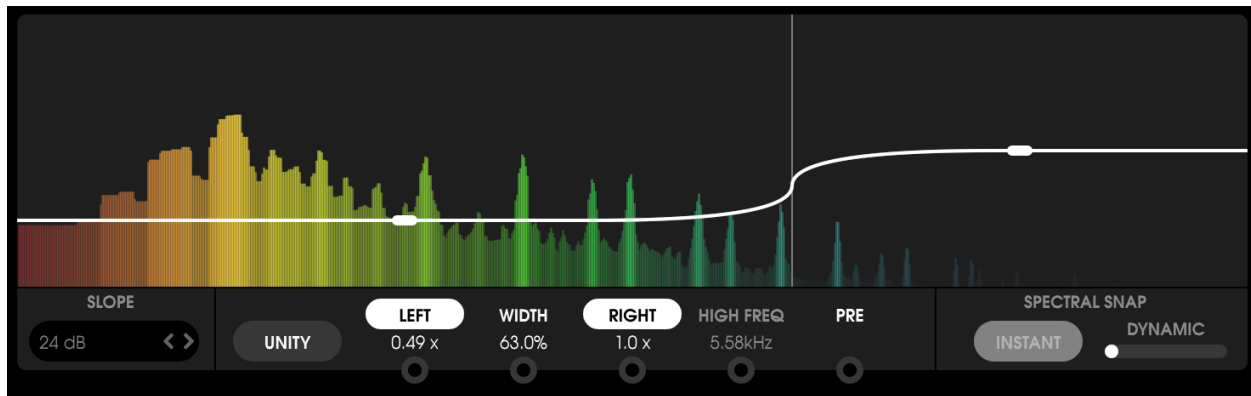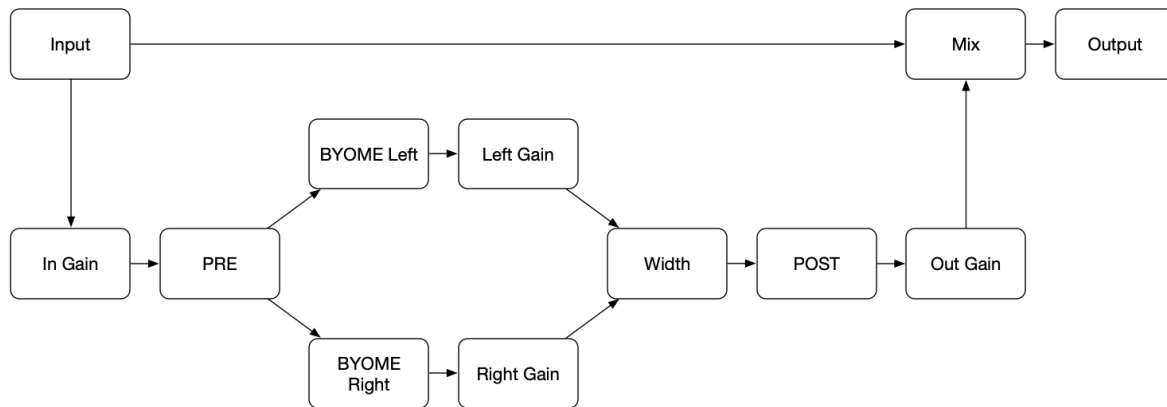
Like the Mid-Side mode, the vertical bar becomes a WIDTH control. However, in this mode, the Width algorithm is applied after the Left and Right channels are processed by their BYOME rows. If you have dramatically different effects on the two channels, decreasing the width can create a more coherent image. If you want to change the width before the Left/Right processing, you could set the Serial band to PRE and add a Stereo Image effect to that row.

BYOME - Row Effects and Modulation

Clicking the BYOME button will expand Triad's interface to show the three BYOME rows along with the Modulation row.

One row will be expanded at a time to show more detailed controls. To expand another row, simply click on that row's title (i.e. "MID") or any of the effect headers on that row.

An innovative Triad feature is ***Row Permutation***. This allows you to quickly swap the functionality of each row. For instance, in 3-band splitter mode, you can swap which effect chains are used to process the High, Mid, and Low bands. If you have an interesting delay effect on the High band but would rather hear what it sounds like on the Mid band, the Row Permutation feature is perfect for that. To use the Row Permutation feature, simply click the arrow buttons between the row titles and effects (the black button with the "1" label in the image above).

Row Functions

Right-clicking a row's title will bring up a menu of row-processing functions.

Randomize Values - Randomize all unlocked controls on every effect on this row.

Lock/Unlock All - Lock or unlock every control on every effect on this row (Locks prevent changes to a knob during randomization).
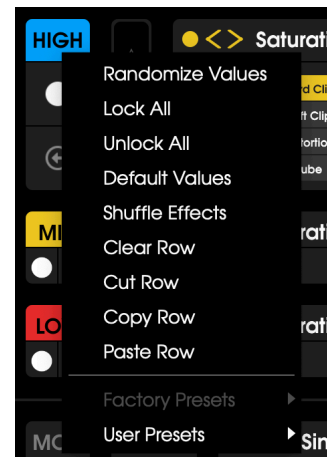
Default Values - Reset every effect on this row to its default state.

Shuffle Effects - Randomize the order of the effects on this row.

Clear Row - Delete every effect on this row.

Cut/Copy/Paste Row - Quick methods for moving the contents of one row to a different row (alternatively, try the Row Permutation feature mentioned above).

Factory/User Presets - This will import a BYOME's preset's effect list to this row. It will only import the effect list and not the modulation. To import a complete BYOME preset, use the main Triad Preset Bar (at the top of the plugin's interface).

BYOME Effect Cells

BYOME ("Build Your Own Multi Effect") is not just a plugin… it's our newest Unfiltered Audio technology! The BYOME row consists of 40 Effect Cells organized into 8 categories.

Each Effect Cell shares the following interface features:

Power Button: Bypasses or enables the effect.

Arrows: Quickly navigate between effects.

Title Tooltip: If Tooltips are enabled, hovering over the Cell's title will bring up a Tooltip with a description of the effect.

Preset Menu: With BYOME, not only do you have the ability to save full plugin presets, but you can also save presets for individual Cells. Here, you can save/load per-Cell presets, or you can randomize individual Cells.

Right-Click Menu: Right-clicking on the Cell's header will bring up a small menu of frequently used tasks for manipulating the Cell.

Minimize: Double-click on a Cell's header to change the Cell to a compact view. Double-click again to re-expand it.

MIX: Controls the balance between the wet output of this Cell and the output of the previous Cell.

VU Meter: Shows the amplitude of each channel output from the Cell. Additionally, a third meter showing Gain Reduction will appear on the Compressor Cells.

Gain: The red line on the VU Meter is an additional output gain control. You can double-click on the VU Meter to return the output gain to +0.0 dB.

DELAYS

This category consists primarily of effects that add distinct echoes to a signal.

All Delay effects share these common controls:
Delay: Sets the length of the delay buffer for either the left channel or both channels (this functionality is dependent on the Delay Link Mode).

Spread: Sets the length of the right channel's delay buffer (this functionality is dependent on the Delay Link Mode)

Delay Link Mode
Linked: Left and Right delay times are identical.
Unlinked: Left and Right delay times are set independently.
Offset: The Right delay time is set as a relative offset to the left delay time.

Feedback: Sets the amount of feedback sent to the delay effect. This determines how many echoes will appear. This control is not present on the Stutter effect.

XFeedback: Sets the amount of feedback sent between delay buffers on opposite channels. This will create a wide, ping-pong style delay effect. This control is not present on the Stutter effect.

Filter: Applies low-pass or high-pass filtering to the feedback loop. This control is not present on the Stutter effect.

Tempo Sync: When enabled, the delay times will snap to divisions of the plugin host's BPM. When disabled, delay times are set in milliseconds.

GLITCH SHIFTER

This effect was created by breaking our Pitch Delay algorithm in an entertaining fashion. It uses a granular window to tear through a buffer in a rather unpredictable way, but we've put a lot of effort into eliminating unwanted clicks or pops. If you attempt to use this as a traditional delay, note that the echo time will always be around 100 ms. Changing the Delay Time instead changes how much the granular window traverses each cycle.
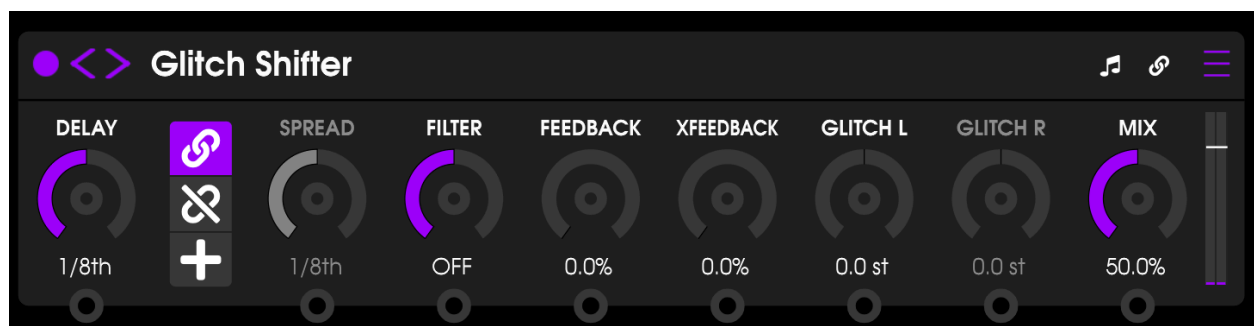
Glitch L/R: Sets the glitch parameter for the channel. It is an unpredictable parameter, but the best way to explain it is as a broken pitch shifter that wildly runs its way through an input buffer.

Link: When active, the glitch parameter will be identical for both channels. When inactive, they can be set independently.

INSTANT DELAY

This is a delay where you can change the Delay Time parameter without introducing audible effects (i.e. clicks or pitch changes). This works really well in tandem with our modulation system.
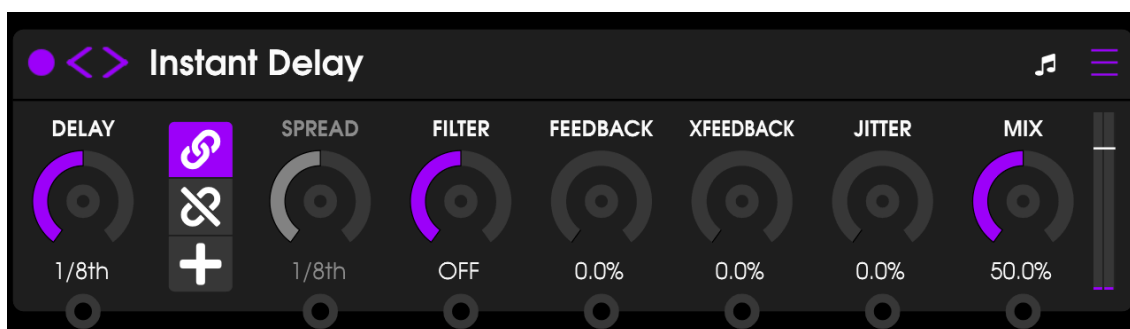
Jitter: When above 0%, the delay time will be periodically randomized. The period of randomization is equal to the previously randomized delay time. This will lead to a different feeling than simply attaching a random modulation signal to the Delay Time input.

