LION

Official Manual
by
unfiltered**audio**

LION

Default

unfilteredaudio

● <> **The Forms**   DRIFT   Unison: Off   Osc 1

TUNE   SUB   SIN TRI   BLEND   SAW SQR   PAN 1
Q   Q   S

0.0 st   0.0 st   50.0%   50.0%   50.0%   Center

<> **Linear XFade**   Mix

FADE   OSC 1   OSC 2

50.0%   100%   100%

● <> **The Forms**   DRIFT   Unison: Off   Osc 2

TUNE   SUB   SIN TRI   BLEND   SAW SQR   PAN 2
Q   Q   S

0.0 st   0.0 st   50.0%   50.0%   50.0%   Center

● <> **LP12**   INVERT   Filter

DRIVE   CUTOFF   RES

50.0%   20.0kHz   0.0%

!   Output

PAN   GAIN   BYOME

Center   0 dB   100%

FX

● <> **Renoun Reverb**

PREDELAY   BANDWIDTH   DAMPING   DECAY   MODRATE   MODDEPTH   DIFFUSE   TAIL   MIX

0 ms   100%   50.0%   80.0%   0.00Hz   0.0%   76.0%   70.7%   50.0%

+

MOD

● Macro

Macro 1

0.0%

≡ **MIDI Input**

12   12

AMP

25.0%

N   V   MW   PB   A   T   #   G   R

C3   C4   C5

⊙ **Master ADSR**

MIDI: All Channels   Voices: Poly 4   Legato: 0.0%   TIME   On: 2x   Off: 2x   Transpose: 0.0 st   Tuning: Default   WIDE   PRESETS

THE BASICS

LION is an extremely versatile synthesizer with a very simple signal path. The LION voice is not much different from a traditional subtractive synthesizer: Two oscillators are mixed together and run through a filter.

Don't let that simplicity fool you, though! LION's mixer alone contains more timbral complexity than many synthesizers, and that comes after LION's flagship feature: each of LION's two oscillators contain over 20 modes, each with an option Stereo Mode and full Unison support.

After each voice is synthesized, the voices are mixed together and sent to our famous BYOME processing engine. Our BYOME row contains 40 effects, each designed for quick operation and deep modulation.

Of course, our signature modulation system is also present, and it now supports polyphony! This means that each voice can be independently modulated. For instance, if you attach an LFO to an oscillator's pan control, you can either pan every voice simultaneously with a monophonic LFO, or pan each voice with independent phase using a polyphonic LFO.

Anatomy of a LION Voice

LION Signal Flow
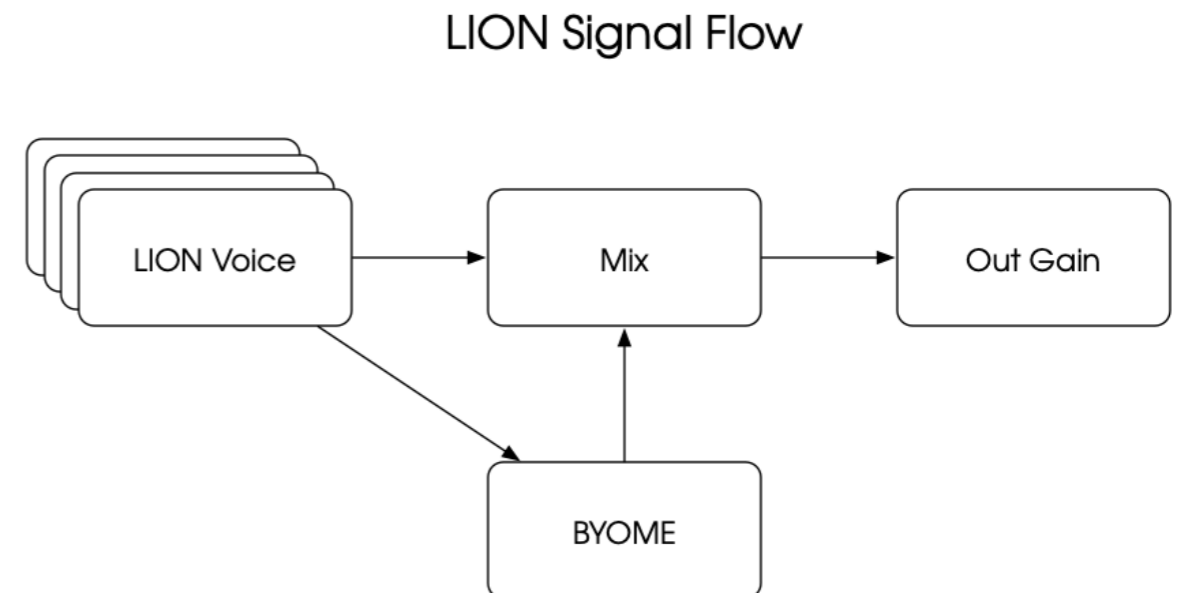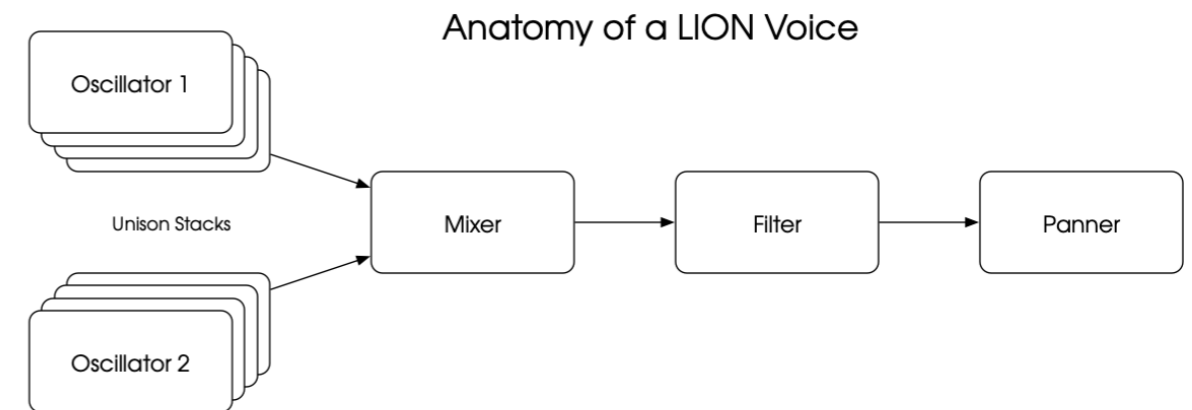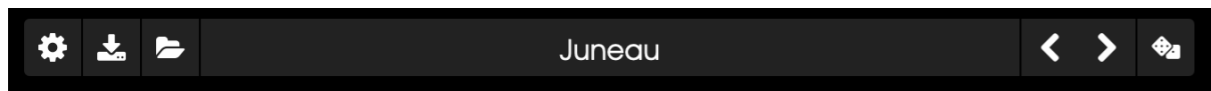
Juneau

Before venturing into LION's synthesis jungle, it might be worth exploring the rich preset ecosystem that comes with it. The Preset Bar contains a number of controls for exploring and randomizing these presets.

Clicking the preset name ("Juneau" in the example above) will bring up a list of all factory presets. These are organized by style or by signature artist. To quickly skip through presets, you can click the arrows next to the preset name.

Clicking the Dice icon will randomize the current preset. By default, each control can wander by a maximum of 20% of the knob. Right-clicking the Dice icon brings up the Randomization Menu.
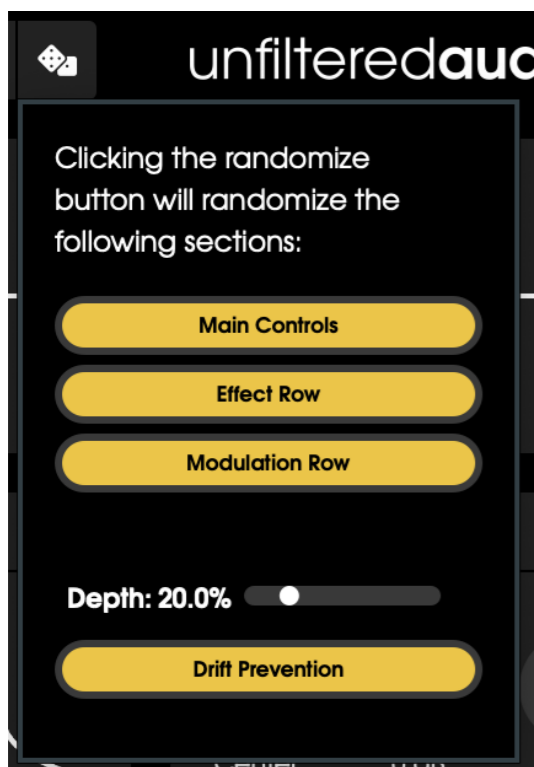

Here, you can decide which sections to randomize, along with how deep the randomization is. Toggling "DRIFT PREVENTION" will keep the knobs from wandering too far past their original values.

Additionally, if you want to prevent individual controls from being randomized, you can right-click the control to Lock it.

Clicking the Gear icon will bring up LION's Options Menu. Here, you can choose whether LION uses a Light or Dark Skin.

When learning LION, you should keep Enable Tooltips on. This will pop up brief hints about a control by hovering your mouse over it
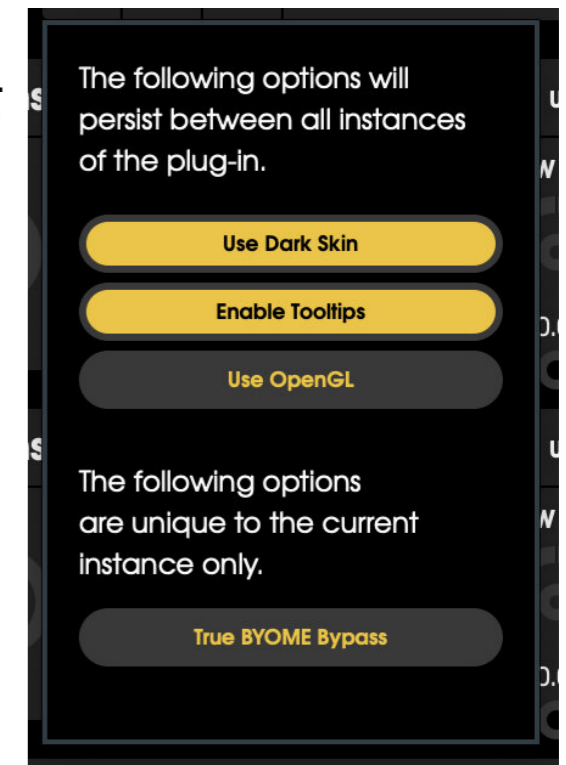
OpenGL is a graphics library that helps speed up LION's interface on most systems. If you are experiencing weird graphics artifacts or slow-down, try disabling this option and re-opening LION's interface.

"True BYOME Bypass" determines what happens if you mute a BYOME effect. With the original behavior (button off), bypassing a BYOME effect would preserve its output gain setting, preventing a bypass from dramatic gain differences. By enabling True BYOME Bypass, bypassing an effect will also bypass its output gain.

MIDI: Selects which MIDI channel LION listens to. Additionally, you can select MPE mode here if you are using an MPE Controller (like ROLI Seaboard or Sensel Morph).

Legato: Determines how long it takes to slide between notes in Legato mode. In TIME mode, this is the same amount of time between any two notes. In RATE mode, the time is also affected by the distance between the notes.

Transpose: This is the master Transposition control. It will change the tuning of both of LION's oscillator sections. You can use this to quickly change the key of LION, or to match an instrument that's slightly out of tune.

WIDE: Activate this to expand LION's interface. This is useful if you have a lot of modulators and/or BYOME effects.

MIDI: All Channels   Voices: Poly 4   Legato: 0.0% ●   **TIME**   On: 2x   Off: 2x   Transpose: 0.0 st ◖◗   Tuning: Default   **WIDE**   **PRESETS**

Voices: Select how many simultaneous notes LION is capable of playing. Monophonic and Legato modes only have one active voice. In Legato mode, there is an audible slide between notes when old notes are held.

On/Off: This determines how much oversampling is performed with LION. Oversampling will reduce aliasing artifacts, but it will also greatly increase CPU usage. The On setting is active during typical use. The Off setting is used for Offline Rendering, or when your DAW is rendering a track for instance.

Tuning: This brings up the custom tuning menu. You can load any TUN file to introduce a custom microtuning to LION.

PRESETS: Click this to bring up the new Preset Browser with tagging and metadata support.

LION OSCILLATORS

LION OSCILLATOR CONTROLS

Power - Enable or disable the oscillator

Arrows - Cycle through various oscillator modes

Title - Click this to bring up the oscillator list

DRIFT - When enabled, the oscillator's tune will vary slightly over time, emulating analog behavior.

Unison - Selects the number of active Unison stacks. Each stack is a copy of the oscillator where each parameter can be slightly different. This can create a very thick, powerful sound.
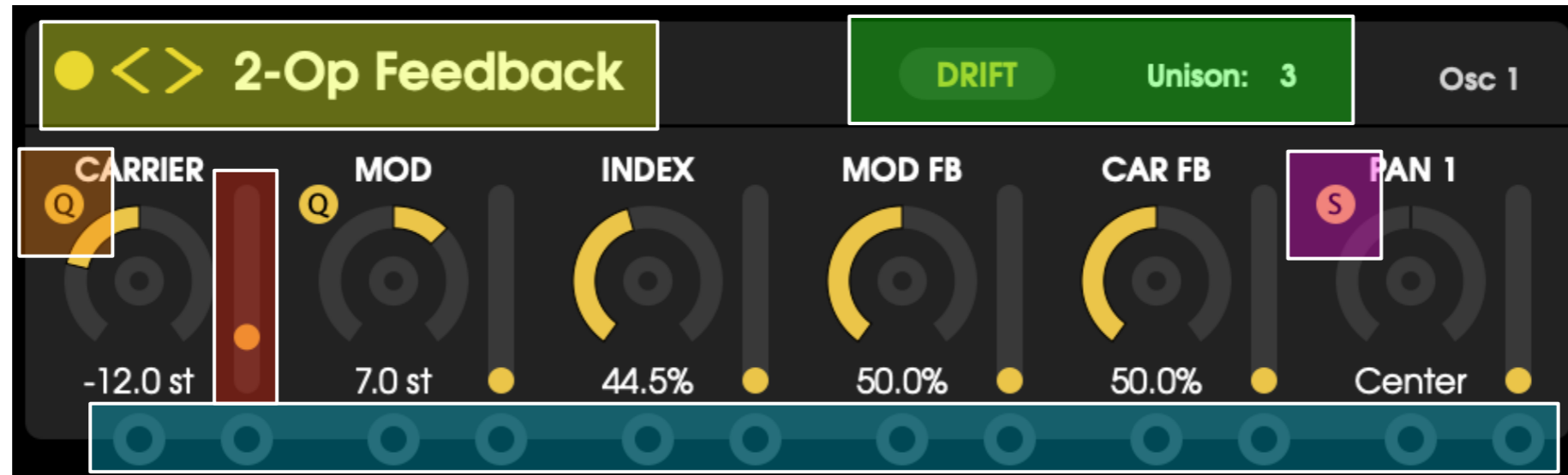
Modulation Ports - Attach a modulation cable here to modulate the control above it

Stereo Mode - When enabled, the oscillator will produce a wider signal using techniques specific to each oscillator mode.

Unison Depth - When Unison is enabled, these sliders will set the amount of variation per unison stack

QUANTIZE- When clicked, this will bring up a list of scales. When a scale is selected, the associated tuning knob will snap to semitone values that are part of the selected scale.

CLASSIC

THE FORMS

This mode covers all of the traditional waveforms in subtractive synthesis. There are two oscillators (Main and Sub), each with independent tuning and waveforms. Since everything is based around crossfading, it is very modulation friendly.

STEREO MODE

In stereo mode, two more oscillators are introduced, but the tuning controls are swapped for the additional pair. The Sub frequency controls the Sine/Tri oscillator, while the Main frequency controls the Saw/Square oscillator.