MULTITAP

This is a delay with up to 16 separate taps. The TAPS and SPACE parameters modify these taps in musically useful ways without the need to dive in and edit each tap separately. This effect works better with longer delay times. With shorter delay times and a neutral SPACE setting, you can treat this like an unusual Comb Filter algorithm.
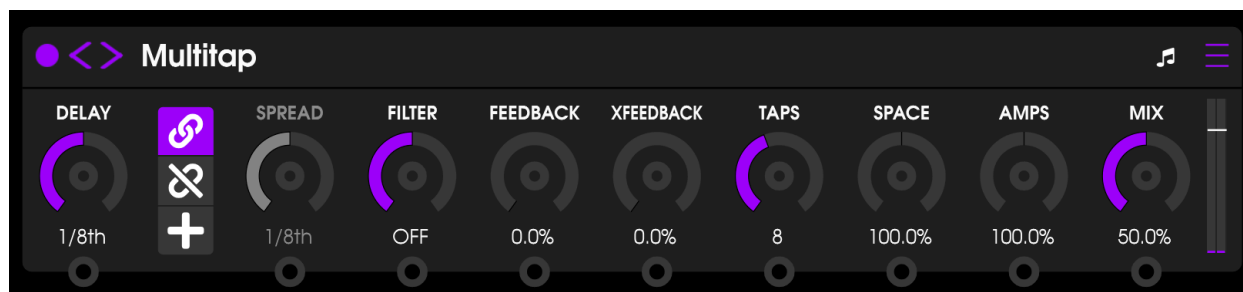
Taps: Sets the number of active delay taps.

Space: Sets the delay time spacing of the taps. At 12 o'clock, they will be evenly spaced from 0 ms to Delay Time. At lower values, the taps will cluster towards earlier times. At higher values, the taps will cluster towards the value set by Delay Time.

Amps: Sets the amplitude curve applied to the delay taps. At 12 o'clock, all taps will have an equal amplitude. At lower values, the taps will decay in amplitude. At higher values, the taps will swell in amplitude.

PITCH DELAY

This effect combines a traditional delay with a granular pitch shifter. The pitch shifting occurs before the delay buffer, so all feedback is continuously shifter. This causes echoes to either climb or descend in pitch. For fun, try setting each channel to a different pitch and turn up XFeedback. This is even more unusual with uneven delay times.

Pitch L/R: Sets the pitch shift amount for the delay line. This pitch shift occurs before feedback, so the pitch shift will be repeatedly applied to all echoes.

Link: When active, the pitch shift parameter will be identical for both channels. When inactive, they can be set independently.
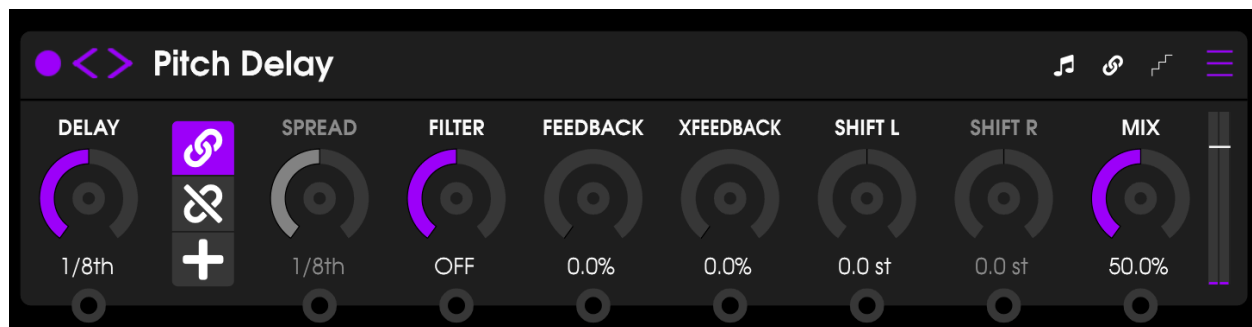
Quantize: When active, the pitch shifter parameter will be locked to integer semitone values.

REVERSE DELAY

This algorithm uses a granular engine to reverse chunks of audio. Feedback has an unusual property in this algorithm: because the granular reversal occurs before the delay buffer, the echoes will alternate between playing forwards and backwards.
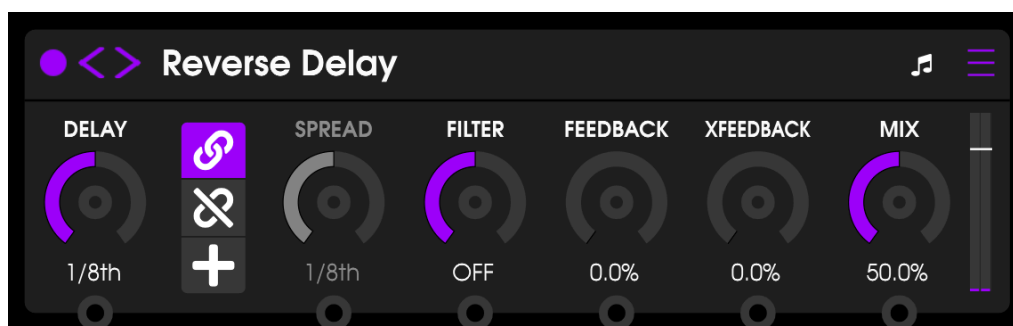
STUTTER

This algorithm uses a granular engine to reverse chunks of audio. Feedback has an unusual property in this algorithm: because the granular reversal occurs before the delay buffer, the echoes will alternate between playing forwards and backwards.

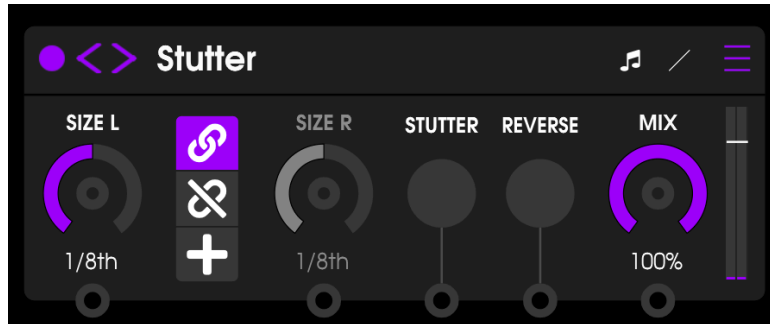**Smooth**: When active, the stutter buffer will be crossfaded to remove clicks.

Stutter: Activates the stutter effect, endlessly repeating the last chunk of audio. The size of the chunk is determined by the SIZE controls.

Reverse: When active, the frozen stutter buffer will be played backwards.

TAPE DELAY

This is a classic delay algorithm that mimics the behavior of old tape delays. Changing the Delay Time parameter will create a pitch slide effect as the delay's read head repositions itself. An additional WOW parameter mimics the sound of an unstable tape as it warbles in pitch.
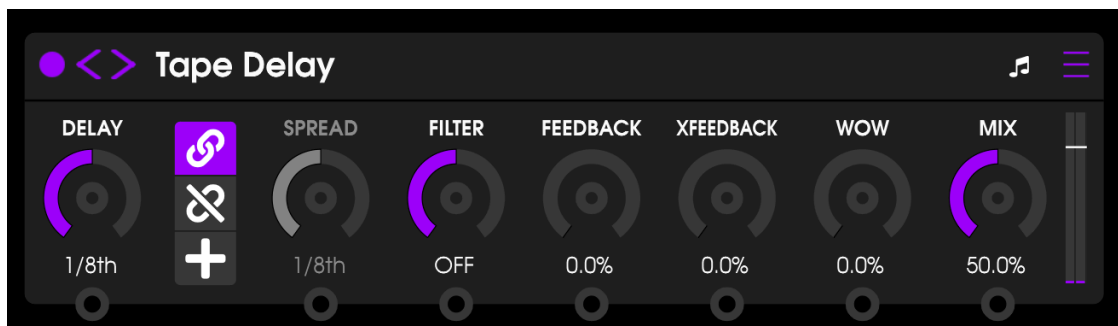
Wow: Sets the amount of pitch wobble on the tape. This effect is much more noticeable with large FEEDBACK amounts.

DISTORTIONS

This effect category consists of mostly non-linear effects for adding harmonic interest to signals.

BITCRUSH

This is a lo-fi effect that mimics the behavior of old digital equipment by lowering the resolution of the input signal. Six different bitcrushing algorithms are included for a variety of timbres.

Bits: Sets the number of bits used to represent the signal.
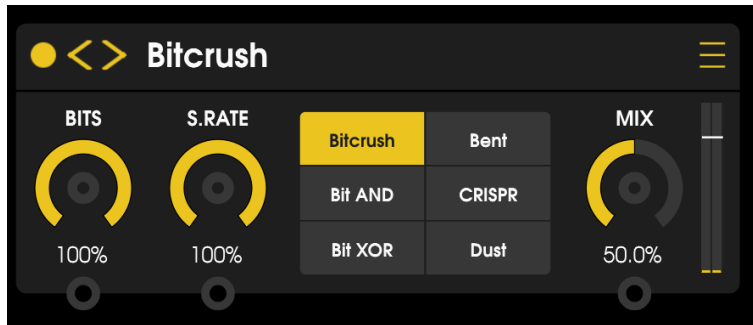
S. Rate: Sets the sampling rate.

Bit Modes
Bitcrush: Traditional bitcrushing.
Bit AND: The input is converted to a 32-bit integer representation. The BITS param generates a 32-bit number. The output is the logical AND of these two numbers.
Bit XOR: The input is converted to a 32-bit integer representation. The BITS param generates a 32-bit number. The output is the logical XOR of these two numbers.
Bent: This algorithm simulates various failures and interactions on a circuit bent instrument.
CRISPR: This is a light algorithm that focuses on crushing mostly inaudible bits. In general, this can add a bright crispness to a sound.
Dust: This is a different type of AND algorithm. It typically produces less aggressive results than the main Bit AND mode.

CHEBYSHEV

This is a multimode distortion unit based on Chebyshev polynomials. A Chebyshev polynomial is an algorithm that takes in a unit sine wave and outputs a harmonic of that sine wave. However, on any other input, this is an unpredictable distortion. The timbre of each harmonic varies wildly, so it is fun to experiment. Please note that some combinations will cause an apparent gain dip due to phase cancellations. As an example, turning up harmonics 1 and 3 simultaneously will lead to a dip in amplitude.

Harmonic 1-8: Sets the amplitude of harmonic 1-8.
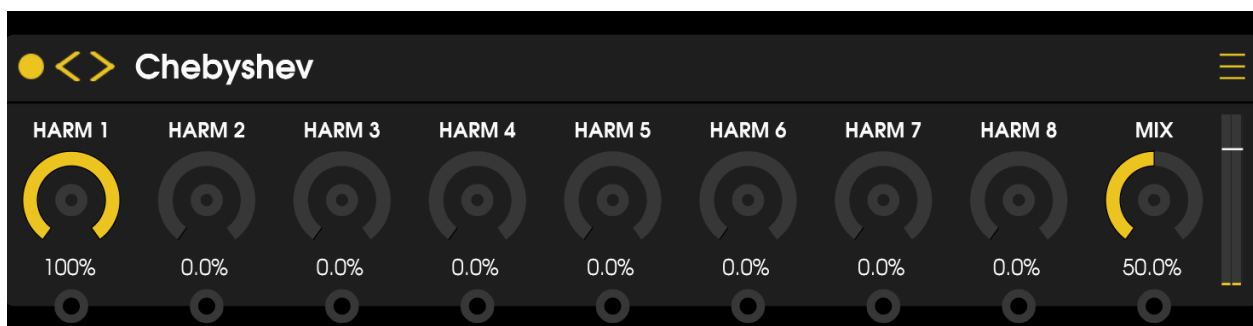
CONTRAST
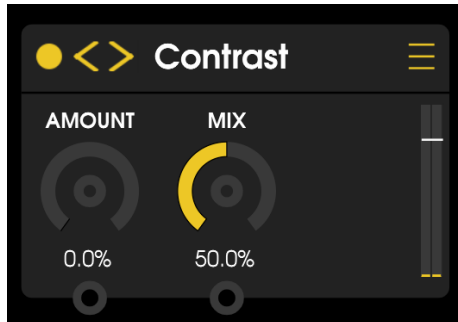
This is a mixing effect that is sometimes referred to as 'Audio MSG'. It is a light type of phase distortion that increases the brightness and presence of a signal. It is recommended to use this sparingly on signals that need it, but not the full mix.

Amount: Sets intensity of Contrast effect.

DENT RECTIFY
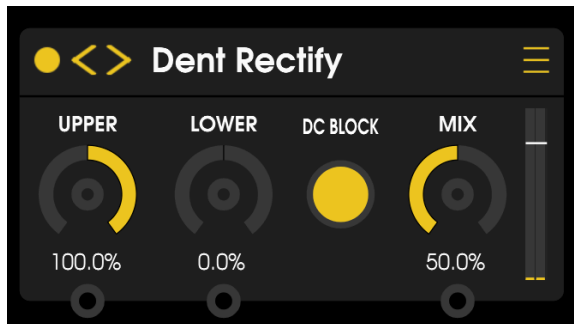
This is a bipolar rectifier taken from Dent's top row of distortion controls. An input signal is split into its positive and negative components. These components can then be manipulated separately. By setting one slider to 0%, you can achieve half-wave rectification. By setting one slider to -100%, you can achieve full-wave rectification.

Upper: Sets the amplitude and polarity of all signal components above 0.

Lower: Sets the amplitude and polarity of all signal components below 0.

DENT SHAPE
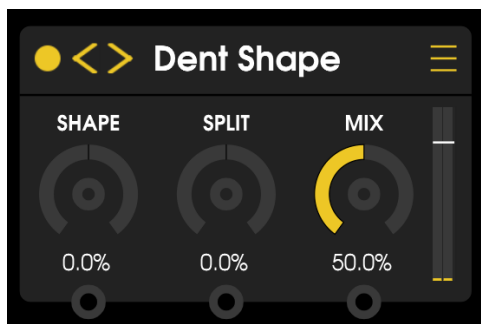
This effect uses the waveshaping algorithm from Dent. The SHAPE parameter takes a waveform and either round it upwards toward a square shape or downwards toward a needle shape. The SPLIT parameter is an aggressive control that adds symmetrical DC bias to the positive and negative components of the signal. This effectively adds a square wave to the signal.

Shape: Sets the intensity of the waveshaping effect. 12 o'clock is neutral. Clockwise will turn the signal into a squarewave, while counter-clockwise will turn the signal into a needle. Quieter signals will drop to silence when using the needle shape, so be sure to use a loud signal if you want to use the needle side.

Split: Adds DC bias to the positive and negative components of the signal in opposite amounts. 12 o'clock is neutral.

OVERDRIVE

This is a non-linear algorithm that boosts signals below a threshold and shapes signals above it.

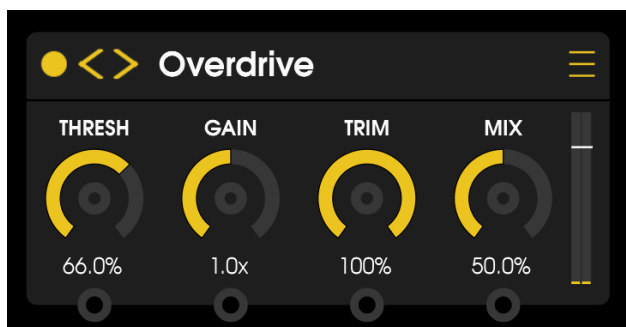**Threshold**: Adjust the threshold. Below the threshold, signals are boosted. Above the threshold, signals are shaped.

Gain: Sets the gain of the signal before it reaches the overdrive algorithm.

Trim: Reduces the signal after overdrive has been applied.

SATURATION
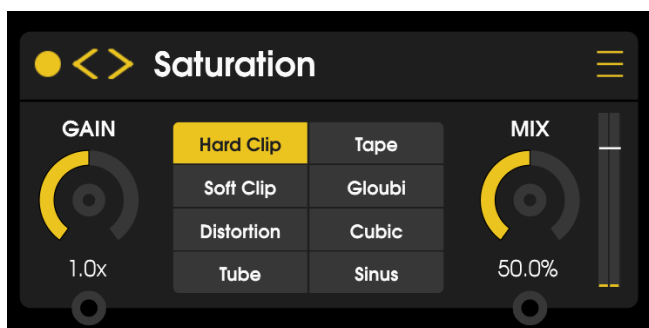
This effect contains a number of clipping and saturation algorithms for adding harmonic content to signals. These algorithms may sound similar on complex material (a drum mix, for example), but they have subtle differences that shine on simpler material (like synth waveforms). It's easy to experiment, so find one that works for your sound! As a note, if you are using this effect it is worth enabling oversampling in the global options menu.

Gain: Sets the amount of gain applied to the input signal before saturation.

Modes
Hard Clip: If the absolute value of the signal exceeds 1.0, it is simply clipped to that boundary. This is an aggressive, digital-sounding form of distortion.
Soft Clip: The signal is saturated using a parabolic equation. This will actually affect the sound of the entire signal in a subtle way. This clipping is more pleasant and analog sounding.
Distortion: A more traditional distortion algorithm. Excellent for guitars.
Tube: A partially rectified saturator that greatly attenuates negative components of signals.
Tape: A modified soft clipper with a parabolic equation that more closely matches the saturation behavior of tape.
Gloubi: 'Gloubi-boulga' waveshaper.

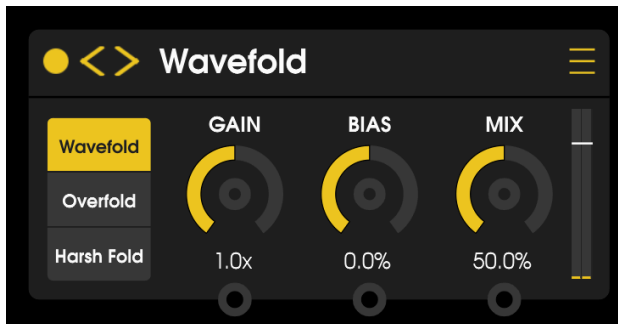**Cubic**: The signal is multiplied by itself three times.
Sinus: A saturation algorithm that shapes a signal using a sine without resorting to folding.

WAVEFOLD

This effect provides three different foldover distortion algorithms. Foldover distortion is a type of distortion that reflects loud signals around their boundaries. This type of distortion sounds especially good on simple harmonic material (like sine waves).

Gain: Applies gain to the input signal, effectively determining the number of folds.

Bias: Adds DC Bias to the input signal, changing the shape and symmetry of the folds.

Modes
Wavefold: Reflects a signal around its boundaries by running it through a sine equation.
Overfold: Combines the Wavefold algorithm with an Overdrive algorithm to provide a warmer sound for some signals.
Harsh Fold: Reflects the signal by using a quick-and-dirty algorithm that creates a much harsher, more digital sound.

WAVESET

This effect provides a basic implementation of waveset processing, a technique developed by Trevor Wishart. A waveset is a set of three zero crossings, typically one full cycle of a waveform (a single cycle of a sine wave, for instance). In this effect, an amount of gain is applied to selected wavesets. These wavesets can be chosen periodically or randomly.
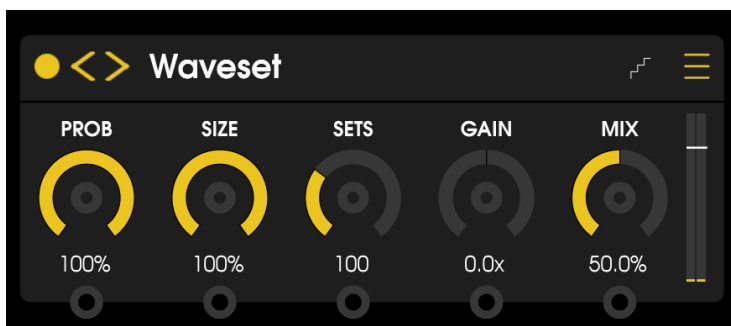
Probability: Sets the likelihood that a waveset within a group will be processed.

Size: Determines what percentage of sets in a group will be processed.

Sets: Determines how many consecutive wavesets appear in a group.

Gain: Sets the amount of gain that will be applied to a processed waveset.

Smooth: Activates an interpolator for gain changes. Even though the gain changes won't introduce a discontinuity, this will help soften up any harsh transitions.

DYNAMICS

AUTO COMPRESSOR

This is a small, intelligent compressor that takes a lot of guesswork out of setting the controls. It uses the auto-threshold algorithm and envelope shapes from Zip to quickly dial in the sound that you need.

Filter: Adds filtering before threshold analysis. This will not filter the output audio. Tilt it left to only duck on low frequency drum hits, for instance.
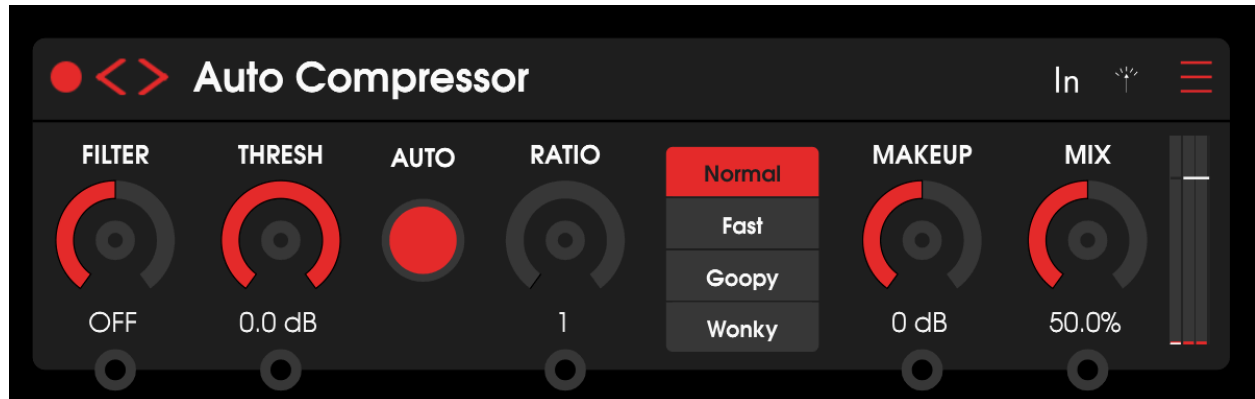
Threshold: Sets the threshold at which the compressor effect will be active. Signals louder than the threshold will be "compressed", or reduced in amplitude.