CONTROLS

|	
TUNE	Changes the primary frequency of the oscillator.
SUB	Changes the frequency of the sub oscillator as an offset from the Main frequency.
SIN TRI	Changes the waveform of the primary oscillator from sine to triangle.
BLEND	Crossfades between the primary oscillator and the SUB oscillator.
SAW SQR	Changes the waveform of the SUB oscillator from sawtooth to square.
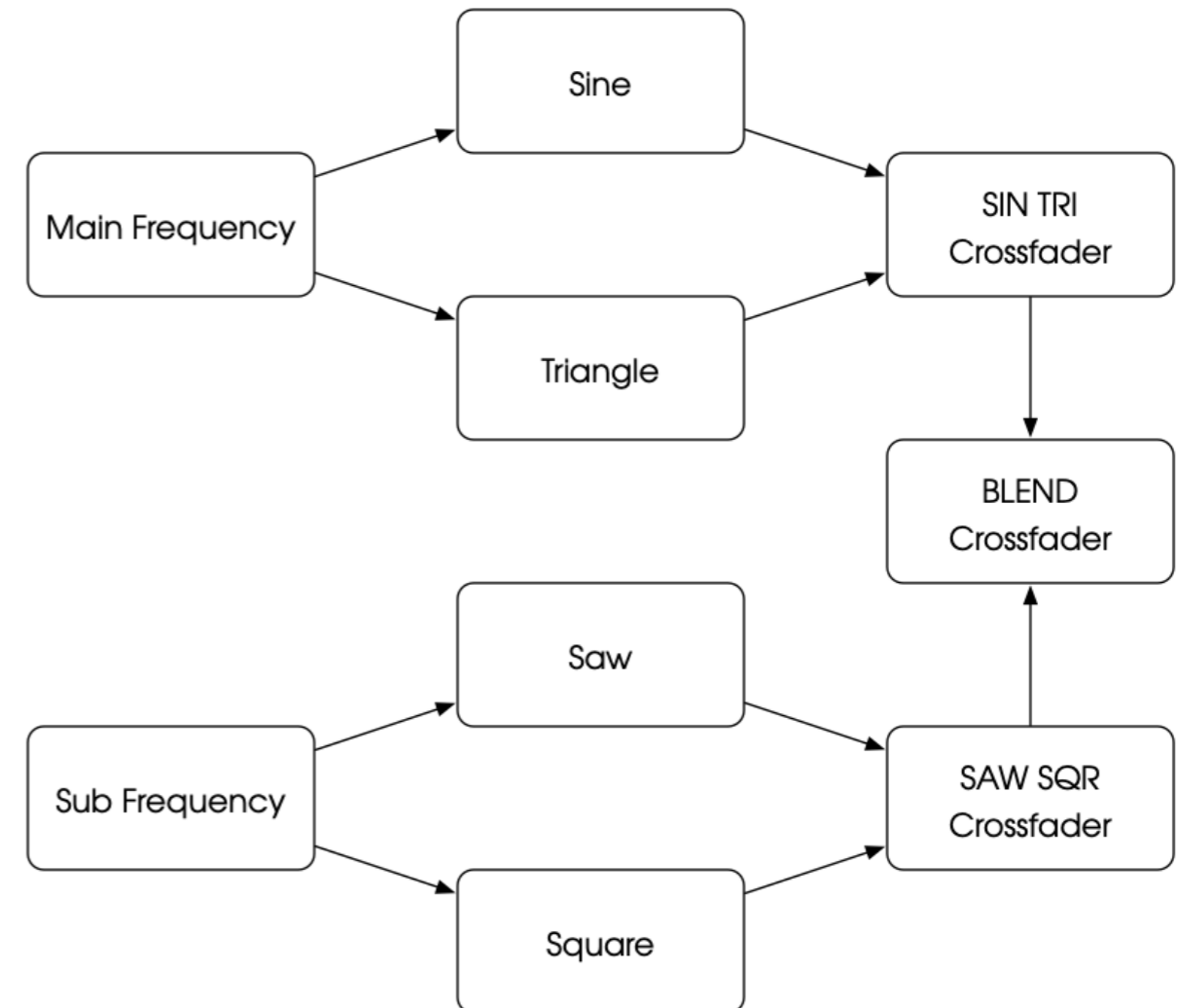
RINGO

This simple algorithm takes three sine waves and ring modulates them together. Despite its simplicity, it can create a wide range of timbres that compare to FM synthesis.

STEREO MODE

The third sine oscillator is used to pan the final waveform around the stereo field.

CONTROLS

|	
SINE 1	Sets the frequency of the first sine oscillator.
SINE 2	Sets the frequency of the second sine oscillator as an interval from the first.
SINE 3	Sets the frequency of the third sine oscillator as a percentage of the first.
ANALOG	Crossfade between a digital multiplication ring modulator model and a more analog one based on diode simulation.
SHAPE	Applies a Dent waveshaper to the final waveform mix.
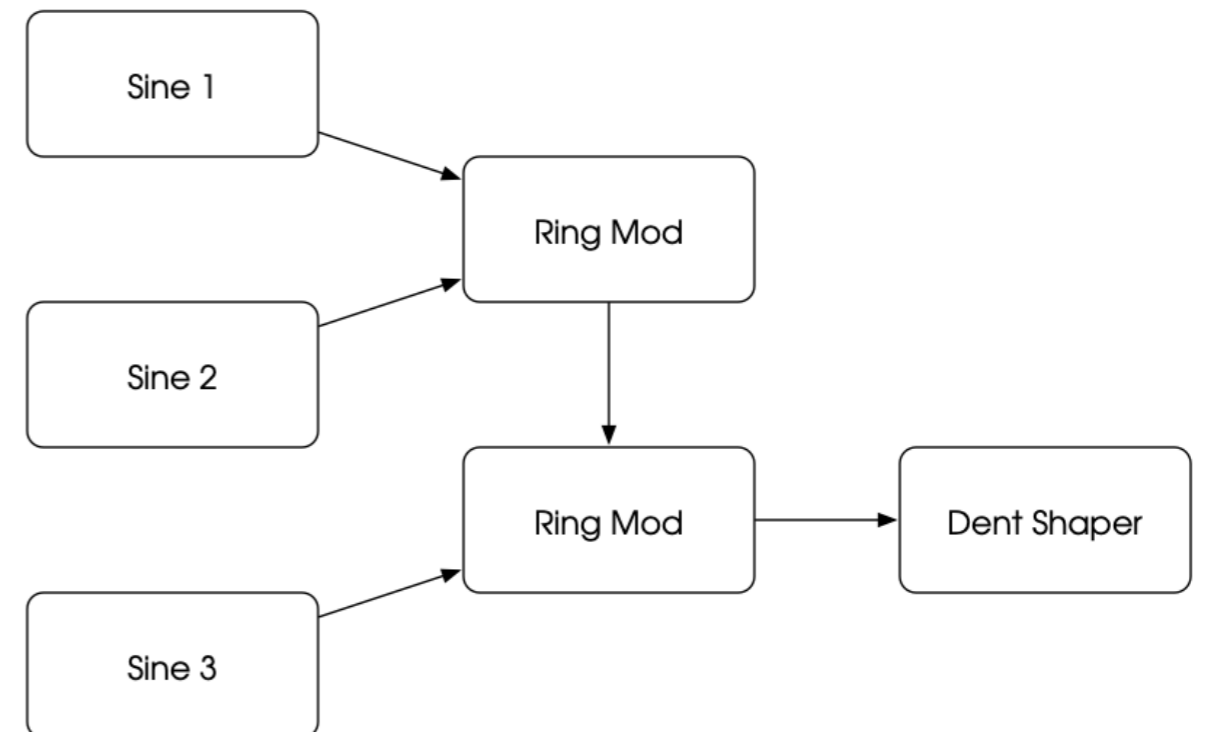
WAVEFOLD

In this mode, two sine wave oscillators are sent through a wavefolder, a type of foldover distortion used frequently in modern modular synthesizers.

STEREO MODE

The BLEND parameter is inverted on one channel.

CONTROLS

|	
SINE 1	Sets the frequency of the first sine oscillator.
SINE 2	Sets the frequency of the second sine oscillator as an interval from the first.
FOLD	Sets the intensity of wavefolding applied to the oscillator mix.
BIAS	Add DC bias to the wavefolder, changing the shape of the waveforms produced.
BLEND	Set the blend of the two oscillators.
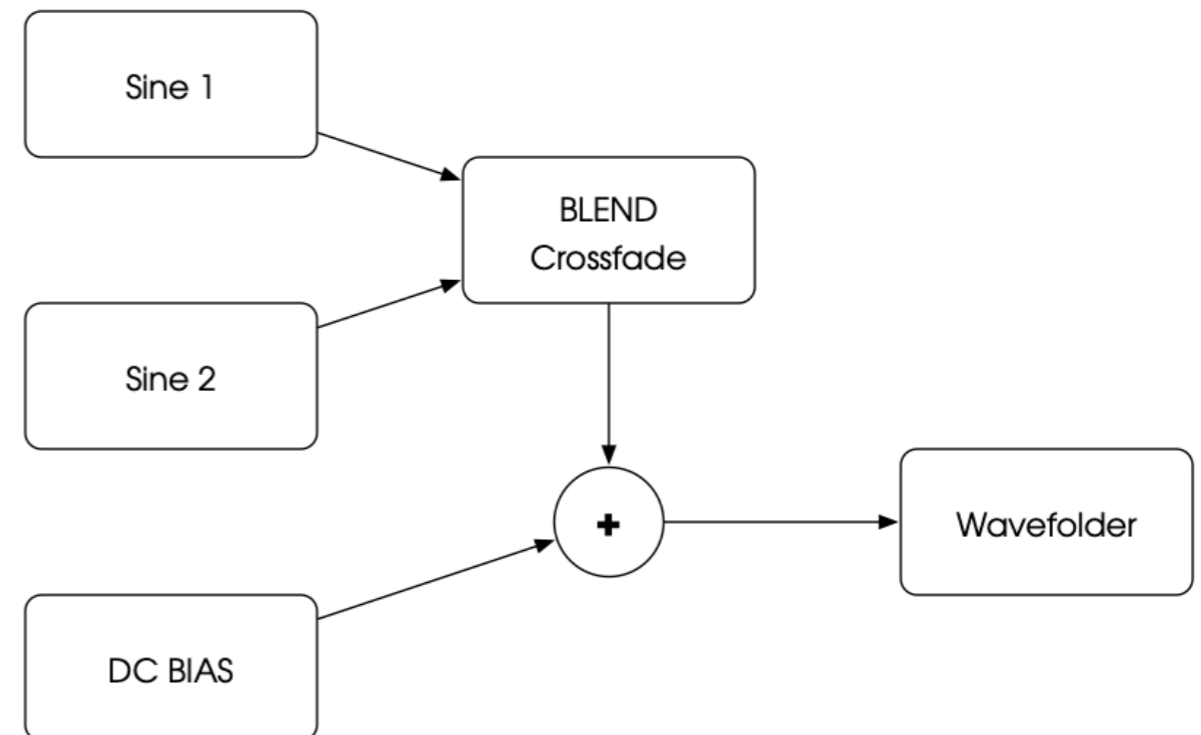
HARMONIC

This harmonic oscillator is actually a set of eight sine waves used to create richer timbres. The sine waves modified using frequency and amplitude ratios from a primary sine wave set to the base frequency. To add more depth, the harmonic oscillator can be blended with an anti-aliased saw and sent through a Dent waveshaper.

STEREO MODE

In stereo mode, the right channel's saw stays at the base frequency instead of being affected by P. RATIO. Additionally, A. RATIO only affects the left channel.

CONTROLS

| TUNE	Changes the primary frequency of the oscillator.
P. RATIO	Sets the frequency offset between each sine wave in the series. Also affects the frequency of the secondary saw.
A. RATIO	Sets the amplitude falloff between each sine wave in the series.
BLEND	Crossfade between an anti-aliased saw waveform and the harmonic oscillator.
SHAPE	Sets the amount of Dent waveshaping that occurs after the BLEND stage.

FM
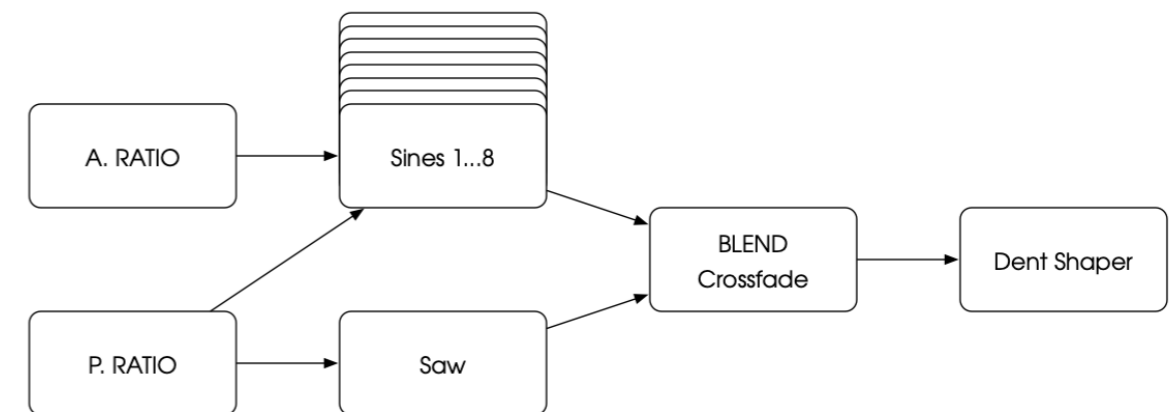
2-OP FEEDBACK

Straightforward 2-operator FM synthesis. The two operators additionally have feedback controls for self phase-modulation.

STEREO MODE

The modulator operator's output is inverted on the opposing channel, leading to a different FM direction.

CONTROLS

|	
CARRIER	Sets the frequency of the carrier oscillator.
MOD	Sets the frequency of the modulator oscillator.
INDEX	Sets the depth of frequency modulation.
MOD FB	Sets the depth of self phase-modulation on the modulator oscillator.
CAR FB	Sets the depth of self phase-modulation on the carrier oscillator.
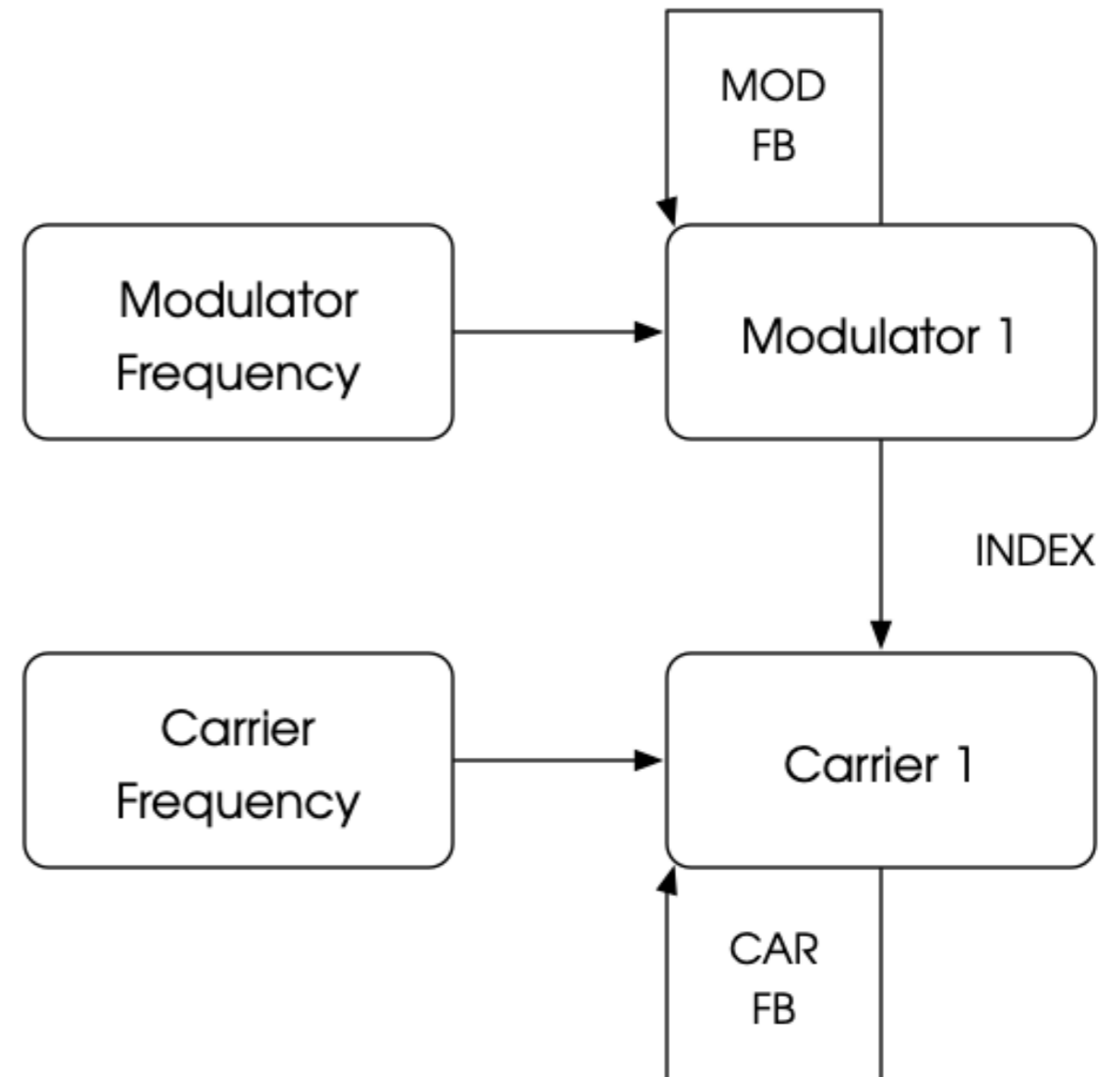
3-OP SPLIT

Three operator FM synthesis. The two modulating oscillators share one frequency control that affects them differently.

STEREO MODE

The modulator operator's output is inverted on the opposing channel, leading to a different FM direction.