Auto Threshold: When enabled, the compressor tries to automatically adjust the threshold in real time to create pleasing results. Use the threshold knob in this mode to nudge it slightly up or down.

Ratio: Sets the intensity of the compressor effect.

Style: changes the speed and shape of the attack and release envelopes.

Styles
Normal: A typical compressor envelope shape.
Fast: A faster envelope optimized for transients.
Goopy: A slower envelope for a more vintage feel.
**Wonky**: An unusual envelope type that works well for lo-fi hip-hop.

Makeup: Sets the amount of gain applied to the signal after compression.

Peak: When enabled, the compressor's meter with use peak threshold detection. When disabled it will use RMS.

COMPRESSOR

This is an essential dynamics effect that reduces the amplitude of signals above a threshold. This reduces the dynamic range of the signal. This can be used to subtly improve a mix or to completely demolish a signal. For a simplified version of this effect, check out the Auto Compressor.
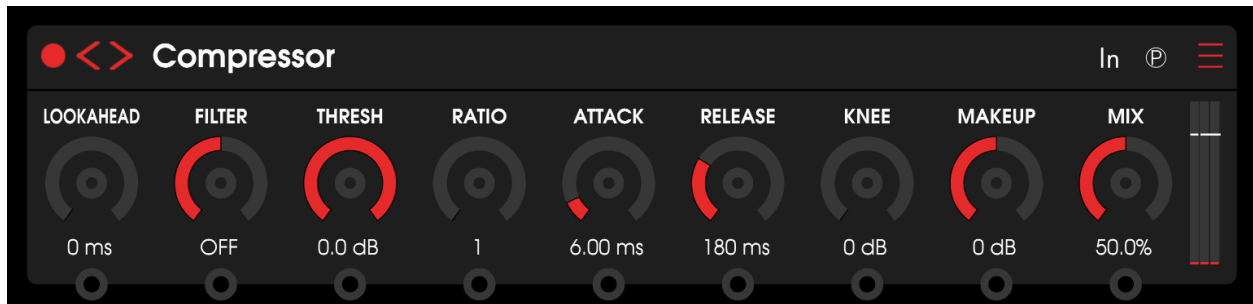
Lookahead: Sets the amount of audio that the compressor will analyze. This will add latency.

Filter: Adds filtering before threshold analysis. This will not filter the output audio. Tilt it left to only duck on low frequency drum hits, for instance.

Threshold: Sets the threshold at which the compressor effect will be active. Signals louder than the threshold will be "compressed", or reduced in amplitude.

Ratio: Sets the intensity of the compressor effect.

Attack: Sets the attack time for the compression envelope, or how quickly the compressor reaches a fully compressed state after the signal exceeds the threshold.

Release: Sets the release time for the compression envelope, or how quickly the compressor reaches a fully uncompressed state after the signal drops back below the threshold.

Knee: Sets the compressor's knee, or how sharp the transition is between the compressed and uncompressed amplitude curves.

Makeup: Sets the amount of gain applied to the signal after compression.

Peak: When enabled, the compressor's meter with use peak threshold detection. When disabled it will use RMS.

LIMITER
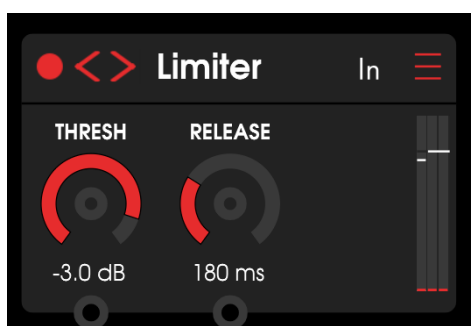
A limiter is essentially an extremely rapid compressor that keeps a signal from exceeding a threshold. This is traditionally used at the end of a mix chain to prevent the mix from clipping.

Threshold: Sets the maximum amplitude level of the input signal.

Release: Sets how slowly the limiter's amplitude envelope will release after the input signal drops below the threshold.

NOISE GATE

A noise gate is a useful utility that reduces signals below an amplitude threshold. Its most traditional usage is to remove background noise from a signal.

Lookahead: Sets the amount of audio that the noise gate will analyze. This will add latency.

Threshold: Sets the amplitude threshold below which the input signal will be gated.

Attack: Sets attack time. Note that the attack value means how quickly the gate opens up, so think about it as a way to change the attack envelope of transients as they go over the threshold.

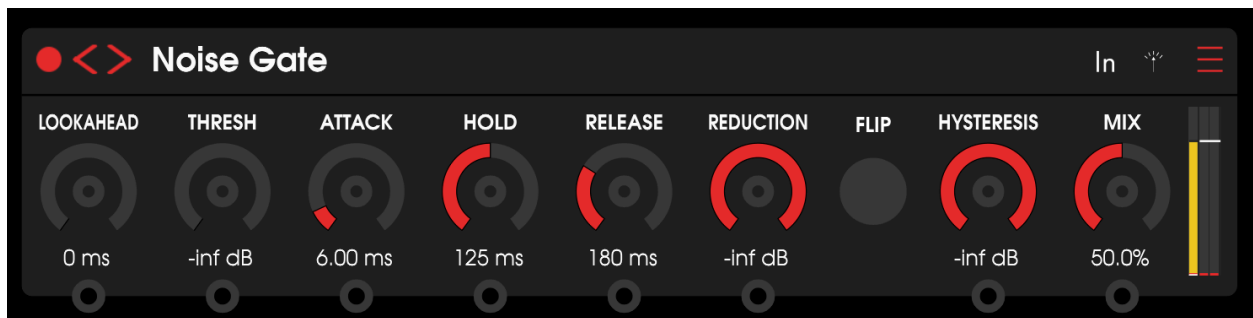**Hold**: Sets hold time. This is the minimum amount of time that the noise gate will stay open for.

Release: Sets release time. Note that the release value means how quickly the gate closes when the signal goes below the threshold.

**Reduction**: Sets the amount that the signal will be reduced when the gate is active.

Flip: When enabled, the gate's threshold detection will be inverted so only sounds above the threshold will be gated.

Hysteresis: Sets hysteresis amount. This sets a safety margin around the threshold to prevent rapid jitter effects.

Peak: When enabled, the gate's meter with use peak threshold detection. When disabled it will use RMS.

F I L T E R S

BASIC FILTER
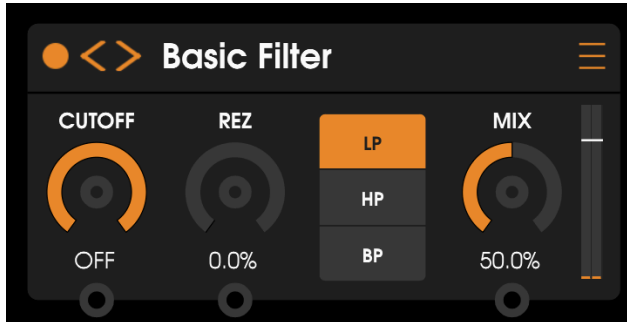
This is a simple, traditional two-pole multi-mode filter. It is used to reduce selected frequency bands in the incoming signal.

Cutoff: Set the cutoff frequency of the filter.

Resonance: Sets the filter's resonance.

Modes
LP: Low-pass. Removes high frequencies.
HP: High-pass. Removes low frequencies.
BP: Band-pass. Permits a small range of frequencies.

COMB FILTER

A comb filter is a delay line tuned to extremely short buffer times. It can be used to add a metallic timbre to a signal. This particular comb filter has an extremely wide frequency range and can shift quickly between being an echo effect and a timbre effect.
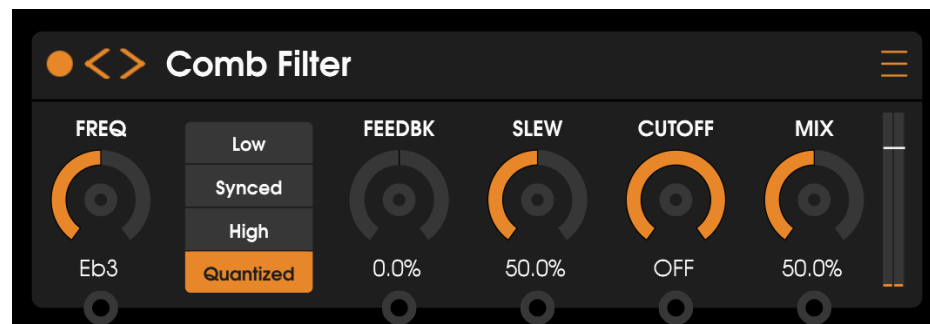
Freq: Sets the frequency of the comb filter.

Range: Sets the possible range of frequencies for the Comb Filter.

Feedback: Changes the feedback level of the comb filter. High values produce a string-type effect. Negative values provide phase cancellation effects.

Slew: Set how quickly the comb filter reaches new frequency values. At low values, the comb filter will travel quickly, but there could be clicks or pops. At high values, it will sound more like a tape delay.

Cutoff: Changes the cutoff of the lowpass filter applied to the output signal. This can help shape the feedback in more pleasant ways.

LADDER FILTER
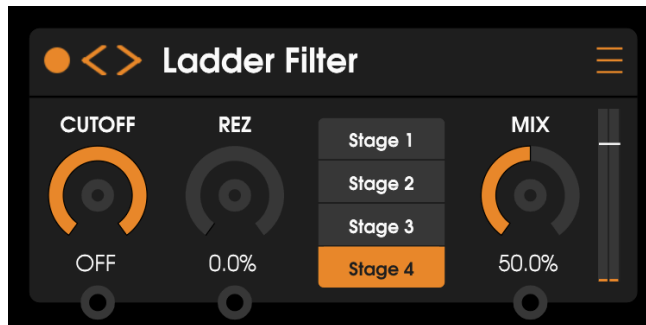
This is a multi-stage low-pass filter with a lot more character than the Basic Filter. It is based on a famous hardware synthesizer filter. Changing the active stage will change the slope and resonance of the filter.

Cutoff: Set the cutoff frequency of the filter.

Resonance: Sets the filter's resonance.

Stages: This determines the overall slope of the filter. Higher stages produce a sharper cutoff slope and deeper filtering.

ONE POLE FILTER

This is an extremely gentle filter. It is useful for taming frequencies without imparting a lot of character or color.

Cutoff: Set the cutoff frequency of the filter.

Modes
LP: Low-pass. Removes high frequencies.
HP: High-pass. Removes low frequencies.

RESONATOR BANK

This is an effect that combines four parallel comb filters. Each comb filter can have a separate frequency and amplitude. This is a great way to add chordal sounds to any material, even inharmonic signals like percussion loops. If you want more control over a single Resonator, look at the separate Comb Filter effect.

**Decay**: Determines how resonant the filters are.

Freq 1-4: Sets the frequency of the chosen resonator.

Gain 1-4: Sets the gain of the chosen resonator.