CONTROLS

|	
CARRIER	Sets the frequency of the carrier oscillator (oscillator 1).
MOD	Sets the frequency of the two modulator oscillators (oscs 2 and 3). The two modulators are set in opposite directions as offsets from the carrier.
2 > 1	Sets the depth of frequency modulation from operator 2 to the carrier.
3 > 1	Sets the depth of frequency modulation from operator 3 to the carrier.
3 > 2	Sets the depth of frequency modulation from operator 3 to operator 2.
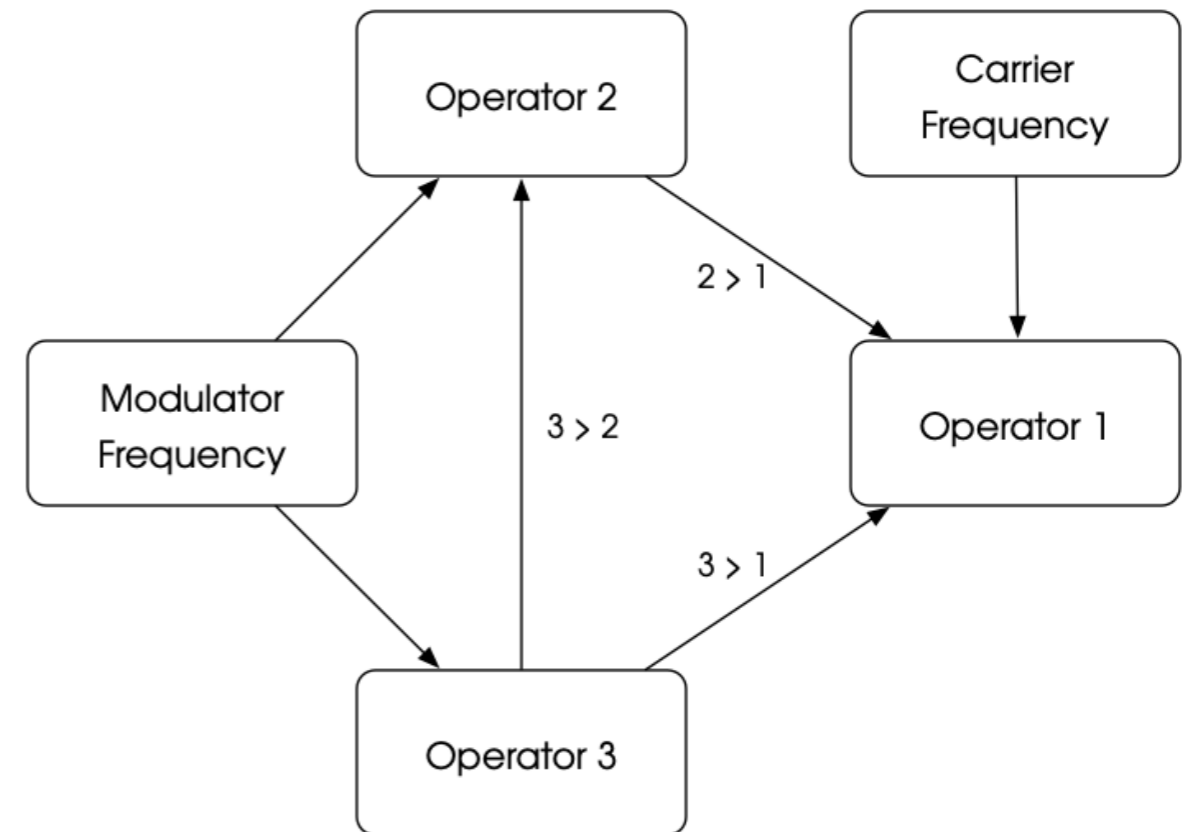
4-OP PARALLEL

DESCRIPTION

Four operator FM synthesis. In this mode, there are two pairs or carriers and modulators. The modulators share a common frequency control that affects them separately. The FM pairs can be crossfaded between. Additionally, there is an X FM control that introduces another stage of FM.

STEREO MODE

The modulator operator's output is inverted on the opposing channel, leading to a different FM direction.

CONTROLS

| CARRIER	Sets the frequency of the carrier oscillators.
MOD	Sets the frequency of the two modulator oscillators. The two modulators are set in opposite directions as offsets from the carrier.
INDEX	Sets the depth of frequency modulation on each carrier-modulator pair.
BLEND	Crossfade between the two carrier-modulator pairs.
X FM	Sets the amount of frequency modulation that one carrier-modulator pair introduces upon the opposing carrier.
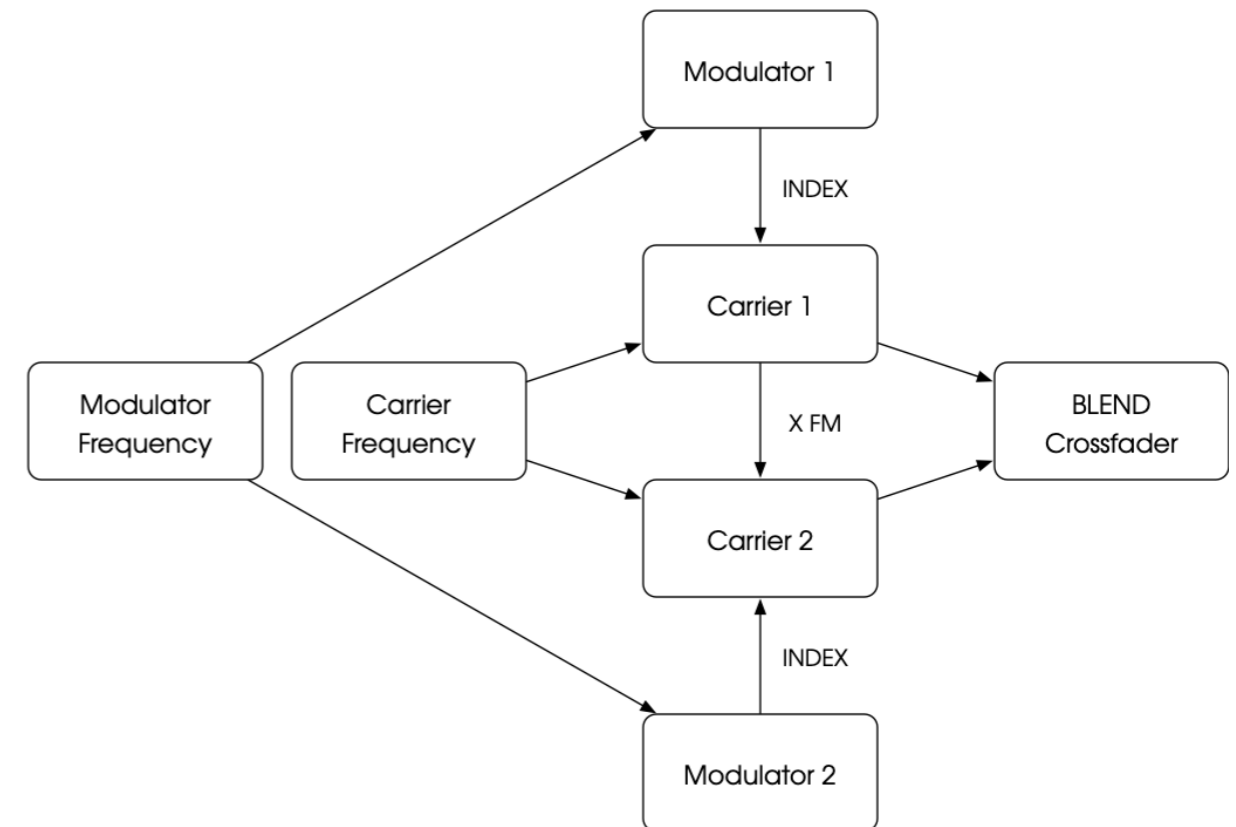
STACKS

SUPERSAW/SINE/SQUARE

The superoscillator modes create dense, moving timbres by creating a stack of oscillators. The main and SUB oscillators make up the primary sound, while the rest of the oscillators are called the swarm. **LION's unison features are disabled for the superoscillator modes**.

STEREO MODE

The Swarm oscillators are distributed evenly across the stereo field.

CONTROLS

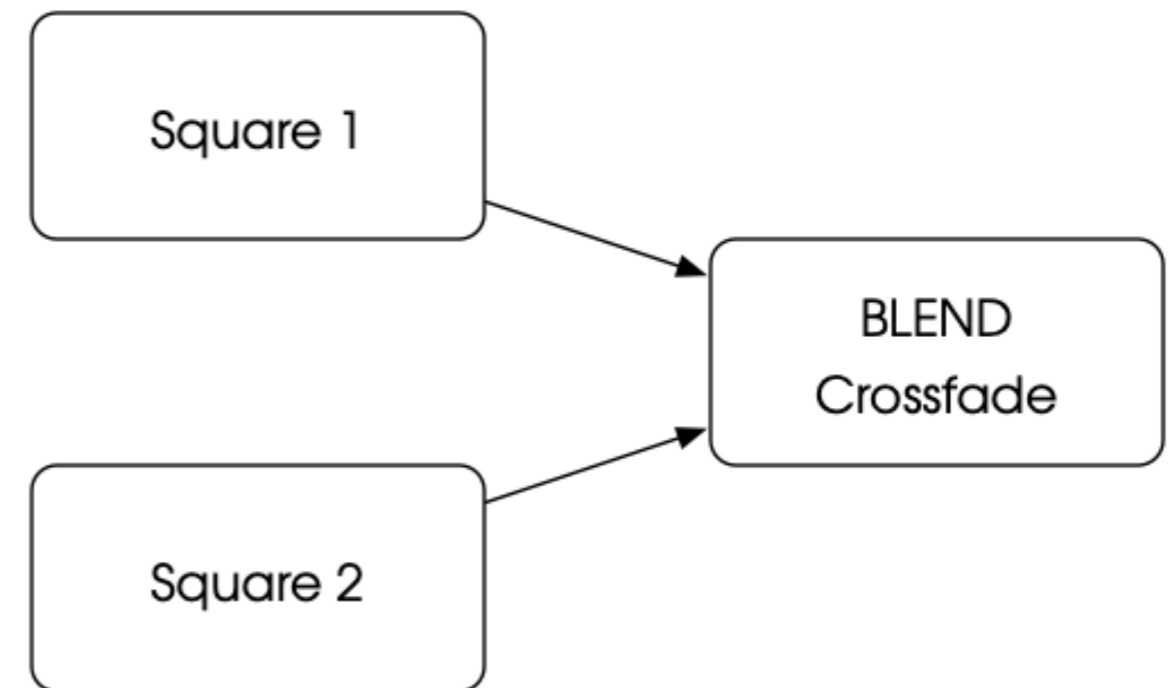|	
TUNE	Changes the primary frequency of the oscillator.
SUB	Changes the frequency of the sub oscillator as an offset from the Main frequency.
OSCS	Set the number of active oscillators in the swarm.
DEPTH	Set the frequency variation between the swarm oscillators.
BLEND	Crossfade between the primary + SUB oscillators and the swarm oscillators.
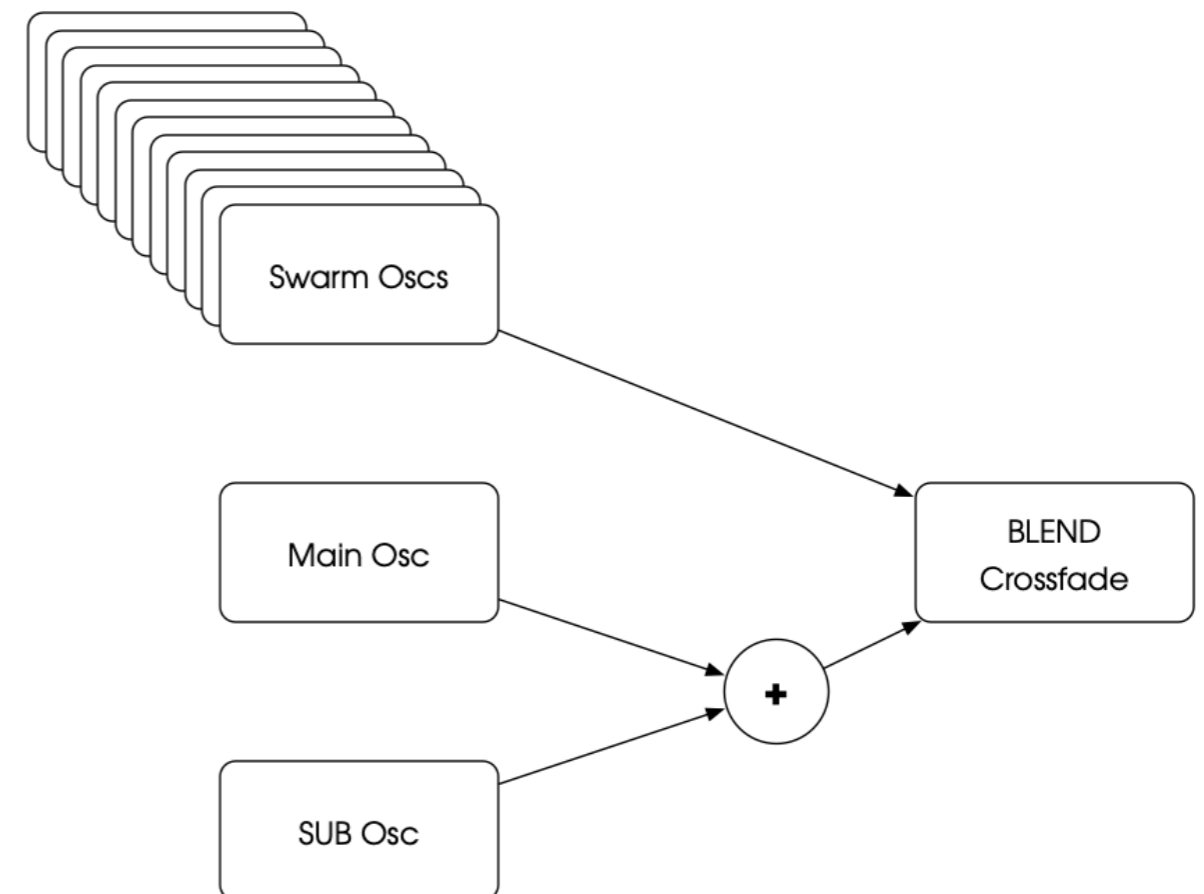
SMOKESTACK

In this mode, two anti-aliased square oscillators are stacked together. Each oscillator has independent tuning and pulse width.

STEREO MODE

In stereo mode, two more oscillators are introduced, but the pulse width controls are swapped on the second pair.

CONTROLS

|	
SQR 1	Changes the frequency of the first square wave.
SQR 2	Changes the frequency of the second square wave as an offset from the first.
PW 1	Changes the pulse width of the first square wave.
BLEND	Crossfade between the two square waves.
PW 2	Changes the pulse width of the second square wave.

SUPERLIMINAL

This is a hybrid between the Subliminal oscillator mode and the various superoscillator modes. **LION's unison features are disabled for this mode.**

STEREO MODE

The Swarm oscillators are evenly distributed across the stereo field.

CONTROLS

|	
TUNE	Changes the primary frequency of the oscillator.
SUB	Changes the frequency of the sub oscillator as an offset from the Main frequency.
DEPTH	Set the frequency variation between the swarm oscillators.
SHAPE	Set the amount of Dent waveshaping that occurs on each oscillator.
SPLIT	Set the amount of Dent splitting that occurs on each oscillator.
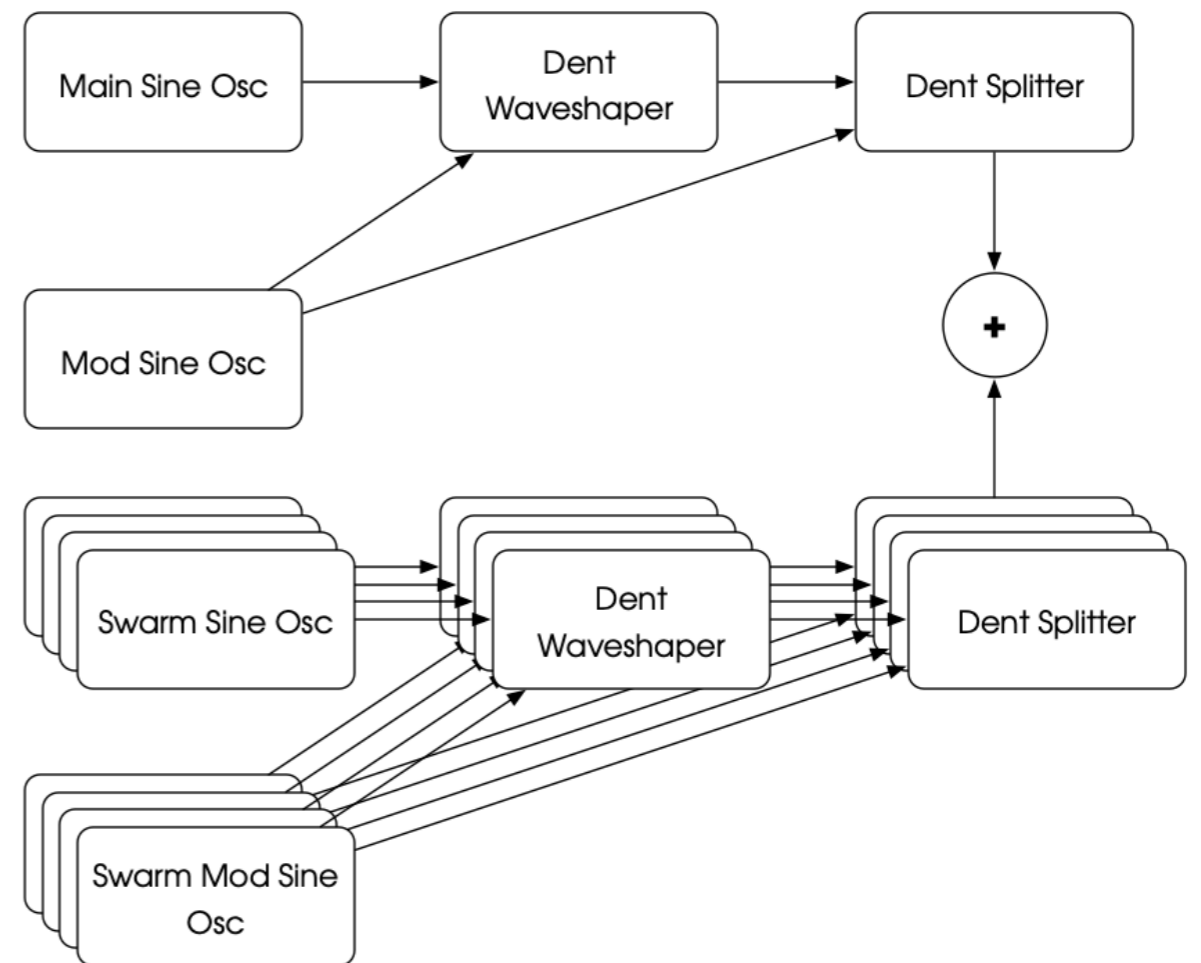
GRANULAR

VOSIM

This mode implements VOSIM synthesis, a type of granular synthesis used for creating vocal tones. Two high-frequency sine waves are enveloped by a lower frequency saw wave, to which they are additionally phase-locked.

STEREO MODE

The stereo features are only audible with Partial FM. When enabled, Partial 2's FM depth to Partial 1 is inverted on one channel, leading to FM occurring in a different direction.

CONTROLS

| TUNE	Sets the frequency of the sawtooth envelope, which is the most dominantly heard frequency.
PARTIAL 1	Sets the frequency of the first internal sine oscillator.
PARTIAL 2	Sets the frequency of the second internal sine oscillator.
BLEND	Crossfade between the two partials.
PARTIAL FM	Sets the depth of frequency modulation between the two partials.
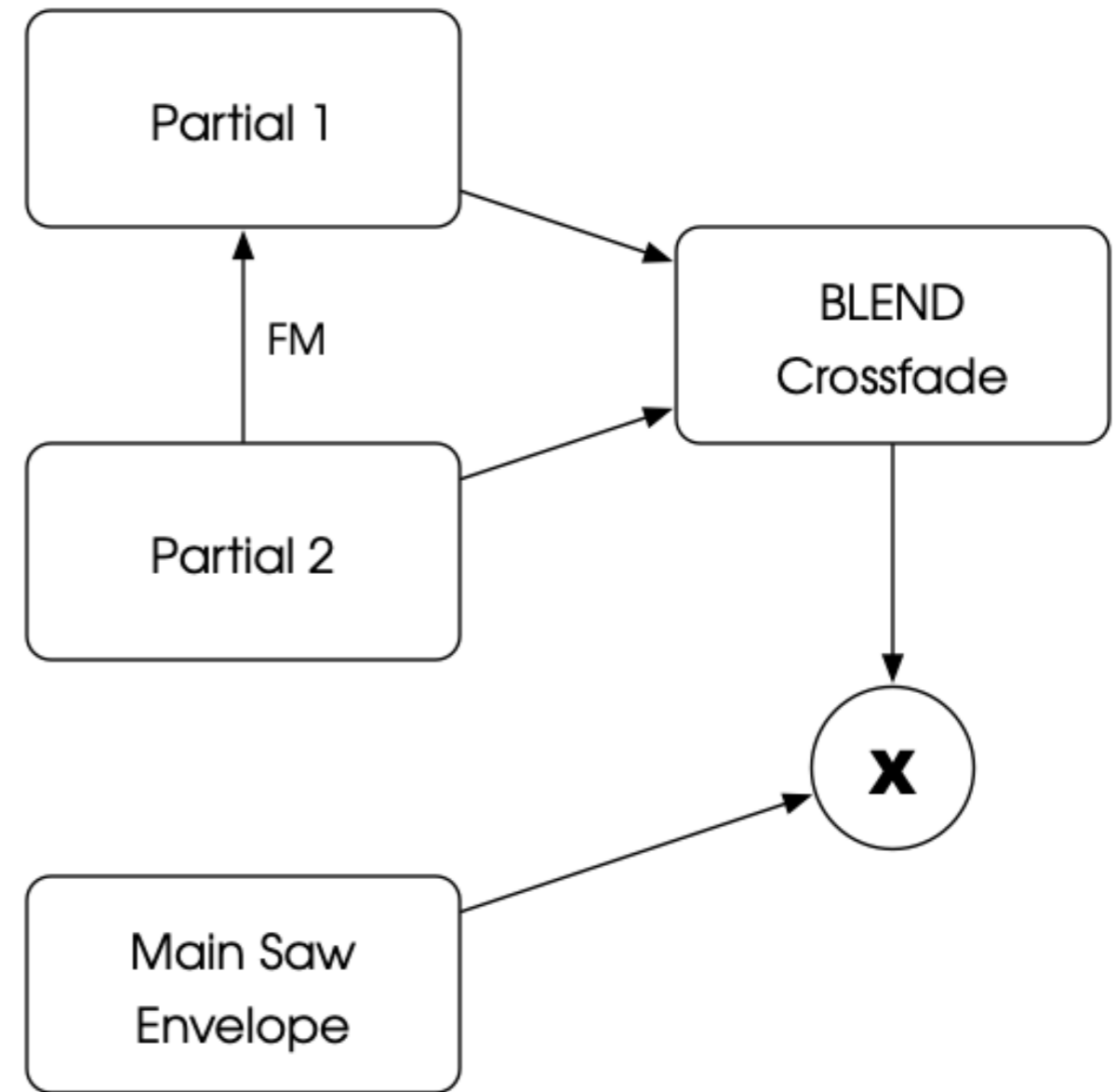
PULSAR

This is a classic mode of granular synthesis described by Curtis Roads. A sine wave oscillator is enveloped by a series of windows. This windows can be applied uniformly, randomly, or through the use of other algorithms. In this implementation, the uniform window stream can be masked randomly or through micro Euclidean rhythms.

STEREO MODE

In stereo mode, the Euclidean mask will be inverted between the two channels and the random chance will be independent.

CONTROLS

TUNE	Sets the frequency of the internal sine oscillator for each grain.
WINDOW	Sets the rate of generation for new grain windows.
FORMANT	Sets the shape of the grain window.
PROB	Sets the likelihood that each window will be silent each cycle.
MASK	Generates a Euclidean masking pattern for the window.

Main Sine Osc → X

Gaussian Envelope → Formant Shaping → Envelope Masking → X
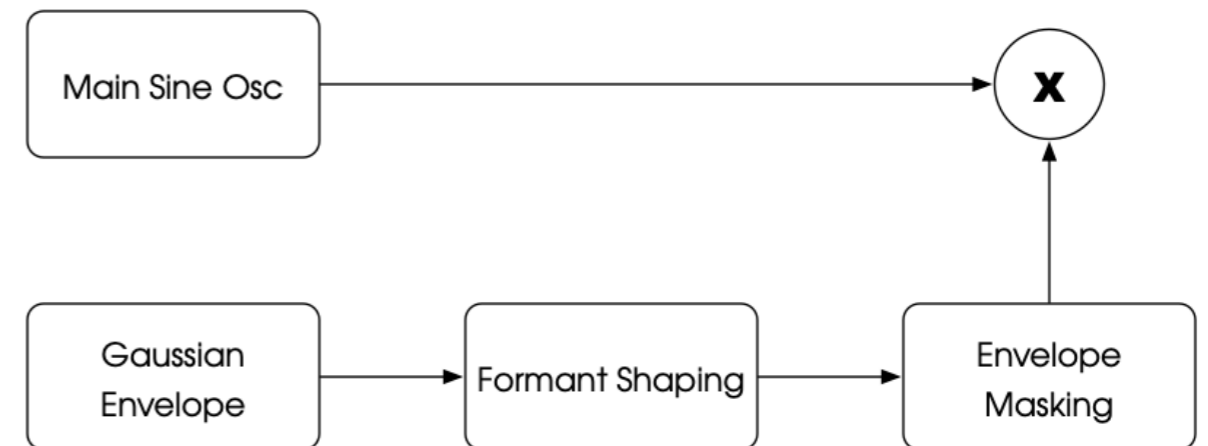
EXPERIMENTAL

SUMSYN

DESCRIPTION

This is an implementation of Summation Synthesis, originally developed by James Moorer and introduced to us by Stephen McCaul of Noise Engineering. This implementation features a lot of our own modifications. In this mode, three sine waves are used to create extremely complex timbres through a deep mix of phase modulation and wave shaping.

STEREO MODE

The BRIGHT algorithm has a number of inverted interactions between the oscillators on both channels.

CONTROLS

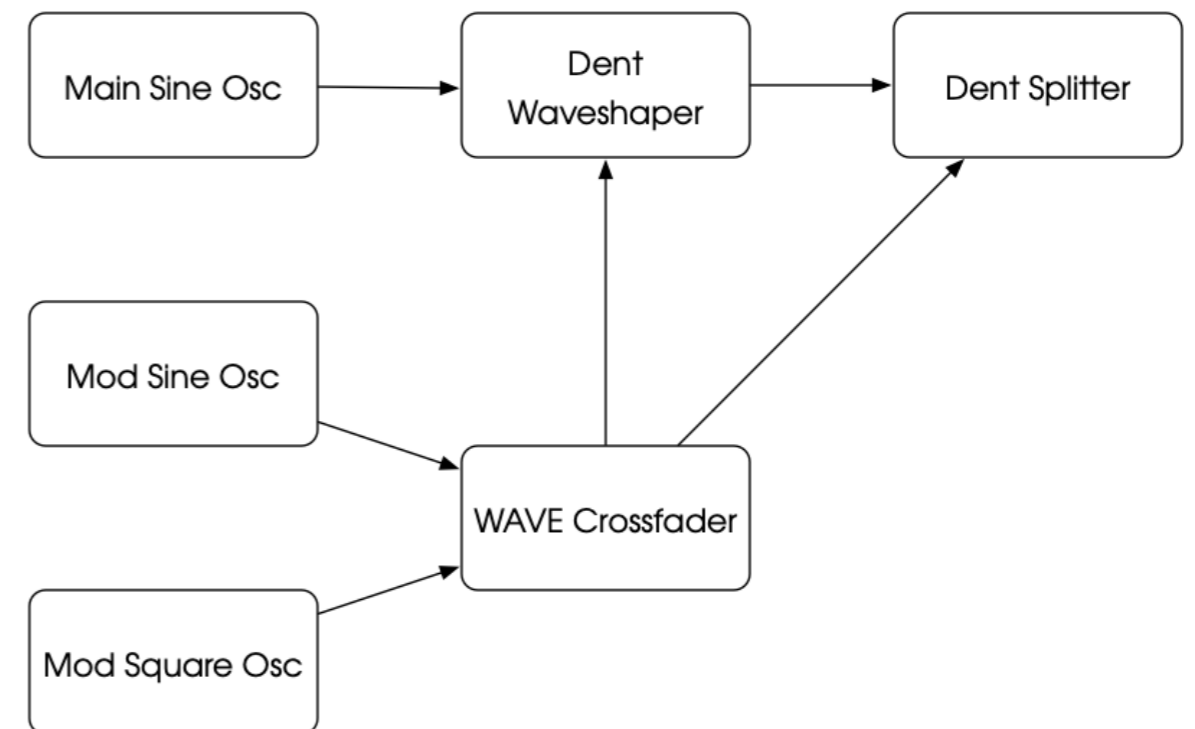| TUNE	Changes the primary frequency of the Sine and Saw oscillators.
SUB	Changes the frequency of Sine 2 as an offset from the Main frequency.
BRIGHT	This is a complex parameter that changes the timbre of the output through various internal interactions.
PM	Sets the depth of phase-modulation for the primary sine waveform from the second sine oscillator.
BLEND	Crossfade between a phase-modulated sine and a stable saw waveform for the primary oscillator.
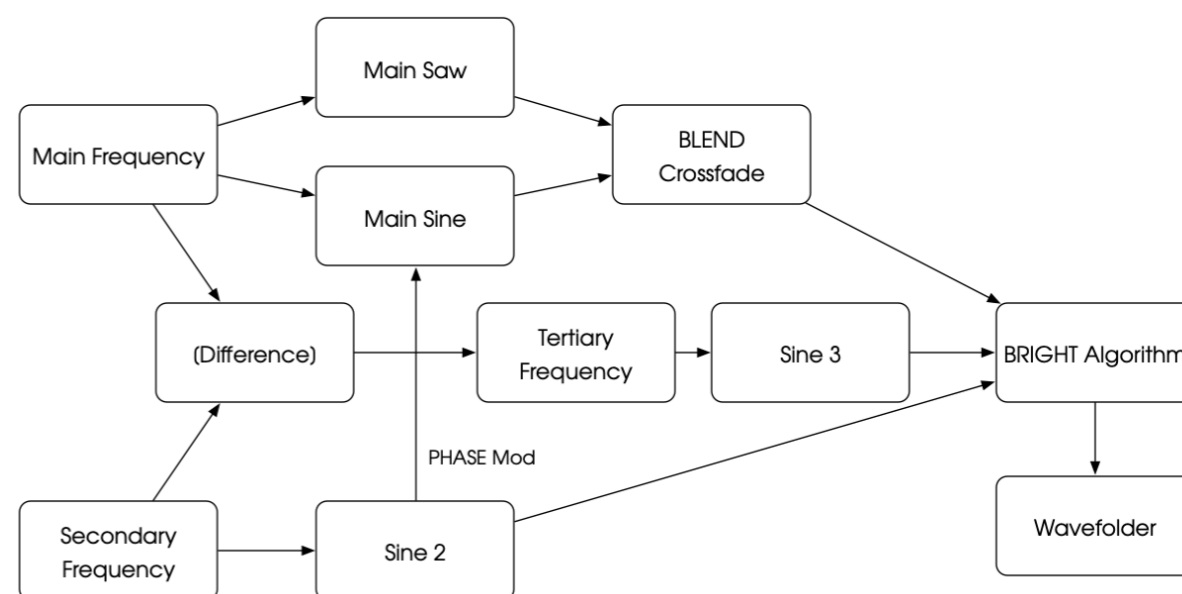
SUBLIMINAL

DESCRIPTION

This mode is a mashup of AM synthesis and our Dent distortion. The primary oscillator is a sine wave that runs through the Dent algorithm. The MOD oscillator is a secondary oscillator that changes the Dent parameters at audio rate.

STEREO MODE

The opposing channel's instance of Dent has a different signal path to create a slight variation on the sound.

CONTROLS

TUNE	Changes the primary frequency of the oscillator.
MOD	Change the frequency of the modulator oscillator as an offset of the primary oscillator.
SHAPE	Change how much the modulating oscillator affects the waveshape of the primary oscillator.
SPLIT	Change how much the modulating oscillator affects the Dent Split of the primary oscillator.
WAVE	Morphs the modulating oscillator between a sine and an anti-aliased square.

Main Sine Osc → Dent Waveshaper → Dent Splitter
Mod Sine Osc → WAVE Crossfader → Dent Waveshaper / Dent Splitter
Mod Square Osc → WAVE Crossfader

CIRCUIT BENT

DESCRIPTION

This is an extremely tangled mode featuring three sine waves that interact in various ways, including sample reduction and bitwise mixing.

STEREO MODE

When enabled, the bitwise mixing operation swaps the order that inputs are received, leading to a slightly different timbre on each channel.

CONTROLS

|	
TUNE	Changes the primary frequency of the oscillator.
SUB	Changes the frequency of the two sine sub oscillators as an offset from the Main frequency.
BLEND	Blends between the main oscillator and a bitwise mix of main and SUB. Additionally, this adds a light amount of sample rate manipulation.
CRUSH	Changes the amount of bit destruction on the main oscillator.
SHAPE	Changes the amount of waveshaping on the oscillators.
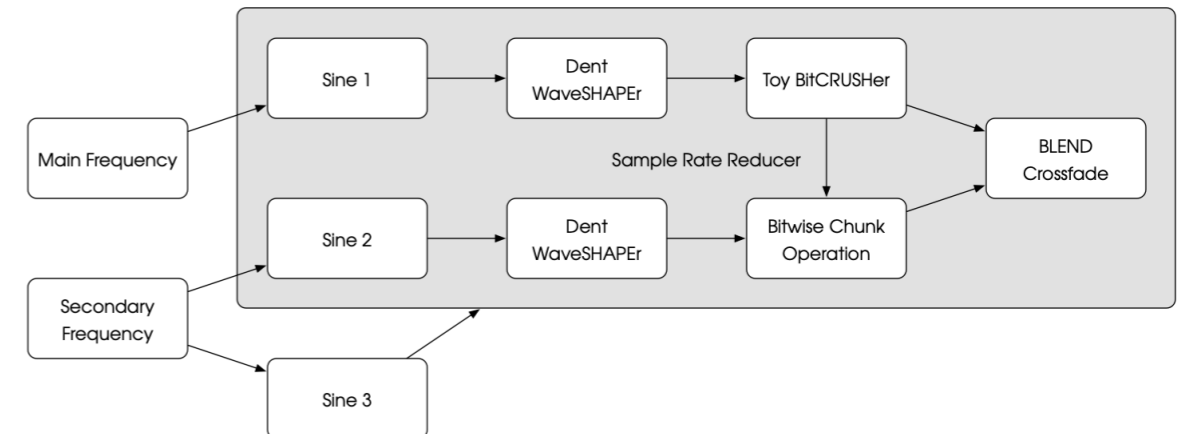
XOR SQUARES

In this mode, two aliasing square waves are XORed together
and run through a samplecrusher.

STEREO MODE

the oscillators are doubled but the pulse width controls are
reversed between the sets.