GRANULAR

GRANULATOR

This is a massive, sophisticated effect that takes incoming audio and breaks it into individual grains (or small chunks of sound). These grains can be manipulated in many ways, creating easy methods for altering a signal's pitch, timbre, speed, or stereo image.
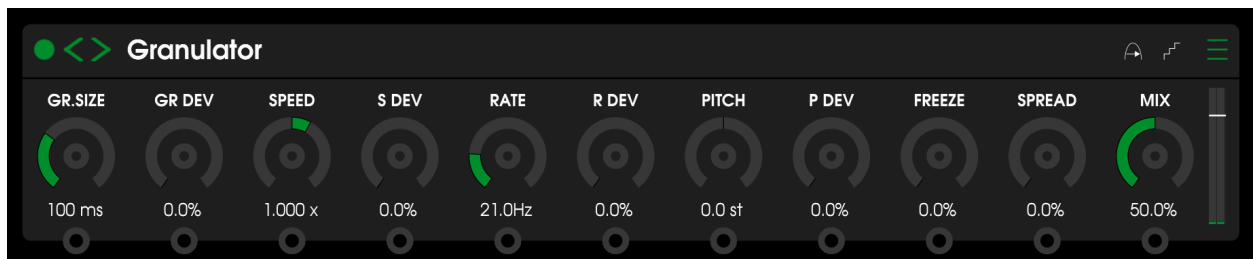
Grain Size: Sets the size of each generated grain.

Grain Size Deviation: Sets the amount of randomization for the size of each grain.

Speed: Sets how quickly each grain plays back its chunk of audio. Negative values play the chunk backwards.

Speed Deviation: Sets the amount of randomization for the playback speed of each grain.

Alternate Speed Mode: When active, granulator speed will stay positive when you go below 0.0 but the pitch speed will be reversed.

Rate: Sets how frequently new grains are created.

Rate Deviation: Sets the amount of randomization for grain generation. At 0%, this is synchronous granular synthesis.

Pitch: Sets the pitch of each grain.

Pitch Deviation: Sets the amount of pitch randomization for each grain.

Quantize: When active, the Pitch control will be locked to integer semitone values.

Freeze: Sets how much new audio gets written to the grain buffer. At 0%, only new audio will be recorded. At 100%, only old audio will be recycled.

Stereo Spread: Increases the stereo image of the granulator by randomly placing each grain in the stereo field. At higher values, an additional pitch warble will be introduced.

PITCH SHIFT
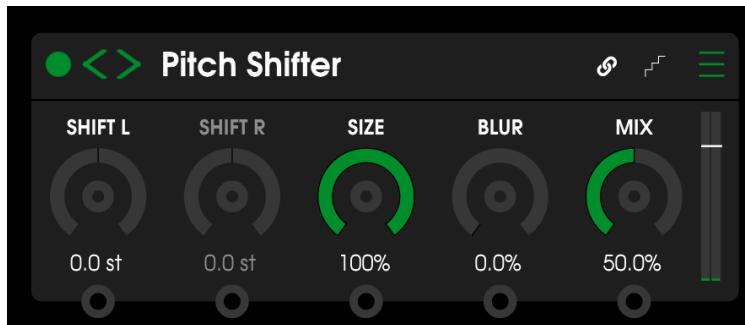
This effect uses a granular algorithm to change the perceived pitch of an input. The input signal is broken down into grains. These grains are then played back faster or slower than the original signal.

Shift L/R: Sets the pitch shift amount for the associated channel.

Link: When active, the value of the right channel's pitch shift will follow the left channel's.

Quantize: When active, both pitch shift parameters will always equal integer semitone values.

Size: Change the size of the grains used for the pitch shift effect, from .1-100 ms. Note that this effect introduces uncompensated latency.

Blur: Randomize grain times to reduce metallic pitch shifting sounds.

PULSAR

This effect changes the amplitude of a sound using a probabilistic grain window. It can be thought of as the granular equivalent of a Tremolo or Ring Modulator. A sinusoidal grain window is generated at PERIOD Hz and is shaped by FORMANT. At low rates, this is a fun, probabilistic panner. At high frequencies, this can impart unusual timbres and stereo width to droning timbres.
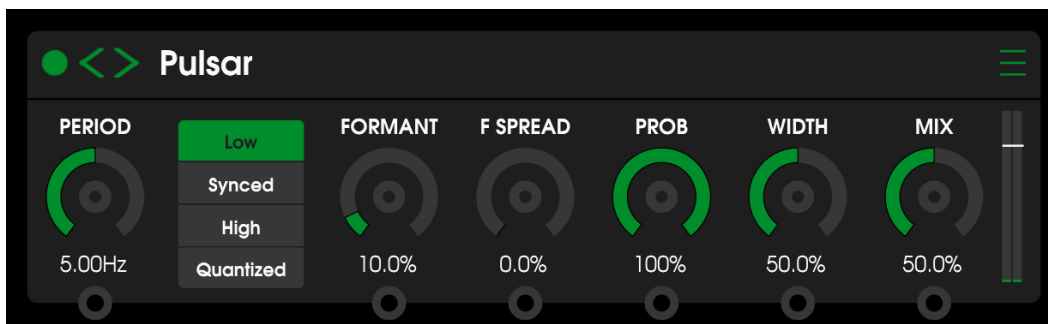
Period: Sets how frequently a Pulsar window is generated.

Range: Sets the frequency range of the pulsar oscillator.

Ranges
Low: 0-20 Hz.
Synced: The Pulsar's period is tied to even divisions of the host's tempo.
High: 0-1000 Hz.
Quantized: The Pulsar's period is tied to scaled note values.

Formant: Sets the width of the Pulsar's grain window.

Formant Spread: Sets the amount of variation in the Formant parameter per Pulsar cycle.

Probability: Sets the probability of a Pulsar being generated each cycle.

Width: Sets the stereo width of the Pulsar generators.

MIXING

3-BAND EQ

An equalizer is an essential tool for mixing. It allows for controlling the amplitudes of individual frequency bands. This equalizer is a quick DJ-style EQ that only breaks a signal into three bands. This is a modulation-friendly equalizer for quick, colorful changes instead of precision.
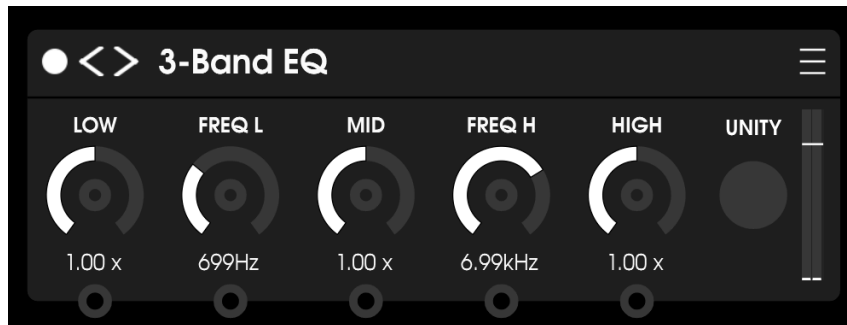
Freq L/H: Set the cutoff frequency for the low and high frequency bands.

Low/Mid/High: Set the levels of each frequency band.

Unity Gain: When this is enabled then gains will always be compensated to maintain an even value when the bands are summed. Changing the individual band gain will raise/lower the others to compensate.

MIX UTILITY
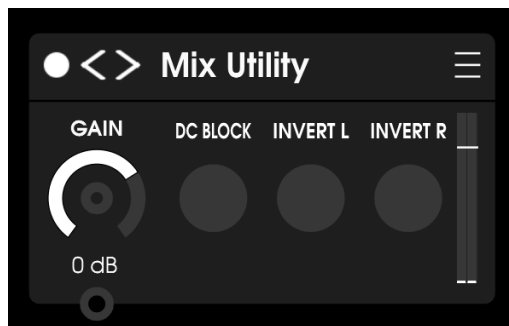
This is a very simple effect that provides some mix housekeeping operations. It is useful as the last effect in a chain.

Gain: Sets the amount of gain applied to the signal.

DC Block: Activates a DC Filter to remove DC offset from the signal.

Invert L/R: Inverts the polarity of the signal on the left or right channel.

PANNER
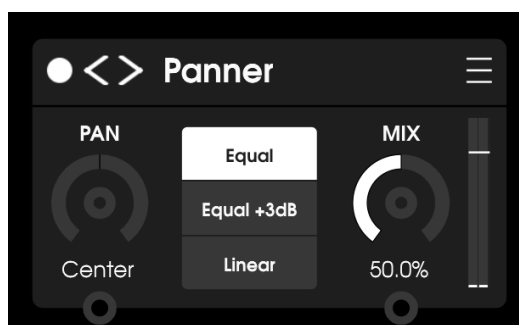
This effect will move a signal around within the stereo field. You can combine this with our modulation system to create a complex auto-panner.

Pan: Sets the position of the audio within the stereo field.

Algorithms
Equal: A standard panning method that keeps the signal at the same perceived loudness across the stereo field. However, it will lower the amplitude of the signal by about -3 dB.
Equal +3 dB: Equal +3dB: This is the same as Equal, but it provides a constant +3 dB boost to offset the gain loss.
Linear: A naive algorithm that changes channel amplitudes in opposite amounts. This causes an amplitude dip in the middle compared to the sides.

STEREO IMAGE
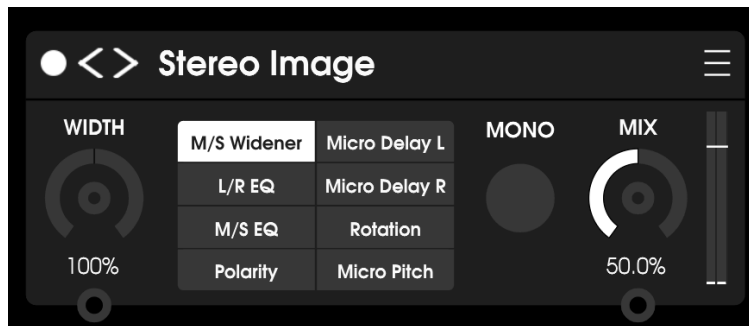
This is a collection of algorithms that are useful for boosting the apparent stereo width of a signal.

Width: Sets the intensity of the stereo effect.

Modes
M/S Widener: Converts a signal into its Mid-Side representation. 0% width = mono, 100% width = stereo, 200% width = super-stereo. This algorithm does not work on mono material.
L/R EQ: Applies opposite amounts of EQ to each channel.
M/S EQ: Applies opposite amounts of EQ to the Mid and Side components of a signal. This algorithm does not work on mono material.
Polarity: This adds an inverted amount of each channel to their opposite side. This does not work on mono material and is especially bad for mono mixdowns.
**Micro Delay L/R**: This applies a small amount of delay to the left or right channel. Watch out for phase cancellation effects in a mono mixdown.
Rotation: This uses an unusual algorithm that treats each channel as a point in space. It then rotates these points around a circle. At 100% Width, the signal is unchanged.
Micro Pitch: This applies a micro pitch shift to the left and right channels in opposite amounts. The pitch shifted copy is summed with the centered original.

Mono: Preview what the stereo signal will sound like in a mono mixdown. These algorithms can have phase cancellation side effects in a mono mixdown, so it is useful to monitor frequently.

M O D U L A T I O N

Each Modulation effect shares a common set of frequency controls:

Frequency: Sets the speed of the effect's internal modulator.

Range: Sets the frequency range of the internal modulator.

Ranges:
Low: Varies for each effect. Typically 0-20 Hz.
Synced: The modulation frequency is tied to even divisions of the host's tempo.
High: Varies for each effect. Typically 0-1000 Hz.
Quantized: The modulation frequency is tied to scaled note values.

CHORUS

This is an effect that creates a rich sound by summing a signal with detuned copies. The detuned copies are created through the modulation of delay lines.
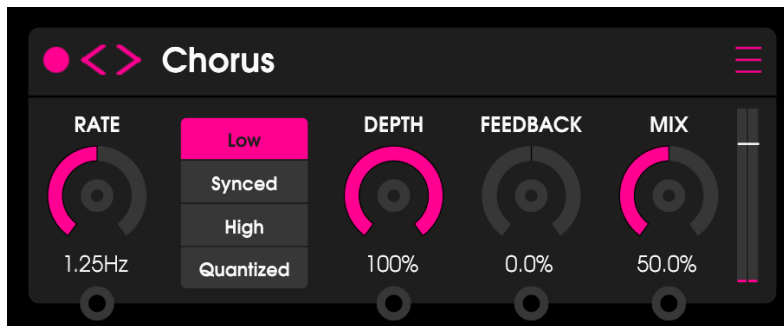
Depth: Sets the intensity of modulation.

Feedback: Sets how much of the wet output is sent back to the chorus input.

FLANGER

This is an effect that creates a characteristic "jet" effect by summing a signal with a modulated, delayed copy of itself.

Delay: Sets a delay buffer offset for the internal modulator. With the modulator's frequency set to 0, this can be manually modulated by the modulation system to create your own flanger!

Depth: Sets the intensity of modulation.
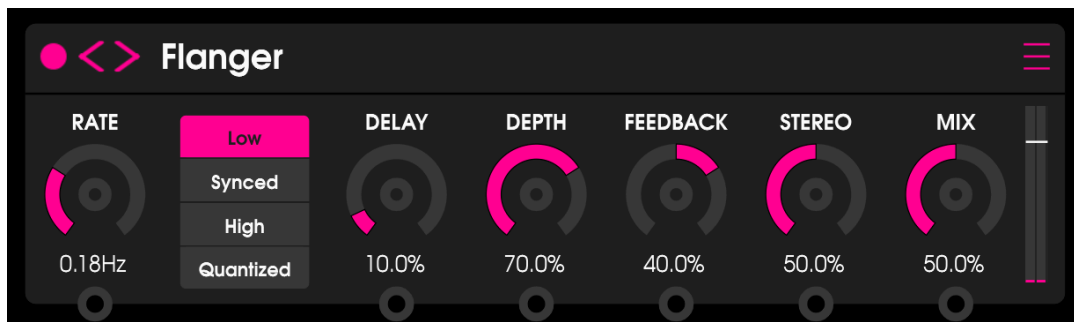
Feedback: Sets how much of the flanger's output is sent back to its input.

Stereo: Determines the difference of phase shift between the left and right channels.

FREQUENCY SHIFTER
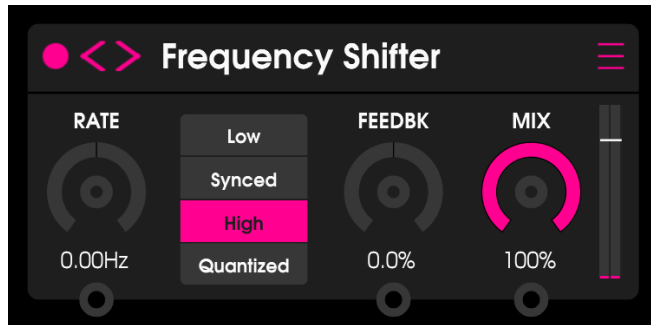
This is a complex effect related to amplitude modulation (audio-rate tremolo) and ring modulation. With a frequency shifter, a user has separate control over upper and lower sidebands. The effect causes a signal to shifts upwards or downwards in frequency. The signal shifts in a linear fashion, causing a very inharmonic sound when compared to pitch shifting.

Feedback: Sets how much of the frequency shifted output is sent back to the input.

PHASER

This is a complicated effect that creates modulated phase shifts through the use of multiple serial all-pass filters. The original signal is summed with the output of the serial filter chain, creating an intentional phase cancellation. This is a great way to add a lot of stereo interest to a sound.
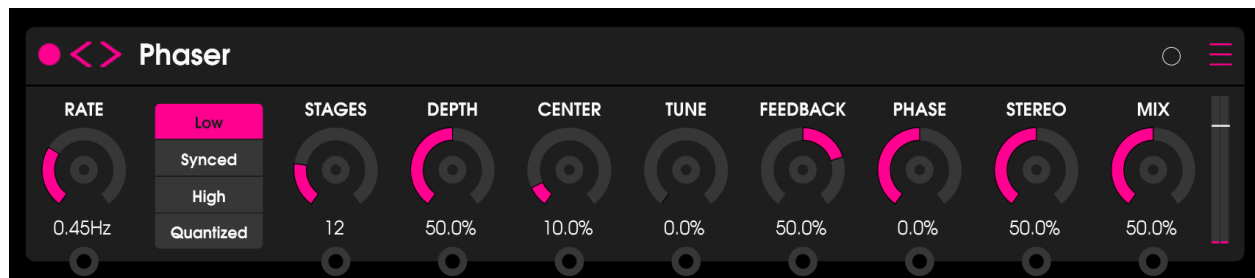
Stages: Sets the number of serial all-pass filters used to create the phasing effect.

Depth: Sets the frequency range of the internal all-pass filters.

Center: Sets the center frequency of modulation for the all-pass filters.

Tune: Sets the base frequency of the all-pass filters. At 0%, every stage will be tuned in an identical manner. At 100%, each filter will be spaced out at a harmonic interval. Try combining this with high feedback values for basic modal synthesis.

Feedback: Sets the amount and polarity of the feedback.

Invert: When active, the polarity of the phaser will be flipped.

Phase: This control sets the phase of the internal LFO. If the LFO is set to 0 Hz, this can be used as a manual control to create your own phaser (try modulating with a different LFO). Note that you will still need to use the DEPTH control to set how audible the phase shift is.

Stereo: Determines the difference of phase shift between the left and right channels.

RING MODULATOR
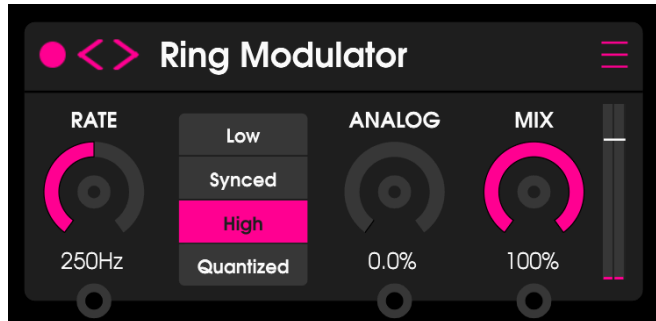
This is a classic, inharmonic timbre effect that is achieved by multiplying a signal with another signal, typically an oscillator. This ring modulator provides both the typical digital multiplication version along with a version that emulates the diode ring in an analog unit.

Analog: Changes the behavior of the ring modulator to sound closer to an analog diode-based modulator.

ROTARY SPEAKER
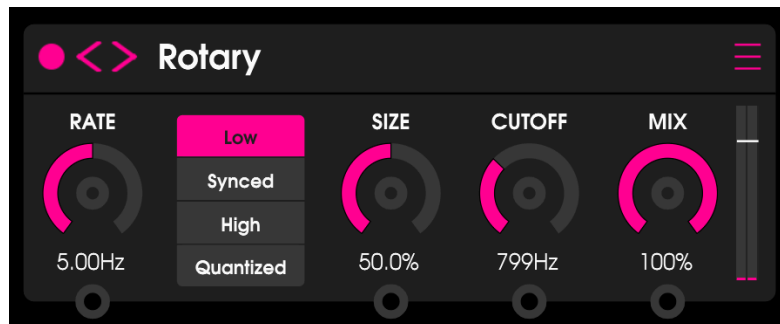
This is an emulation of a classic hardware effect that rotated a speaker rapidly inside of a cabinet.

Frequency: Sets how quickly the simulated speaker rotates.

Size: Sets the apparent size of the rotation.

Cutoff: Sets the split frequency between the two speaker rotors. Signals above this frequency are sent to one rotor, while signals below this frequency are sent to the other. This can be used to shape the timbre of the effect.

TREMOLO
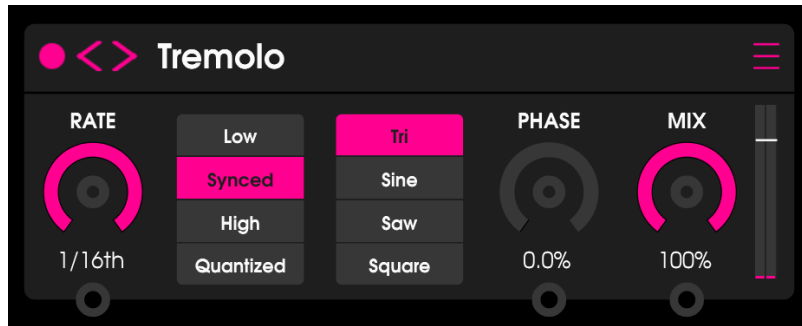
This is a very simple effect that is created by periodically modulating the amplitude of a signal.

Waveform: Sets the waveform used by the internal LFO.

Phase: Sets the initial phase value of the internal LFO. When restarting your project's playback, the LFO will reset to this phase.

VIBRATO

This is a simplified version of the Chorus effect. Instead of creating multiple detuned copies of a signal, this provides only one detuned copy.

Depth: Sets the intensity of modulation.
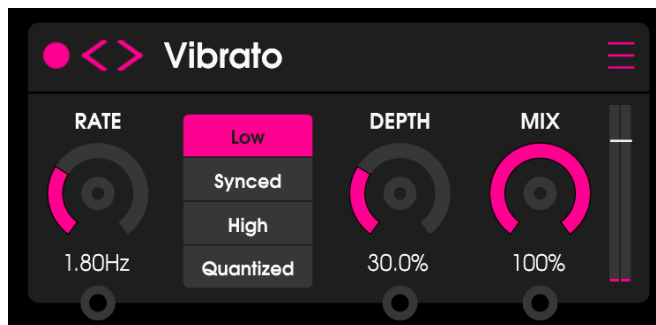
R E V E R B S

DEEP REVERB

This is a rich, powerful reverb sound that works well on just about anything. It is especially capable of producing long, frozen tails that sustain seemingly forever.
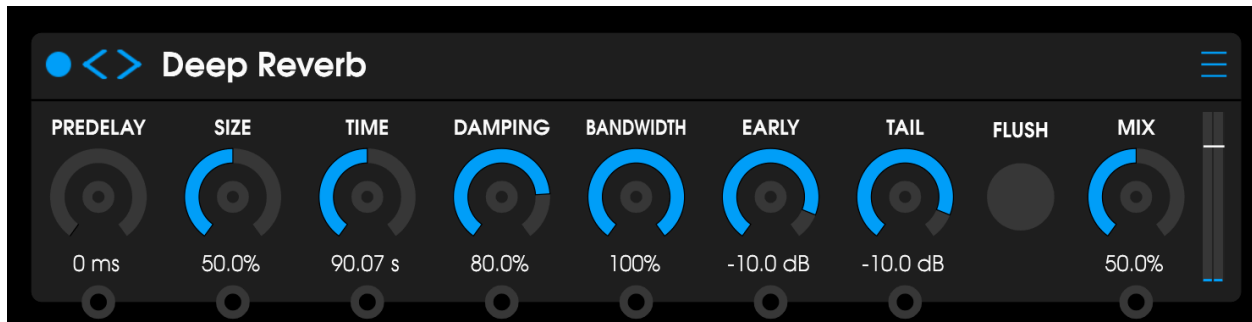
Pre-delay: Sets the amount of time the dry input is delayed for before being sent to the reverb algorithm.