CONTROLS

|	
TUNE	Changes the primary frequency of the oscillator.
SUB	Changes the frequency of the sub oscillator as an offset from the Main frequency.
S. RATE	Sample crush the oscillators. Does not affect the frequency of the oscillators.
PW	Changes the pulse width of the main oscillator.
PW SUB	Changes the pulse width of the SUB oscillator.
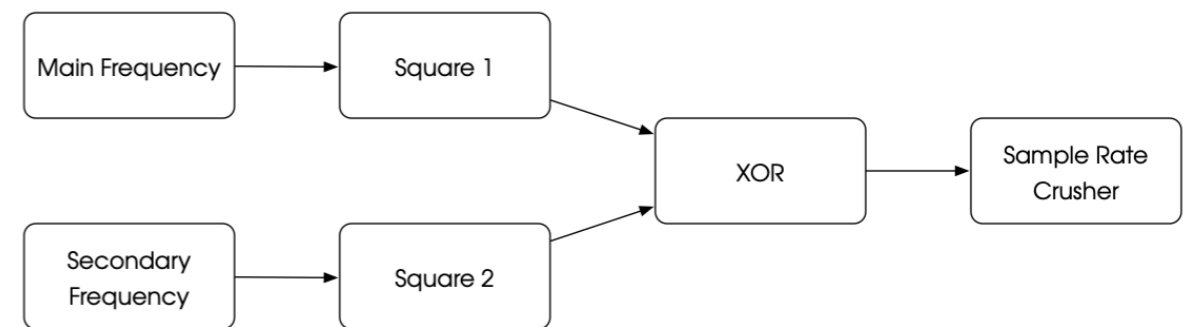
CHAOS

In this mode, two sine oscillators are chaotically intertwined.
Each time an oscillator completes a cycle, it changes frequency
based on the depth of the CHAOS controls and the current
amplitude of the other oscillator.

STEREO MODE

When stereo mode is activated, four oscillators are used
instead and two are placed on each channel.

CONTROLS

TUNE	Changes the primary frequency of the oscillator.
SUB	Changes the frequency of the sub oscillator as an offset from the Main frequency.
CHAOS 1	Determines how much the SUB oscillator can affect the pitch of the main oscillator.
BLEND	Crossfade between the main oscillator and the SUB oscillator.
CHAOS 2	Determines how much the main oscillator can affect the pitch of the SUB oscillator.
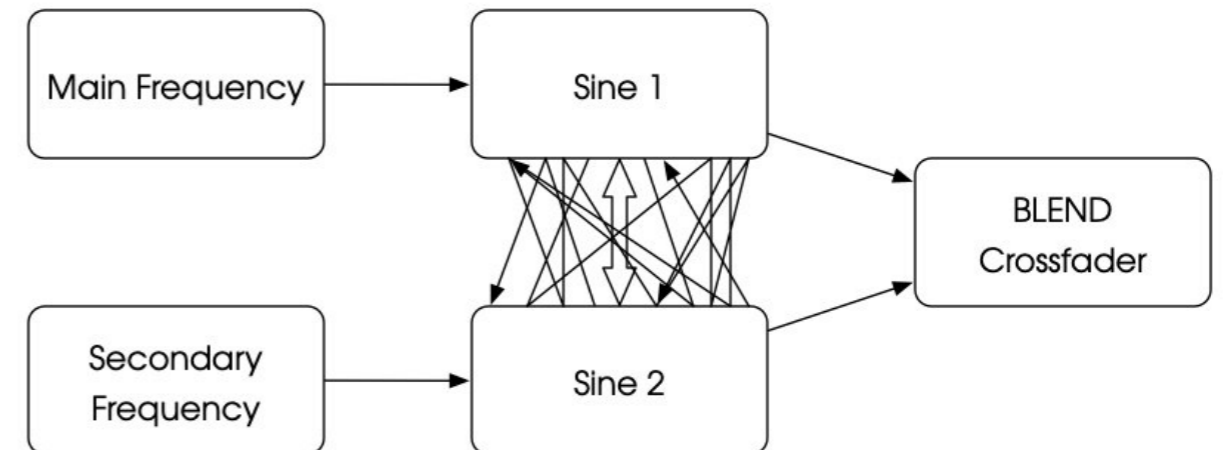
SIMPLE

SIMPLE SINE/TRI/SAW/SQUARE

Each of the Simple modes provides two oscillators of the given waveform. These oscillators have the standard tuning controls but no other parameters. These modes are very low-cost and are useful when used with LION's Unison features and Mixer.

STEREO MODE

No stereo features are available for these modes

CONTROLS

TUNE	Changes the primary frequency of the oscillator.
SUB	Changes the frequency of the sub oscillator as an offset from the Main frequency.
CONTROL 3	Not used
CONTROL 4	Not used
CONTROL 5	Not used

NOISE

NOISE

This mode provides a traditional noise source, where a white noise generator is passed to a filter.

STEREO MODE

A different noise generator is used on each channel.

CONTROLS

TUNE	Not used.
SUB	Not used.
DIGITAL	Blends between white noise and blocky noise.
S. RATE	Changes the sampling rate on the noise generators.
CUTOFF	Applies low- or high-pass filtering to the noise. At 50%, no filtering is applied.

DUST

DESCRIPTION

Dust is a classic noise uGen from SuperCollider (a uGen is a "Unit Generator," or SuperCollider's rough equivalent of a module). It creates variable-height impulses by comparing a noise source against a target threshold. At low Density settings, these are infrequent pops that can be useful for triggers, especially for asynchronous granular synthesis. At high density settings, this turns into crunchy digital noise, and at 100% density, this becomes a white noise signal. This implementation of Dust also includes two parallel band-pass filters for turning

STEREO MODE

A different noise source is used on each channel.

CONTROLS

FILTER 1	Sets the frequency of the first band-pass filter.
FILTER 2	Sets the frequency of the second band-pass filter as an offset from the first.
DENSITY	Sets how frequently new impulses are generated.
BLEND	Crossfade between the raw Dust impulses and the filtered version.
RAND FREQ	Sets the depth of random frequency modulation applied to the filters with every new impulse.

CRACKLE

Crackle is a classic chaotic uGen from SuperCollider. It uses a deterministic, seeded equation to create hisses, pops, and other pleasant, vinyl-like noises. In addition to the original algorithm, this includes the Broken mode introduced in the Euro Reakt collection for Reaktor 6. This Broken mode creates intense trapped-buffer stutter effects and self-oscillations at high Chaos settings, leading to digital squeals and modem noises. This implementation also includes two parallel band-pass filters for shaping the final signal.

STEREO MODE

The Crackle generators are seeded differently on each channel, leading to different outcomes.

CONTROLS

| FILTER 1	Sets the frequency of the first band-pass filter.
FILTER 2	Sets the frequency of the second band-pass filter as an offset from the first.
DENSITY	Influences the timbre of the Crackle modes. In the regular Crackle mode, raising this will introduce pops and skips. In Broken mode, this will produce self-oscillations and modem noises.
BROKEN	Crossfade between the original Crackle algorithm and the Broken mode.
FILTERED	Crossfade between the raw Crackle output and the filtered variation.

THE MIXER
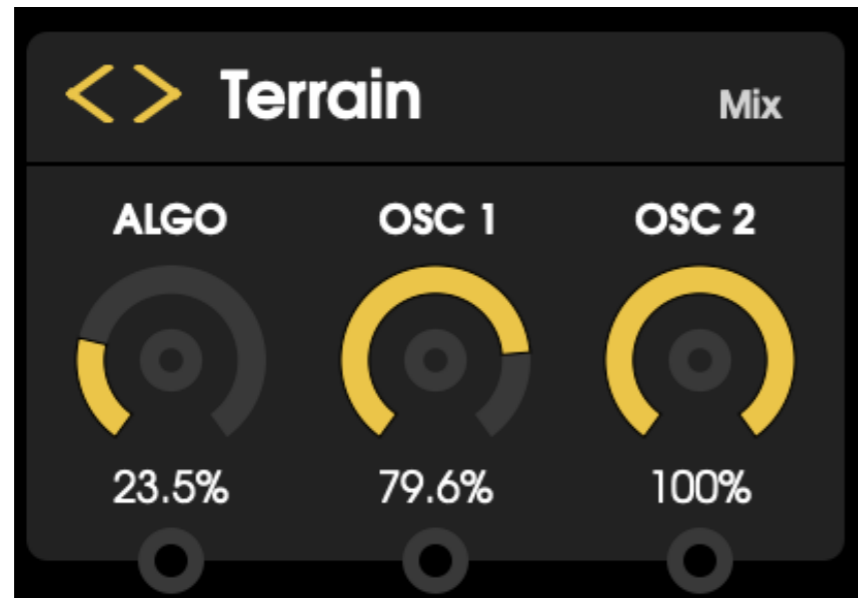
LION's mixer is not just an ordinary summer. Instead, it is designed as an extension of LION's synthesis algorithms and rich modulation section. You can create some surprisingly deep sounds using LION's mixer combined with even the "Simple" oscillator modes.

There are eight mixer modes, each with three basic controls. Two of the controls, OSC 1 and OSC 2, are simply gain controls for each oscillator going into the mixer. Depending on the algorithm, these controls can actually cause dramatic changes.

The leftmost knob changes from mode to mode. In general, it acts as a crossfader. Depending on the mode, it crossfades either between the two inputs or between various output permutations.

LINEAR XFADE	The mixer crossfades between Oscillators 1 and 2 using a linear algorithm.
EQUAL XFADE	The mixer crossfades between Oscillators 1 and 2. At 50%, both oscillators will be summed at their current amplitude without any gain reduction.
MIN-MAX	The amplitudes of the two oscillators are analyzed and the minimum and maximum values are found between the two. The mixer then crossfades between the two. At 50%, you willhear the sum of both oscillators as their minimum and maximum components are summed together.
RING MOD	The two oscillators are ring modulated together. This mode simulates both digital multiplication ring modulation and a more analog style with diode simulation.
COMPARE	Many comparisons are found between the two oscillators. Additionally, some algorithms add rectification distortion. For instance, one algorithm will sum the positive half of Oscillator 1 with the negative half of Oscillator 2.
TERRAIN	A series of wave terrains are created which are then navigated by the two oscillators. The ALGO knob crossfades between the various terrains, essentially creating a three-dimensional terrain.
BITTER	Many bitwise operations are applied to the oscillators. Examples include XOR, bitwise interleave, and bitshifting.
ASH	Various sample-and-hold algorithms are applied to the oscillators. For instance, Oscillator 2 will sample-and-hold Oscillator 1 at zero-crossings or vice-versa.

FILTER MODES
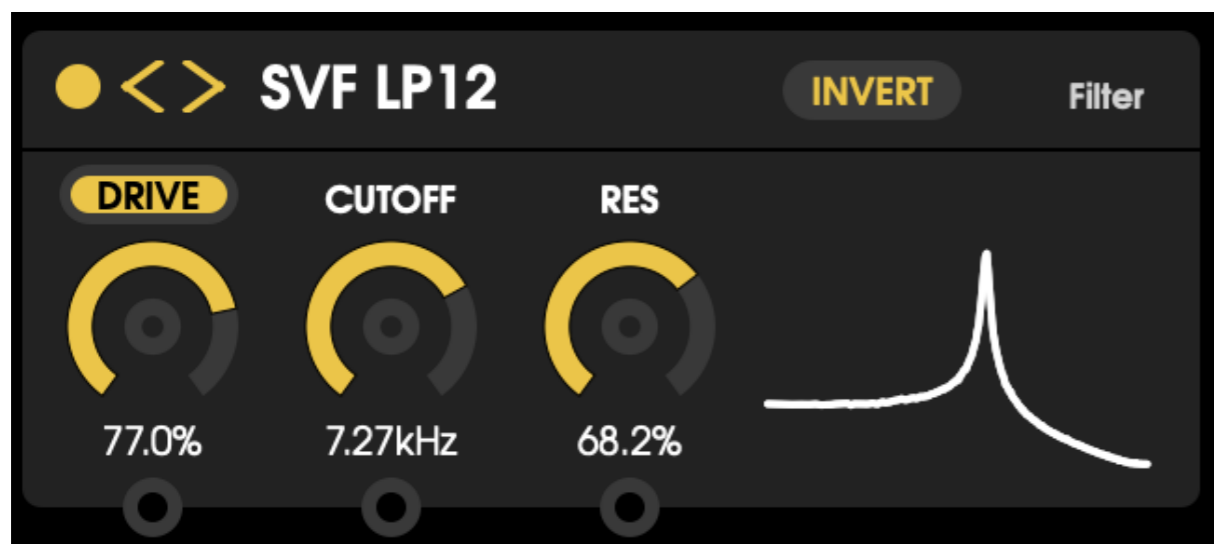
LION's filter section contains many different modes, each with a unique character. Each filter contains two additional modes: DRIVE and INVERT, meaning that there are many permutations to discover.

When enabled, the **DRIVE** parameter turns on non-linear behavior that is unique to each filter mode. The knob below DRIVE acts as a gain controller. At 50%, the signal goes in without any additional gain. Beyond 50%, the signal will start to push against various saturators and clippers inside of the filter.

When **INVERT** is toggled, the filter's topology is inverted by subtracting the filter's wet output from the dry input. These inverted topologies are different from their counterparts (i.e. an inverted Ladder LP will sound different from the regular Ladder HP, despite both being high-pass in nature).

Like LION's mixer, the filter section rewards exploration and modulation.

LP12/HP12/BP12	This clean, neutral filter models are the same models present in our Dent/Indent saturation series. These use very little CPU and do not impart much character of their own.
MS LP/HP	These are very aggressive, resonant filters that model the Sallen-Key topology of a popular semi-modular. The DRIVE circuit is in the feedback path and reacts violently to high resonance settings.
ACID LP	This is a filter modeled off of a classic silver bass box. With default settings, it is quieter than the other filter models. Be sure to add some extra DRIVE and resonance. This reacts well to modulation, especially envelopes.
LADDER LP/HP/BP 12/24	These six modes are modeled off of the ladder topology used in a legendary mini synth. The "24" versions are 24 dB/Oct, meaning that they have a sharper, more dramatic cutoff with more prominent resonance.
SVF LP/HP/BP	These modes are modeled after a famous filter expander. These are similar to the basic LP/HP/BP modes, but impart more of an analog character on the signal.

BYOME EFFECTS

FX

● < > Renoun Reverb ☰

PREDELAY	BANDWIDTH	DAMPING	DECAY	MODRATE	MODDEPTH	DIFFUSE	TAIL	MIX
0 ms	100%	50.0%	80.0%	0.00Hz	0.0%	76.0%	70.7%	50.0%

● < > Basic Filter ☰

CUTOFF	REZ		MIX
OFF	0.0%	LP / HP / BP	100%

+

DELAYS

GLITCH SHIFTER
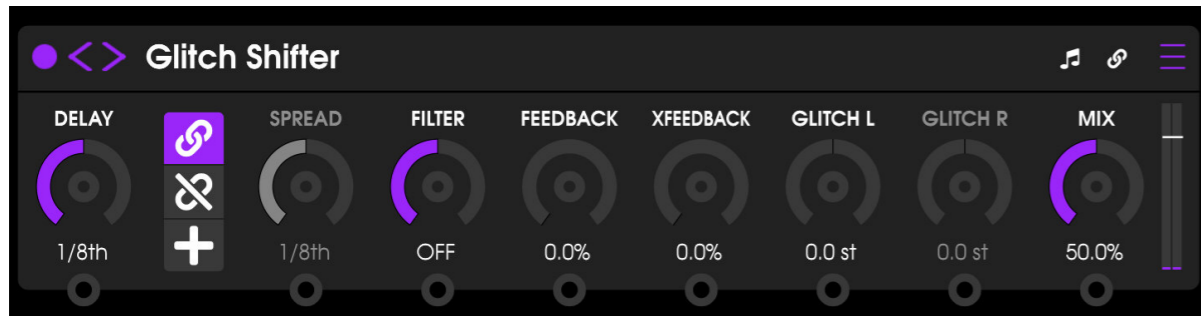
DESCRIPTION

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.

CONTROLS

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
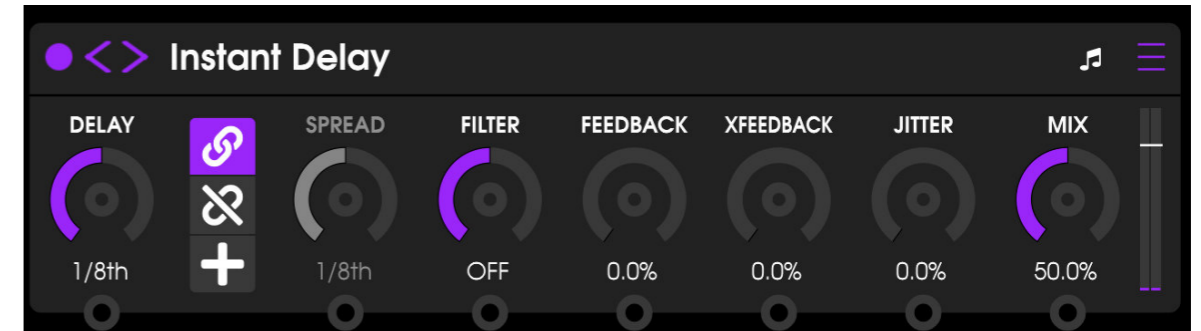
DESCRIPTION

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.

CONTROLS

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
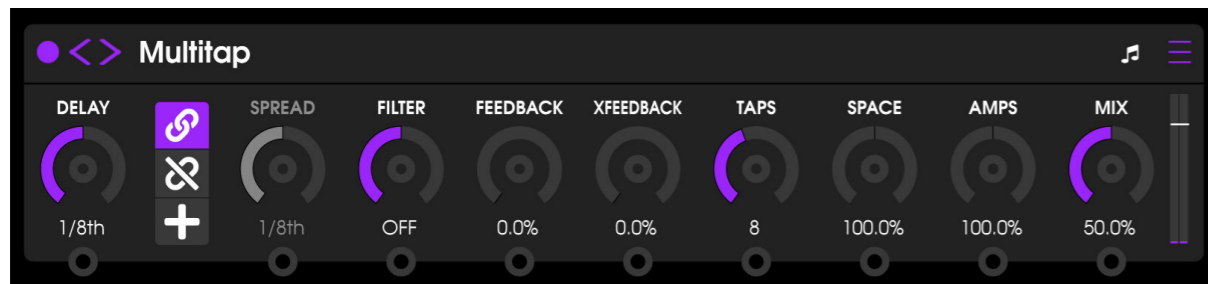
DESCRIPTION

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.

CONTROLS

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
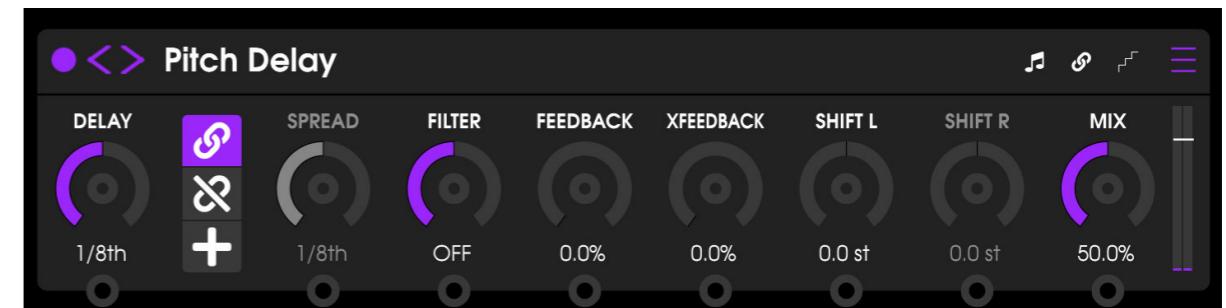
DESCRIPTION

This effect combines a traditional delay with a granular pitch shifter. The pitch shifting occurs before the delay buffer, so all feedback is continuously shifted. 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.

CONTROLS

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.
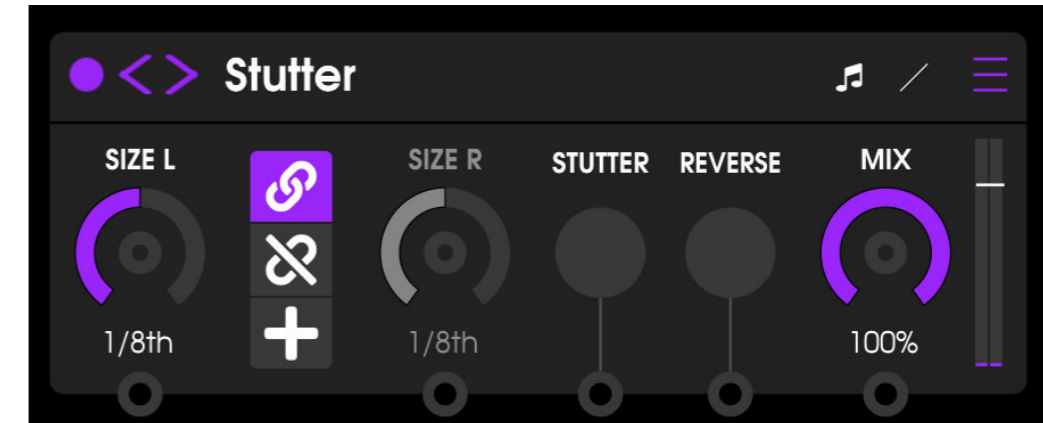
REVERSE DELAY
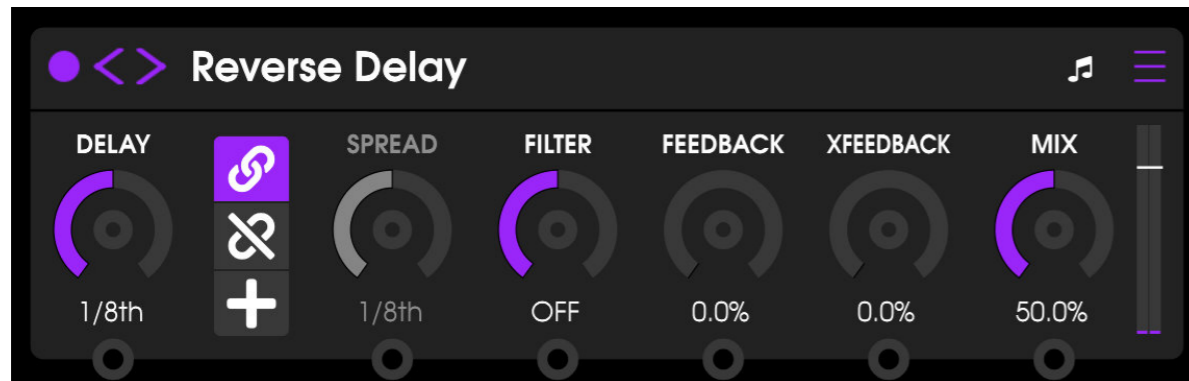
DESCRIPTION

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.

CONTROLS

(No unique controls)

STUTTER

DESCRIPTION

This is a buffer freeze effect that uses elements of Sandman Pro's SLEEP algorithm. When the STUTTER button is activated, the last chunk of audio will be repeated until the button is disengaged. This buffer can be reversed from its current position. Modulating these parameters can create generative glitch-type sounds.

CONTROLS

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
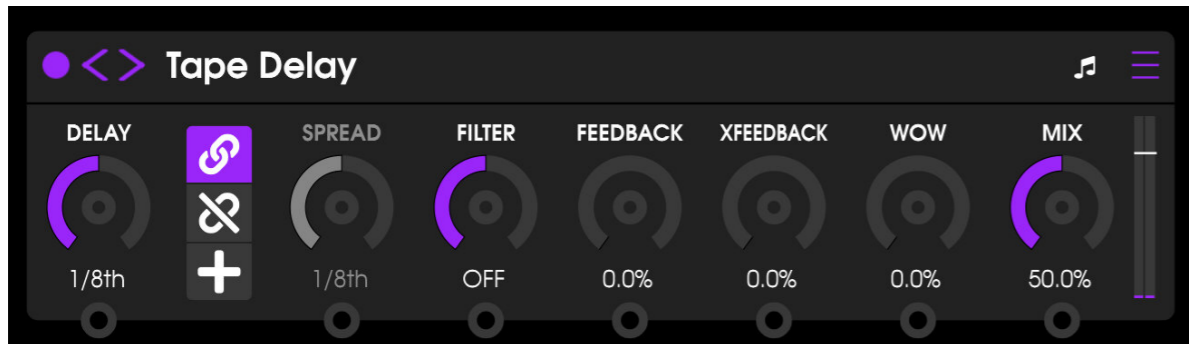
DESCRIPTION

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.

CONTROLS

Wow: Sets the amount of pitch wobble on the tape. This effect is much more noticeable with large FEEDBACK amounts.

DISTORTIONS

BITCRUSH
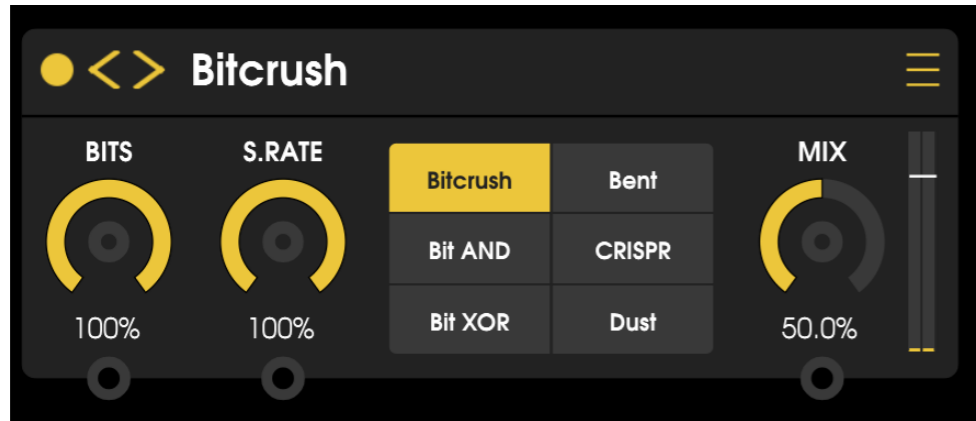
DESCRIPTION

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.

CONTROLS

Bits: Sets the number of bits used to represent the signal.

S. Rate: Sets the sampling rate.

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
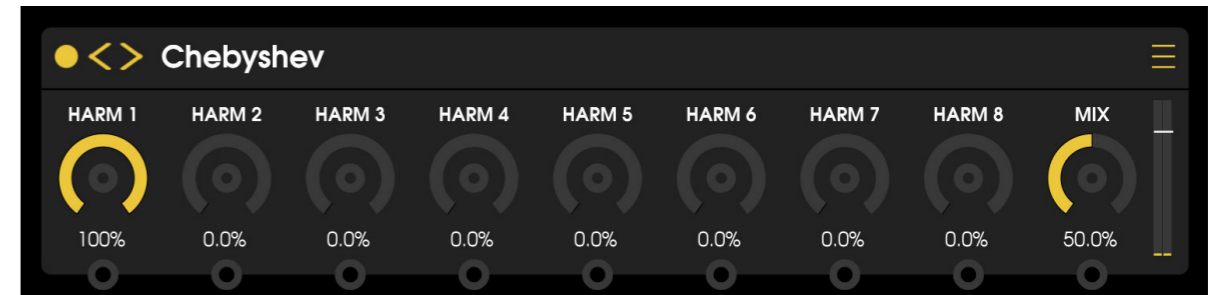
DESCRIPTION

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.

CONTROLS

Harmonic 1-8: Sets the amplitude of harmonic 1-8.

CONTRAST

**AMOUNT** **MIX**

0.0% 50.0%
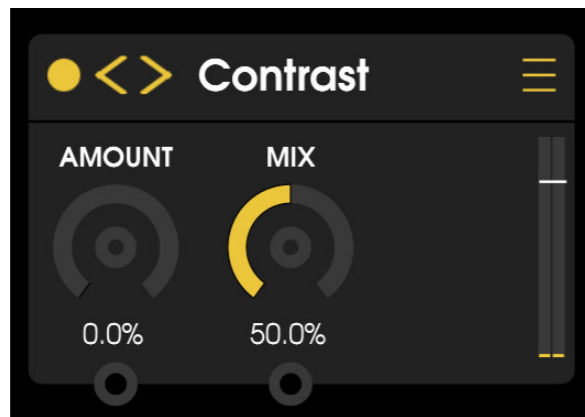
DESCRIPTION

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.

CONTROLS

**Amount**: Sets intensity of Contrast effect.

DENT RECTIFY

UPPER **LOWER** **DC BLOCK** **MIX**

100.0% 0.0% 50.0%
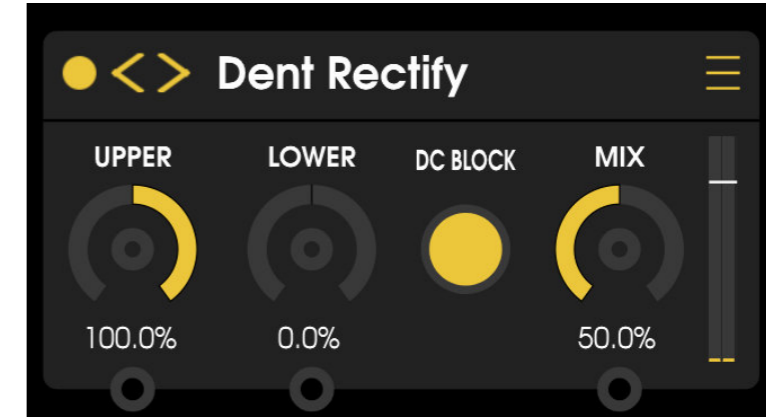
DESCRIPTION

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.

CONTROLS

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
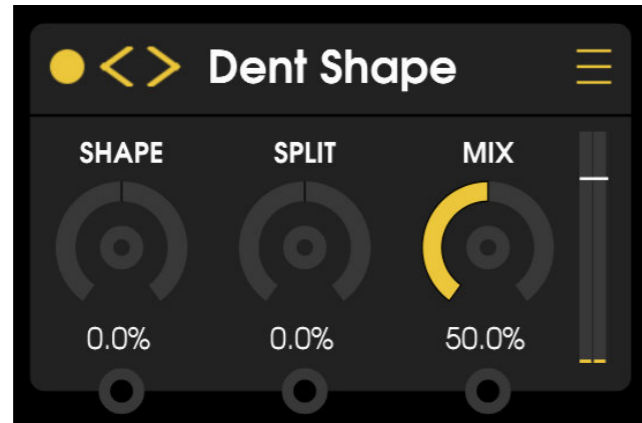
DESCRIPTION

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.

CONTROLS

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
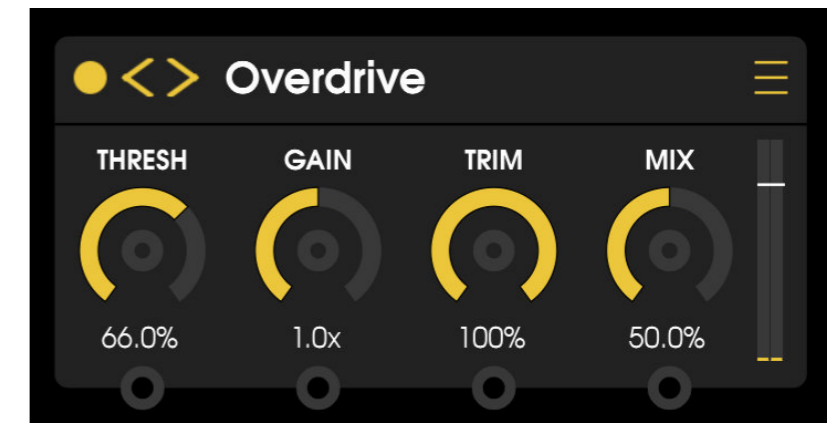
DESCRIPTION

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

CONTROLS

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
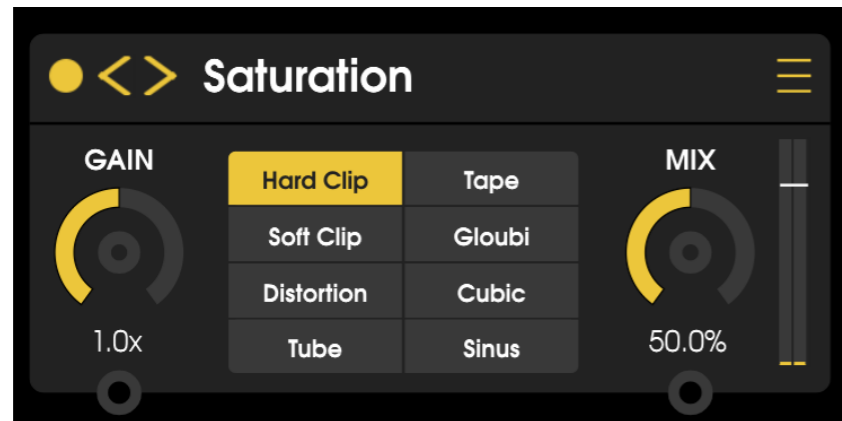
DESCRIPTION

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.

CONTROLS

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

WAVEFOLD
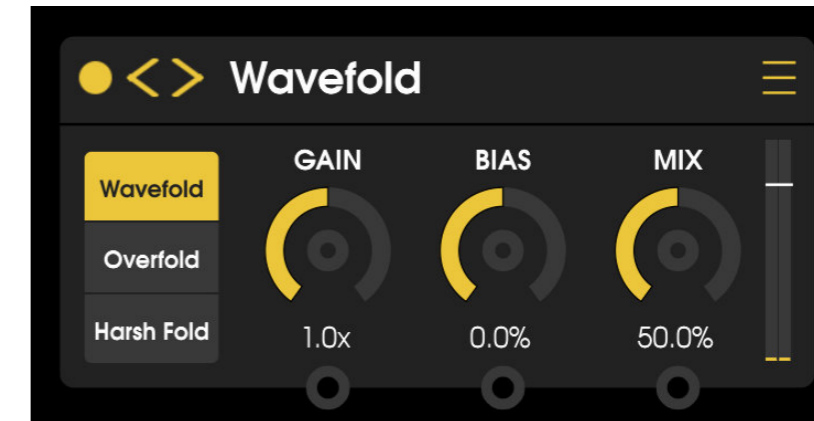
DESCRIPTION

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).

CONTROLS

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.

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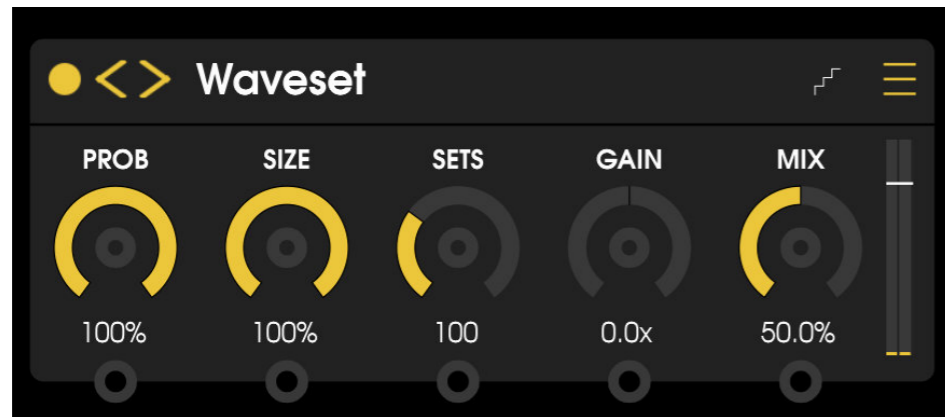
This effect provides a basic implementation of waveset processing, a technique developed by Trevor Wishart. A waveset is a set of three zero crossings, typically one full cycle of a waveform (a single cycle of a sine wave, for instance). In this effect, an amount of gain is applied to selected wavesets.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

CONTROLS

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

DESCRIPTION

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.