Size: Sets the apparent room size.

Time: Sets the overall length of the reverb.

Damping: Sets the intensity of low-pass filtering applied to the echoes. Higher values produce a darker sound.

Bandwidth: Sets the reverb's bandwidth, or how much low-pass filtering is applied to the tail. Lower values will tighten up the echoes by removing the more diffuse tail.

Early: Sets the amplitude of the early reflections.

Tail: Sets the amplitude of more diffuse echoes occurring later in the reverb.

Flush: Clears the reverb buffer.

LO-FI REVERB

This is our take on a Schroeder Reverb, the digital reverb algorithm used by many famous computer music pioneers. While these algorithms have a bit of a metallic sound compared to modern reverbs, they excel when applied to synthetic sounds. We have included a large number of tunings to experiment with.
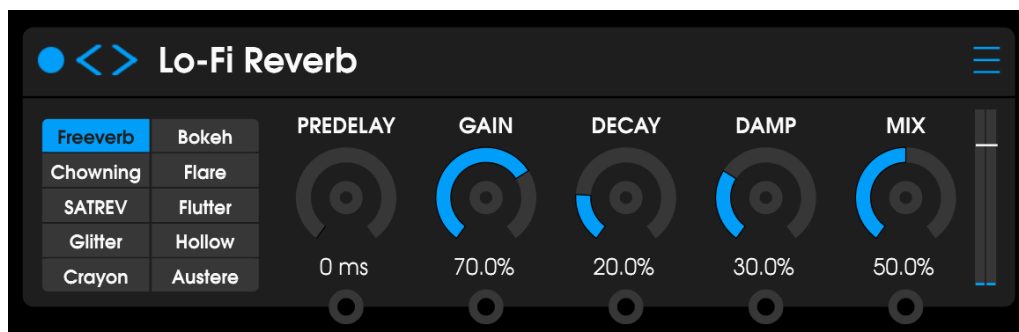
Pre-delay: Sets the amount of time the dry input is delayed for before being sent to the reverb algorithm.

Gain: Sets the amount of gain applied to the input before being sent to the reverb algorithm.

Decay: Sets how long the effect will reverberate for.

Damp: Sets the amount of low-pass filtering that occurs in the internal delay buffers. Higher Damping values will create a darker sound.

RENOUN REVERB

This reverb algorithm is taken from our first plugin, the Renoun Reverb Rack Extension for Reason. It is an emulation of a classic blue-box hardware reverb, but we've expanded the control set and ranges to more experimental extremes. You can use this for bright shimmer effects or noisy sci-fi modulation bursts.
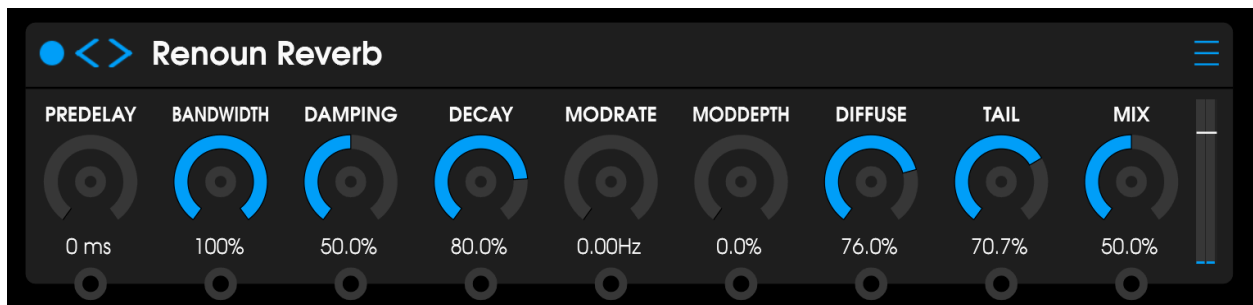
Pre-delay: Sets the amount of time the dry input is delayed for before being sent to the reverb algorithm.

Bandwidth: Applies low-pass filtering to the input. At 100%, the signal is untouched. At 0%, the signal is entirely filtered out.

Damping: Sets the amount of low-pass filtering that occurs in the internal delay buffers. Higher Damping values will create a darker sound.

Decay: Sets the overall length of the reverb.

Mod Rate: Sets how rapidly the internal delay lines are modulated.

Mod Depth: Sets the intensity of modulation for the internal delay lines.

Diffuse: Sets the amount of crosstalk that happens between each section of the reverb.

Tail: Sets the amplitude of later echoes.

SANDMAN DIFFUSE

This effect copies the algorithm behind Sandman Pro's DIFFUSE control. It is a simple all-pass network that works well when applied to a delay. Try placing this after Instant Delay for a more spacious sound.
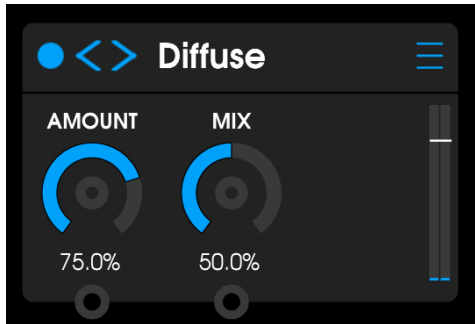
Amount: Sets the amount of feedback for the internal diffusers. Higher values will lead to more echoes.

Modulation System

Introduction

Triad utilizes Unfiltered Audio's flagship modulation system, with which you can quickly patch anything from simple parameter changes to complex and even self-evolving systems. Starting off is very easy due to the direct cable-routing user interface.

The first step is to reveal the modulation manager by clicking the "Modulation" button in the bottom corner of the interface. You'll see the plugin interface get taller, revealing modulation sources, outputs, and inputs on each modulatable plugin parameter.

Patching

Input and Output Ports

Outputs and inputs are both denoted by a simple circle. To avoid confusion, just remember that outputs are always located on the top left of each modulator. Everything else is an input.

Output ports have a small circular scaling knob below them which can be used to alter its calculated output value. This knob is set to 1.0 by default (fully clockwise), but can be turned down to 0.0 (center) to zero out its values entirely, or -1.0 (fully counter clockwise) to invert them.

Creating Connections

To create your first modulation "patch" simply click and drag on one of the outputs. You'll see a cable appear next to your mouse, the end of which you can drag to any input on the interface. When you get close enough you'll see the cable "lock" into place, showing you that the path is valid. Once you release the mouse, the connection completes and you'll see the modulation start to animate in real-time.

You can create a second connection from the same output source by clicking on it again. Likewise, try clicking and dragging from an input to an output, which works in the same fashion.

Moving or Removing Connections

In order to move a connection, click on one of the thicker ends of the cable, near the input or output port. You'll see the cable lock onto your mouse, allowing you to move it to a different input/output source. If you release the mouse while the cable is not "locked" onto any port, it will be removed entirely, which is how connections are deleted.

Inter-Modulation

As you might have noticed, all of the knobs on each modulator have input ports of their own, capable of receiving modulation just like parameters on the main section of the interface. Any output port can be connected to these inputs, meaning that a modulator can even modulate its own parameters! Experimenting with complex inter-modulation chaining can lead to dynamically evolving behavior, and even emergent systems.

Automation

We've removed our old clunky automation system and replaced it with a new Macro system. Check out the next chapter ("Macro System") to learn more.

Adding Modulators

You can add a new modulator by clicking the "+" button in the modulation header. At this time a maximum of 6 modulators can be used at once, although that is a somewhat arbitrary maximum that we plan to increase in the future.

Removing and Duplicating Modulators

Modulators can be removed or duplicated by right clicking on them and then selecting the appropriate option.

Mute Modulation

The "Mute All" button in the modulation row's sidebar allows you to quickly mute and un-mute all modulation sources.

Saving and Loading

All modulation routing, values, automation, and general state is saved with the preset and/or with the DAW's session. Presets are a great way to quickly share interesting modulation patches with other users.

Mute Modulation

The "Mute All" button in the modulation header allows you to quickly mute and un-mute all modulation sources.

Compact View

Double-click on a modulator's header to switch to a compact view. Double-click on the header again to make the modulator full-sized.

MODULATOR TYPES

You can change a modulator's type by clicking the dropdown menu next to its symbol. There are currently six types to choose from, each with its own characteristics and use case. Common parameters such as frequency will remain at their selected value, making the process of auditioning different modulator types both quick and easy.

LFOS: SINE/SAW/TRI/SQUARE

The LFO family of modulators all use common waveforms which are useful for classic "envelope" style parameter modulation.

The Amplitude knob is common to all of them, corresponding to modulation depth. This can be either Bipolar or Unipolar which is determined by the state of the +/- button in the modulator's bottom left corner. Bipolar modulation fluctuates between negative and positive, whereas unipolar modulation is always positive when the amplitude is above 0% and always negative when it is below 0%.

The Frequency knob corresponds to modulation speed. It is set in Hz (cycles per second) when in non-tempo-synced mode. Tempo Sync can be activated by clicking the "Hz" button on the left side of the modulator, which will turn into a musical note symbol indicating that it has been initiated. In tempo sync mode, modulation speed is synchronized to divisions of the DAWs global tempo setting- for instance setting the frequency to ¼ will correspond to quarter-note modulation cycles.

The Phase knob determines the value of the LFO at its starting point. The starting point occurs when the DAW's transport (playback) resets. The LFO can be manually reset with the trigger button on its left side, or via the reset modulation input below that. An input signal transitioning above zero here will trigger the reset.

The Saw/Tri modulator has an additional Shape knob which determines the slope of the triangular wave. At the default value of 0.5, you get a normal triangle wave. At 0.0, you get a downward sloping triangle wave and at 1.0 you get an upward sloping triangle.

The Square modulator has a Width knob that changes the pulse width of the square wave. At the default value of 0.5, you get a normal half-up, half-down square wave. As you approach 0.0, the wave will be almost entirely down, and conversely as you approach 1.0, the wave will be almost entirely up.

SAMPLE AND HOLD NOISE

The sample and hold noise modulator chooses random values at the chosen frequency rate. The Slew knob controls the amount of interpolation time whenever a new value is set. At a slew of 0.0 values change instantly, whereas at 1.0 they interpolate over the entire cycle period. Tempo Sync and Uni/Bipolar can be set on this modulator just like LFOs (see above).

LOOPING RANDOM

The Looping Random modulator is essentially a more advanced version of the sample and hold noise modulator that remembers the last few values generated and can loop or evolve sequences.

TRIG: Trigger input. If the button is clicked or the modulation input transitions to a positive signal, the sequencer's active stage will increase by one.

RESET: Reset input. This includes both a manual button and a modulation input. If the button is clicked or the modulation input transitions to a positive signal, the sequencer's active stage will reset immediately to the first stage of the sequence.

RAND: Randomize input. This includes both a manual button and a modulation input. If the button is clicked or the modulation input transitions to a positive signal, every value in the sequence will be randomized. If you set RAND to 0%, triggering this can provide a stream of new, looping sequences.

STEPS: Choose the number of steps in the sequence. To get a better understanding of how this works, try setting RAND to 0% and change this value.