CONTROLS

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.

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

COMPRESSOR
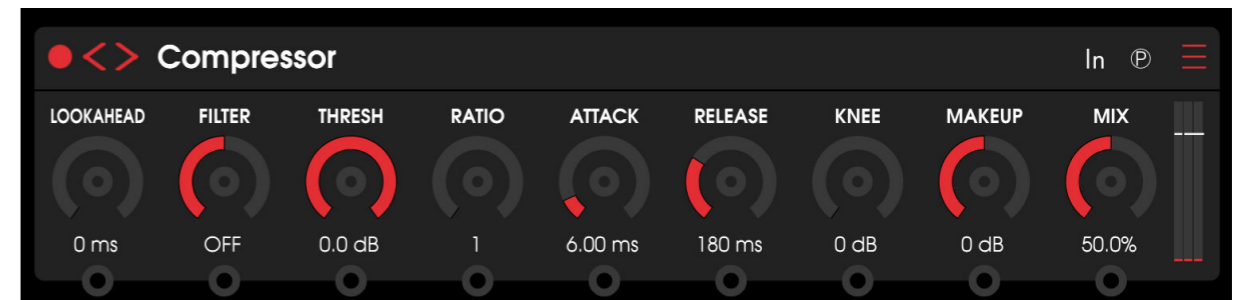
DESCRIPTION

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.

CONTROLS

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
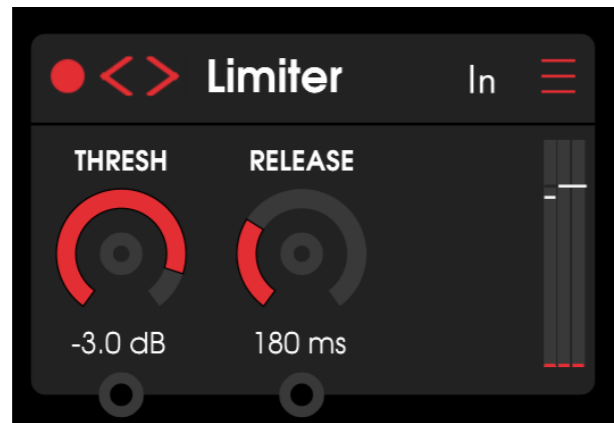
DESCRIPTION

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.

CONTROLS

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
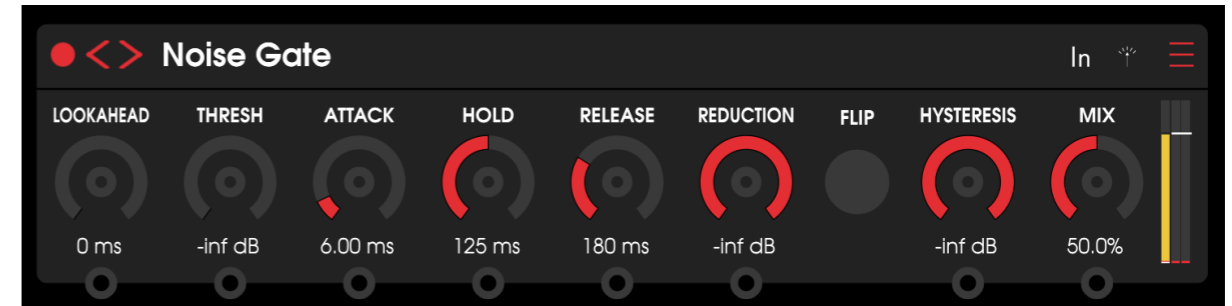
DESCRIPTION

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.

CONTROLS

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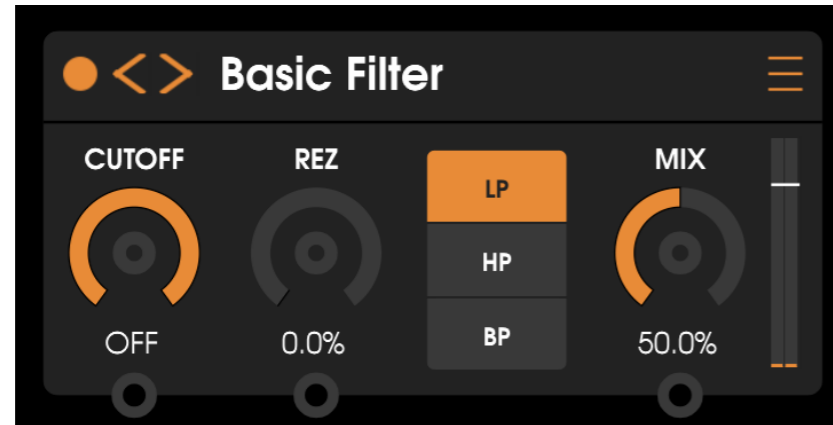
Flip: When enabled, the gate's threshold detection will be inverted so only sounds above the threshold will be gated.

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.

FILTERS

BASIC FILTER

DESCRIPTION

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

CONTROLS

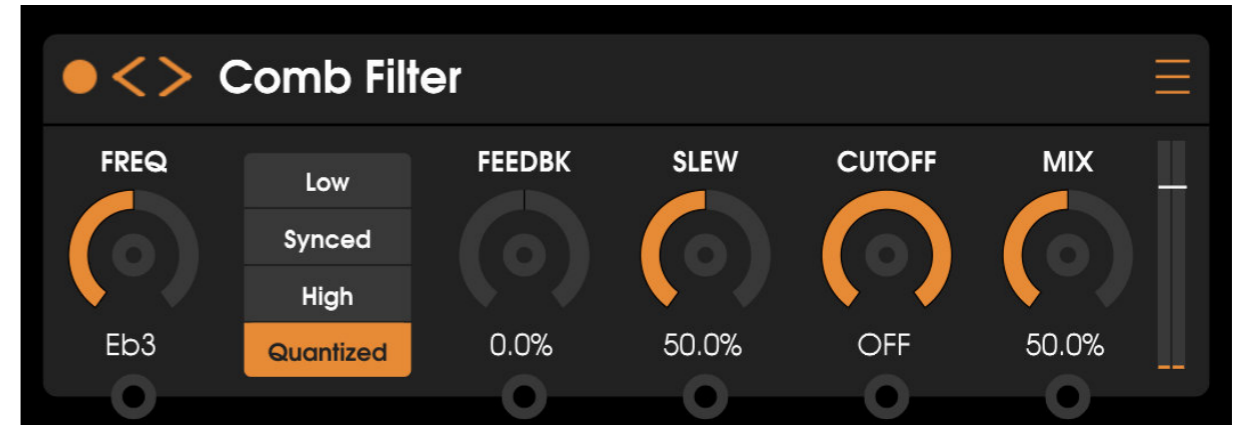**Cutoff**: Set the cutoff frequency of the filter.

Resonance: Sets the filter's resonance.

LP: Low-pass. Removes high frequencies.

HP: High-pass. Removes low frequencies.

BP: Band-pass. Permits a small range of frequencies.

COMB FILTER

DESCRIPTION

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.

CONTROLS

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
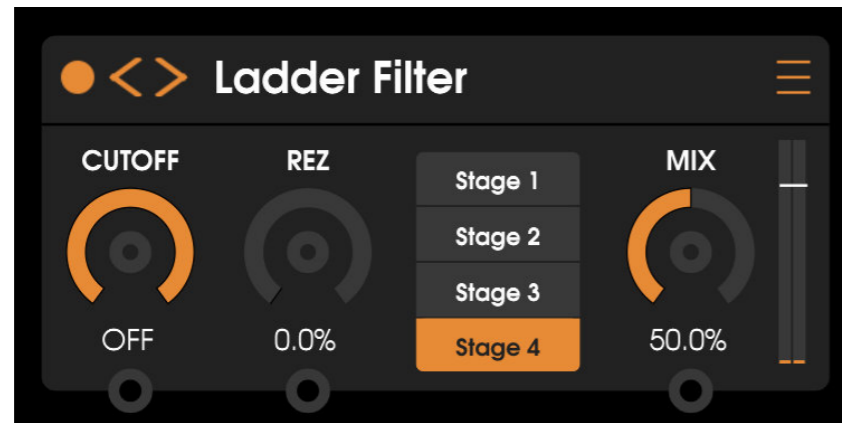
DESCRIPTION

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.

CONTROLS

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

DESCRIPTION

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

CONTROLS

Cutoff: Set the cutoff frequency of the filter.

LP: Low-pass. Removes high frequencies.

HP: High-pass. Removes low frequencies.

RESONATOR BANK
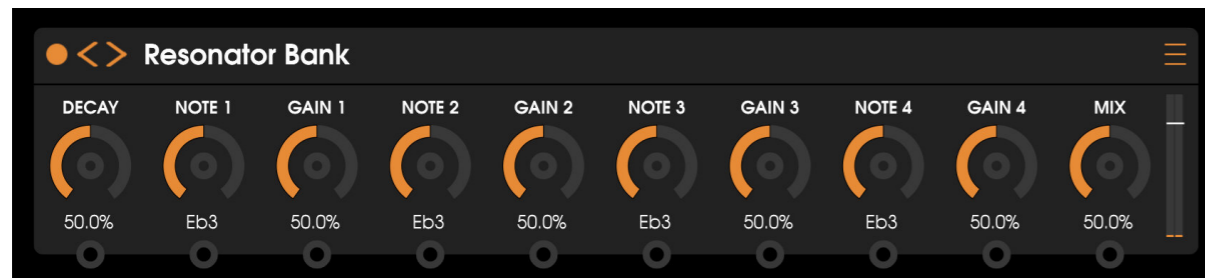
DESCRIPTION

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.
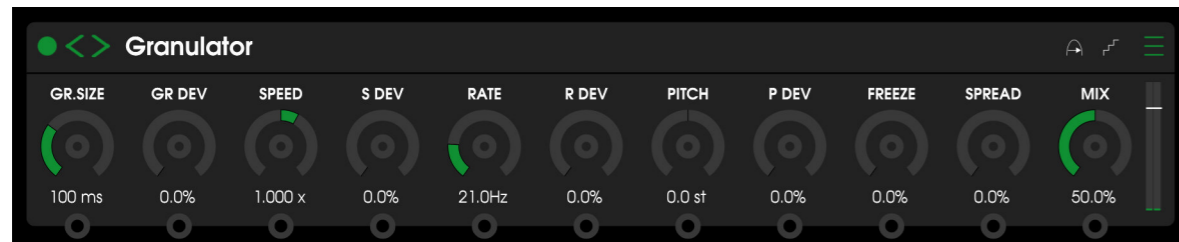
CONTROLS

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

DESCRIPTION

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.

CONTROLS

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 SHIFTER
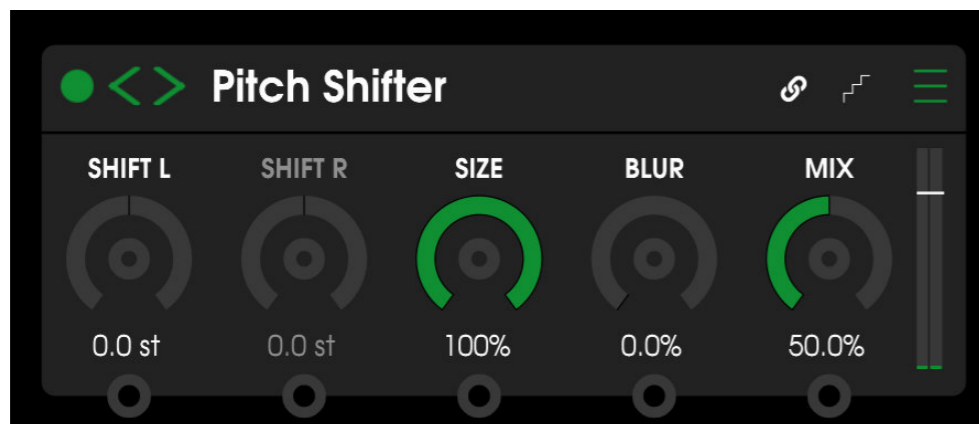
DESCRIPTION

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.

CONTROLS

**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
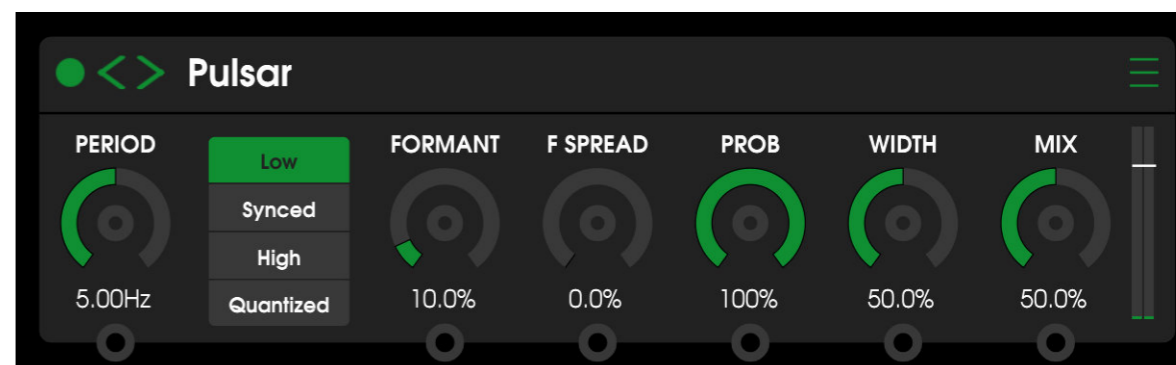
DESCRIPTION

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.

CONTROLS

Period: Sets how frequently a Pulsar window is generated.

Range: Sets the frequency range of the pulsar oscillator.

Low: 0-20 Hz. **High**: 0-1000 Hz.

Synced: The Pulsar's period is tied to even divisions of the host's tempo.

Quantized: The Pulsar's period is tied to scaled note values.

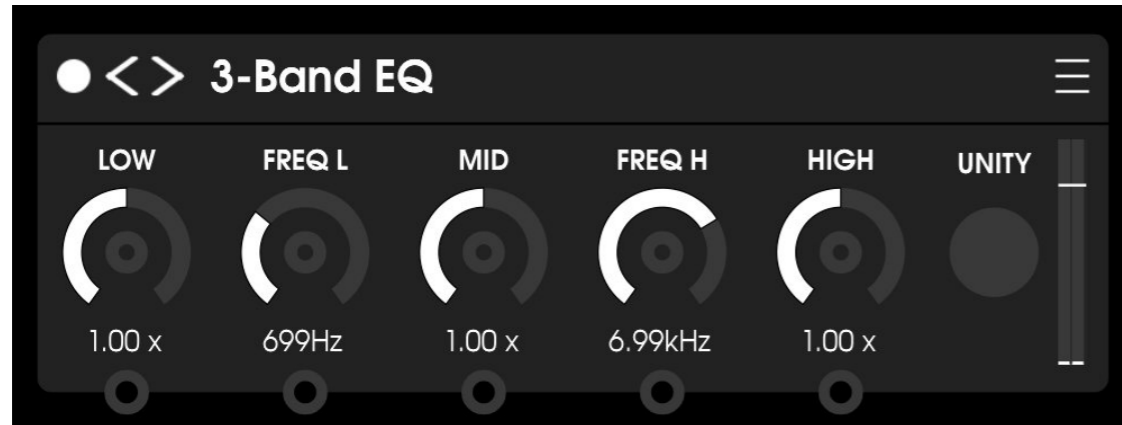**Formant**: Sets the width of the Pulsar's grain window.