RANDOM: Whenever a step is accessed, this determine the probability that the step will be a new, random value. At 0%, a perfect loop will occur. At 100%, a fully random stream of values will occur. Around 5-10% is a good setting for creating a steadily evolving sequence.

INPUT FOLLOWER

The input follower modulator allows you to easily "sidechain" parameters to the incoming audio stream. The Smooth knob determines the speed of the RMS meter used to track the incoming audio. Lower values correspond to a more instantaneous reading, while higher values smooth everything out.

META CONTROL

The Meta modulator can act as a manual control knob capable of outputting values to multiple parameters. This allows control over many parameters with a single gesture. Alternatively, you can patch many modulation sources into the amplitude knob's input port, turning the Meta control into a summing modulation bus.

STEP SEQUENCER

The step sequencer is a modulator that cycles through up to 16 manually chosen values.

T: "Trigger" input. This includes both a manual button and a modulation input. If the button is clicked or the modulation input transitions to a positive signal, the sequencer's active stage will increase by one.

R: Reset input. This includes both a manual button and a modulation input. If the button is clicked or the modulation input transitions to a positive signal, the sequencer's active stage will reset immediately to the stage set by the START control.

AMP: Sets the maximum amplitude of the modulation outputs.

FREQ: Sets the rate at which the sequencer automatically increments. With tempo sync off, this can be set to 0 Hz, making the sequencer only increment based on the state of the T input.

LENGTH: Sets the number of active steps in the sequence.

START: Sets the first stage in the sequence. This stage is selected when playback restarts or the Reset input is triggered.
SLEW: Determines how quickly the output values change. At 0.0, the modulation output is stepped. As SLEW increases, new values are smoothly selected, adding a "slide" effect between values.

ROLI LIGHTPAD

This modulator connects to the excellent ROLI Lightpad (https://roli.com/products/blocks) for use as a performance controller. Please note that because this is part of the modulation system, this modulator is not intended for writing automation. If you want to use the Lightpad for automation, you can use the ROLI Blocks Dashboard for loading the interface of your choice and sending MIDI CC.

To get started, connect the Lightpad to your computer using either its USB-C cable or Bluetooth. For instructions on setting up a Bluetooth connection, see the Lightpad's manual.

Once a Lightpad is connected to your computer, it should appear in the drop-down menu on this modulator. Select it and click "Connect". The interface should turn white. You are now in Location + Pressure Mode. To change the active mode, click the large control button on the side of the Lightpad.

LOCATION + PRESSURE MODE

In this mode, the modulator will track the position of one finger on the Lightpad along with its pressure. There are separate outputs for X, Y, Z (Pressure), and On (Touch Gate: on when a finger is touching the Lightpad, otherwise off).

SLIDERS

In this mode, the modulator will output the value of four vertical sliders. Due to the resolution of the Lightpad's screen, it may appear that there are only 15 possible values for the slider. However, the information that is read is higher resolution.

TOGGLES

In this mode, the modulator will output the value of four toggle switches.

QUAD PRESSURE

In this mode, the modulator will output four separate pressure values (one for each quadrant of the Lightpad).

New modulators will have descriptions and tooltips in the next edition of the manual

CHAOS

This modulator is similar to the Noise modulator, but its output is based on the Logistic Map. The Logistic Map is a chaotic algorithm that can oscillate in a predictable fashion (at low

CHAOS values) or generate wild, random, linear shapes. Unlike the Noise modulator, the output can frequently fall into cyclical patterns.

GATE SEQUENCER

This is a simplified version of the Step Sequencer. While the value of the Step Sequencer's outputs can be smoothly determined, the Gate Sequencer's outputs can only be true or false. This sequencer is a much quicker way to program rhythmic patterns. The T (Trigger) output is useful for triggering other modulators (like the AD Envelope or a Step Sequencer).

ADSR

This is the classic Attack-Decay-Sustain-Release envelope, perhaps the most common type of envelope seen on synthesizers. Upon the reception of a gate, the rising Attack stage is started. Once the Attack stage is complete, the falling Decay stage will lead to an amplitude set by Sustain. The Sustain stage will be active for as long as the input gate is true. Once the input gate goes low, the falling Release stage will bring the envelope down to zero.

AD ENVELOPE

This is a percussive Attack-Decay envelope. This envelope does not have a Hold/Sustain stage, so the length of the incoming gate or trigger does not matter. It generates a rising Attack stage that is followed immediately by a falling Decay stage. This type of envelope is frequently used for synthesizing drum hits.

PROBABILITY GATE

This is essentially a square wave LFO, but the state is determined randomly at the beginning of each cycle. For instance, if the cycle length is 1/8th, every eighth note the value of the output will randomly be high or low, the probability of which is determined by the PROB control.

COMPARATOR

This takes in another modulation signal and produces a gate. The gate is active when the modulation signal exceeds a threshold. As an example, you could use this to create a square wave from a sine wave. The pulse width of the square wave would be related to the Threshold setting.

XY PAD

This modulator provides a useful widget for controlling two outputs at once with a mouse.

SCOPE

This is a simple tool for visualizing your modulation signals.

SPECTRAL FOLLOWER

Like the Input Follower, this modulator uses the plug-in's audio input to derive a modulation signal. While the Input Follower only uses the input's amplitude, the Spectral Follower analyzes the input's frequency content.

Brightness: The follower output is related to the amount of high-frequency content in the signal. It is the inverse of Darkness.
Darkness: The follower output is related to the amount of low-frequency content in the signal. It is the inverse of Brightness.
Noisiness: The follower output is related to how much noise or unstable frequency content is present in the signal. It is the inverse of Tonalness.
Tonalness: The follower output is related to how harmonically stable the input signal is. It is the inverse of Noisiness.

LOGIC MIX

This modulator provides arithmetic combinations of two modulation inputs. It provides the sum, difference, and multiplication of the two signals. Additionally, > and < outputs will provide which input is the loudest or quietest of the two.

EXPONENT

This will change the shape of a modulation signal by raising it to a power. At 12 o'clock, the signal is raised to the first power and is unaffected (linear). Clockwise, the signal reaches its square and becomes more exponential. Counter-clockwise, the signal reaches its square root and becomes more logarithmic.
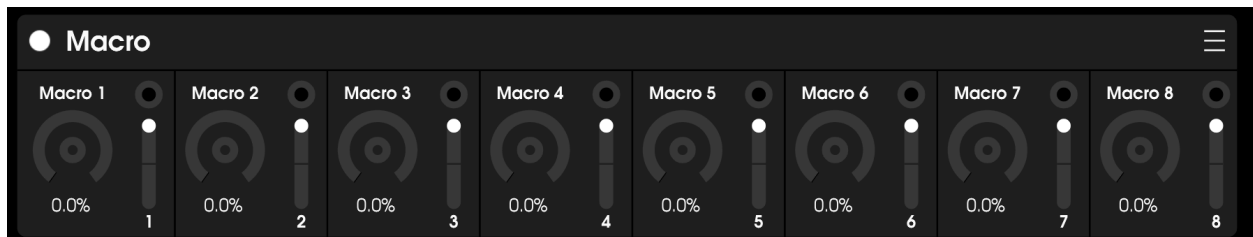
Macro System

Due to Triad's infinite nature, we needed to create a quick and easy system for automating any control. In previous Unfiltered Audio plugins, we used a right-click mapping system. This has been scrapped and replaced with a faster, more flexible Macro system.


The Macro Modulator is a modulator that is present in every preset. It is always the first modulator in the row.

When creating a new patch, you will only see the Macro 1 knob. Double clicking the Macro header will expand the Macro Modulator into "patching view". Here, you will see all 8 Macro knobs along with their outputs.

Each macro output behaves identically to the rest of the modulation system. Simply drag a cable from the Macro knob's output to the control that you wish to automate. Turning the Macro knob will then manipulate that control via the modulation system.
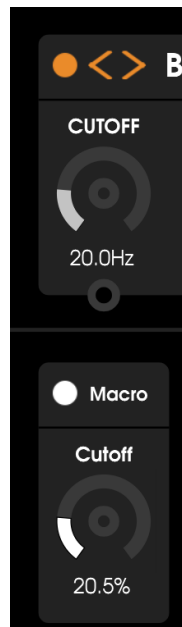
Two important things to keep in mind:
1) You can connect multiple modulation cables to the same input. In that situation, every connection will be added together.

2) You can create multiple cables from the same output. This means that you can use one Macro knob to modulate as many controls as you want. You can do something as simple as a one-to-one automation by connecting a Macro to a single knob, or do something as complex as a preset morph by using one knob to modulate every knob on the screen.

After creating your mapping, you can rename the Macro knob by double-clicking on its title (i.e. "Macro 1"). The title will be saved with your current preset. Due to limitations of many DAWs, the Macro's name will not show up in the DAW's automation list (instead, the parameter names are always named "Macro 1," "Macro 2," etc.).

When you have finished assigning Macros, double-click the Macro header to exit the patching view. Now, only the assigned Macros will be visible.

Options

To bring up the Triad option menu, click on the gear icon to the left of the preset menu.

Global Options
These options apply to all instances of Triad. Some options will not take effect until the interface is closed and re-opened.

Use Dark Skin: This is enabled by default. Triad has two skins. The dark skin is ideal for dim, indoor studio environments, while the light skin is intended for very bright areas.

Enable Tooltips: This is enabled by default. When enabled, hovering your mouse cursor over a control will show a pop-up that describes the function of that control.

Use OpenGL: This is enabled by default. When enabled, Triad's interface will be rendered using your computer's graphics hardware instead of the CPU. Depending on your setup, this can improve Triad's performance. We recommend that you update your computer's graphics drivers when using Unfiltered Audio plugins.

Per-Instance Options
These options only apply to the current instance of Triad and are saved with your DAW's project. These options are not saved with presets.

True BYOME Bypass: When enabled, bypassing a BYOME effect will also bypass the amount of gain set on the effect's VU meter.

Recipes and Ideas

Per-Band Widener

• Add a Stereo Image effect to multiple frequency bands. You can use this to make the bass frequencies more mono while widening the high-frequency bands.

Multi-Band Auto-Compression

• Add the Auto Compressor effect to each frequency band. Tweak each band individually, or use the Row Permutation feature to find surprising results.

Pitch Shifted Chords

• Create a Pitch Shifter effect. Use different pitch settings for the left and right channels.
• Set the Pitch Shifter to about 50% Mix.
• Add a Stereo Image effect after the Pitch Shifter.
• Select the M/S Widener algorithm and set the Width close to 0% to turn the signal towards mono.

Credits and Thanks

Triad is written by Joshua Dickinson and Michael Hetrick.

We'd like to thank:

Toby Pitman for creating almost all of the factory presets.

Benn Cooper

Mutable Instruments for many contributions to DSP education.

Lance Putnam for his flexible DSP library, Gamma.

Dirk Ulrich, Mo Volans, and Plugin Alliance for helping us to do this full-time and for providing us with a thorough QA team.

Ryan McGibben for coming up with the name!

To our awesome alpha testers, thank you for your feedback:
Hans Besselink (Oddiction)
Mathias Brüssel
BT
Benjamin Wynn (Deru)
——

Our families, for putting up with the constant stream of weird sounds coming out of our offices (and random stress over esoteric bugs).
http://www.unfilteredaudio.com