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

DESCRIPTION

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.
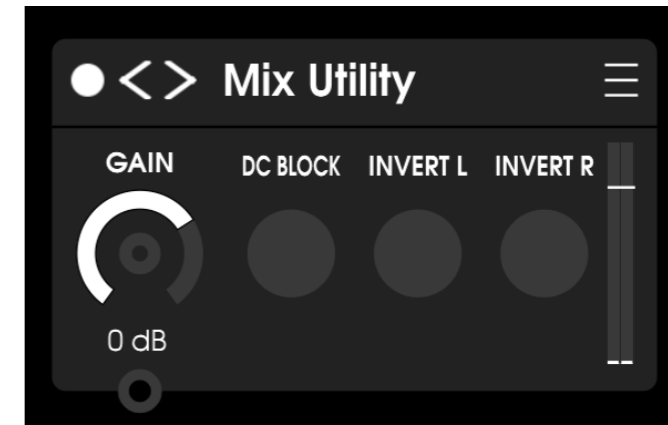
CONTROLS

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

DESCRIPTION

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

CONTROLS

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
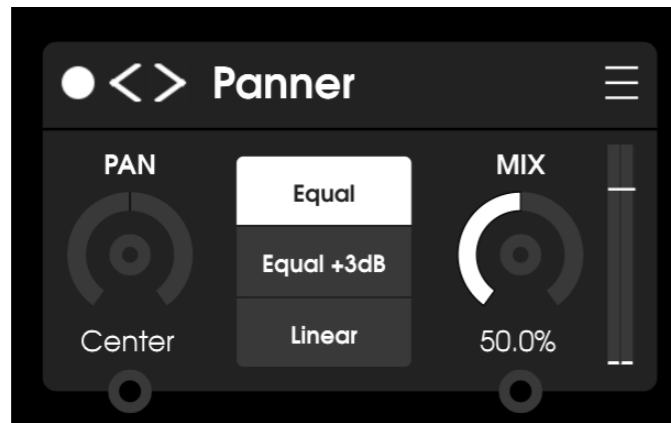
DESCRIPTION

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.
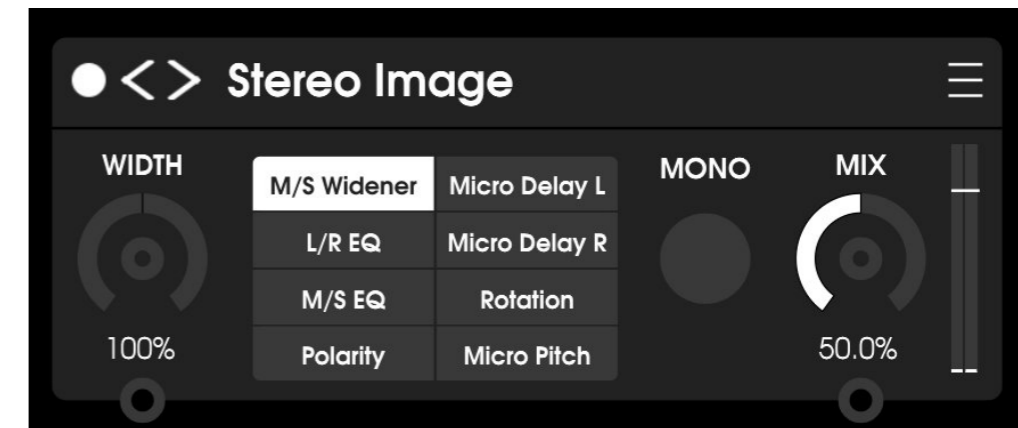
CONTROLS

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

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: 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

DESCRIPTION

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

CONTROLS

Width: Sets the intensity of the stereo effect.

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.

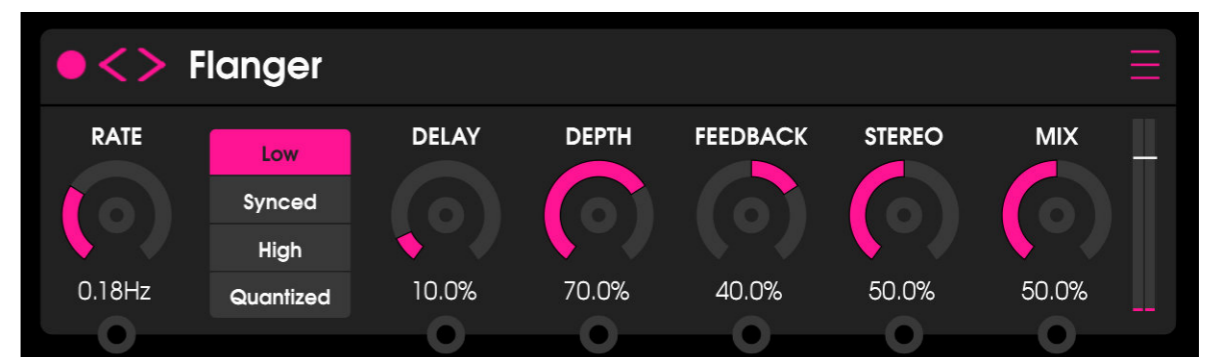# MODULATION

CHORUS
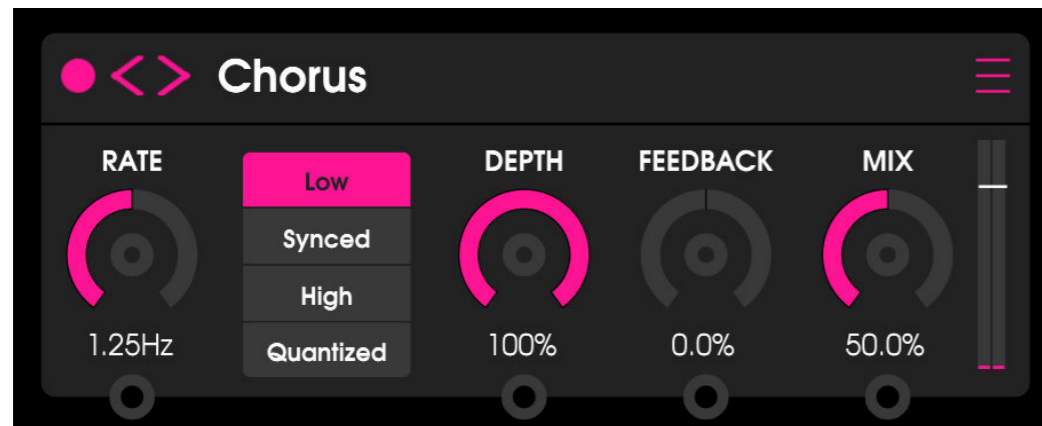
DESCRIPTION

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.

CONTROLS

Depth: Sets the intensity of modulation.

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

FLANGER

DESCRIPTION

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

CONTROLS

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
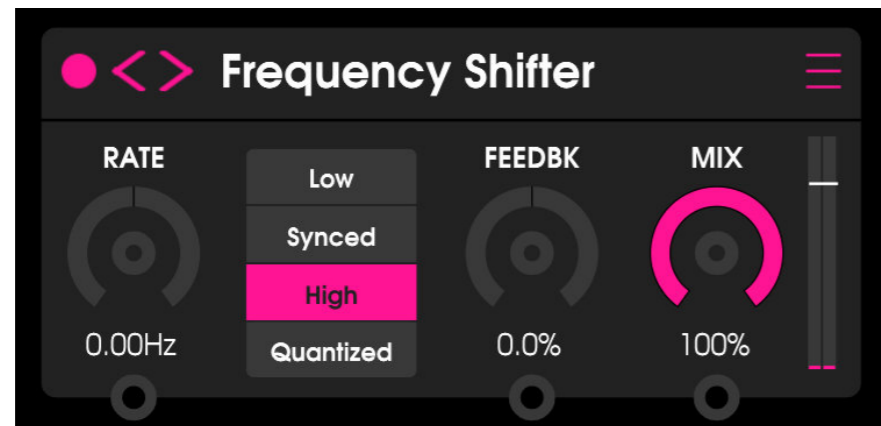
DESCRIPTION

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.

CONTROLS

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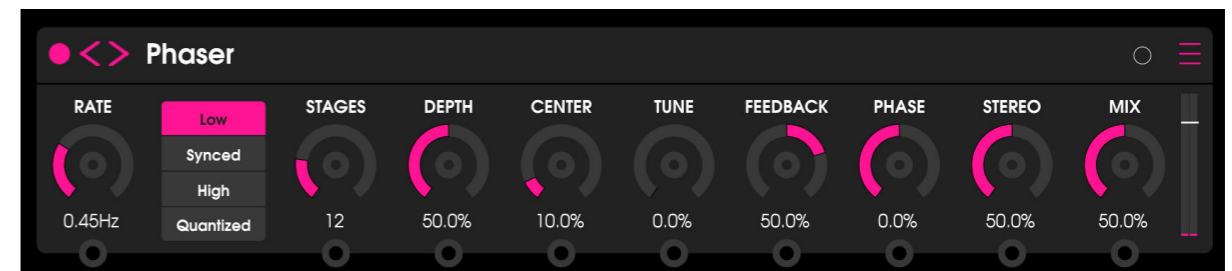
DESCRIPTION

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.

CONTROLS

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
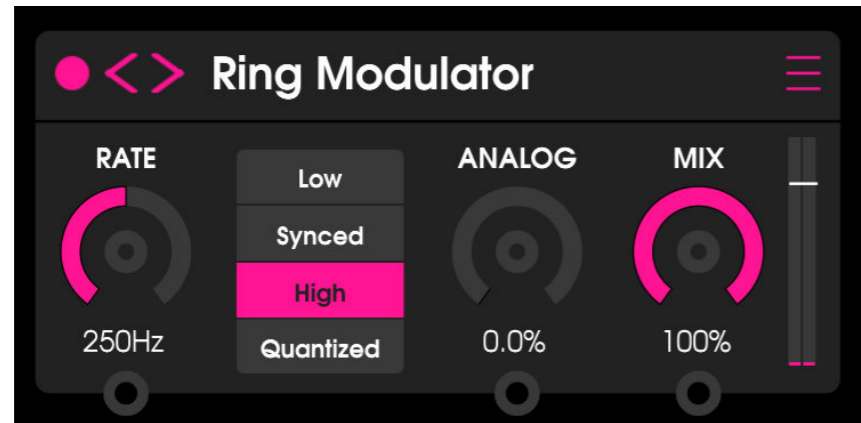
DESCRIPTION

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.

CONTROLS

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

ROTARY SPEAKER
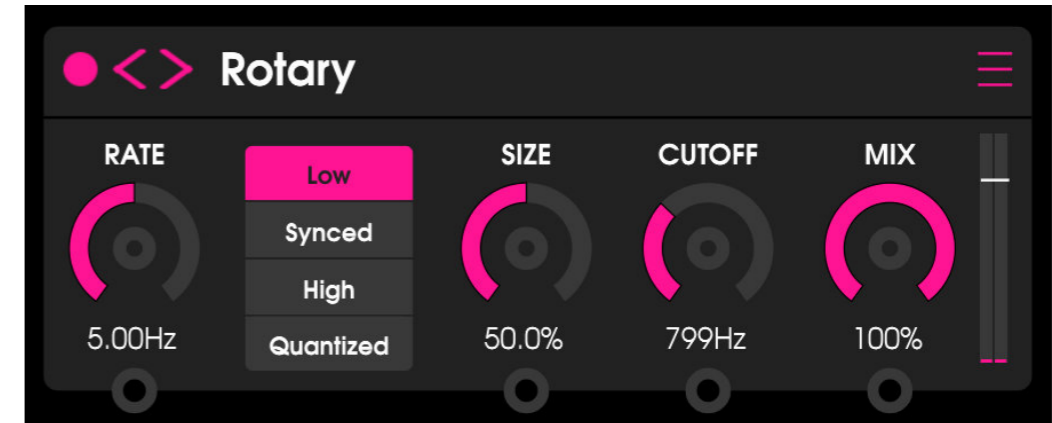
DESCRIPTION

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

CONTROLS

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.
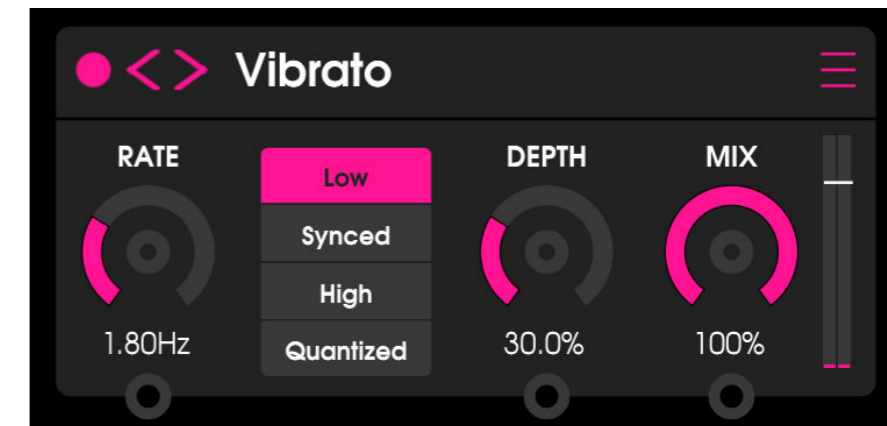
TREMOLO
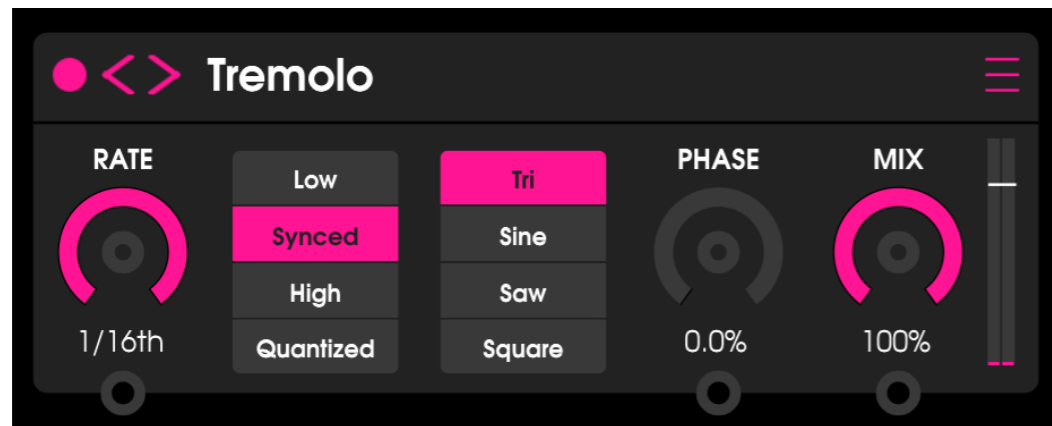
DESCRIPTION

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

CONTROLS

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.
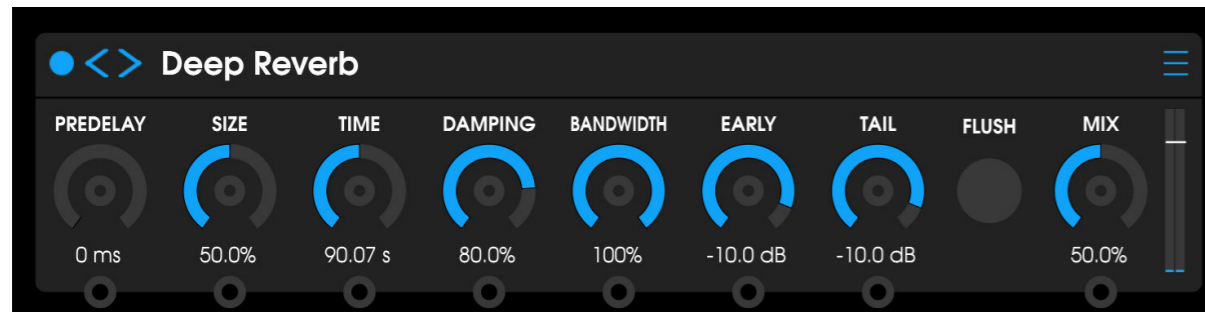
VIBRATO

DESCRIPTION

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

CONTROLS

Depth: Sets the intensity of modulation.

REVERBS

DEEP REVERB

DESCRIPTION

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.

CONTROLS

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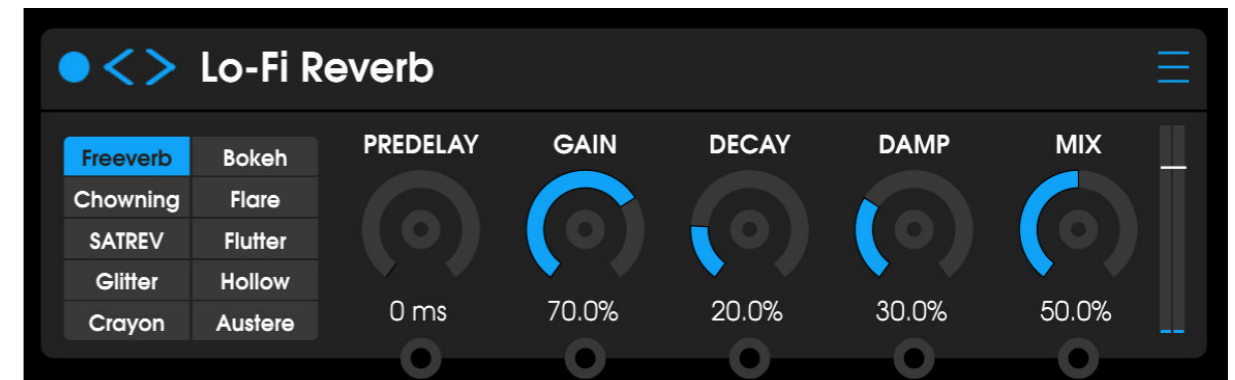
Bandwidth: Sets the reverb's bandwidth, or how much low-pass filtering is applied to the tail. Lower values will tighten up the echoes by removing the more diffuse tail.

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

DESCRIPTION

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.
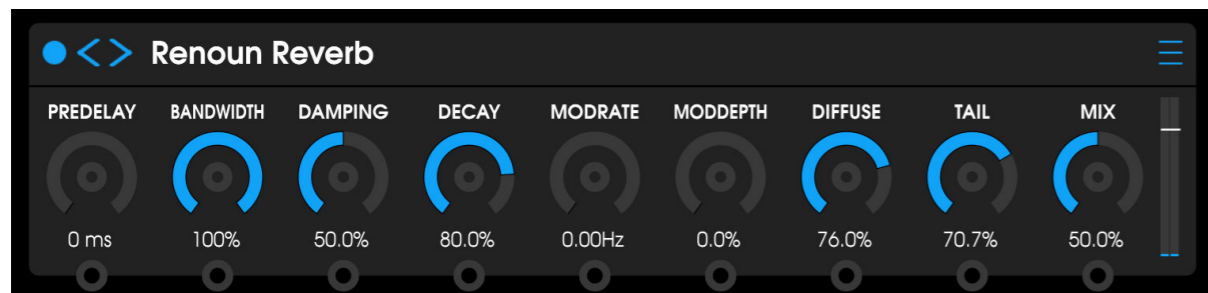
CONTROLS

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

DESCRIPTION

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.

CONTROLS

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.
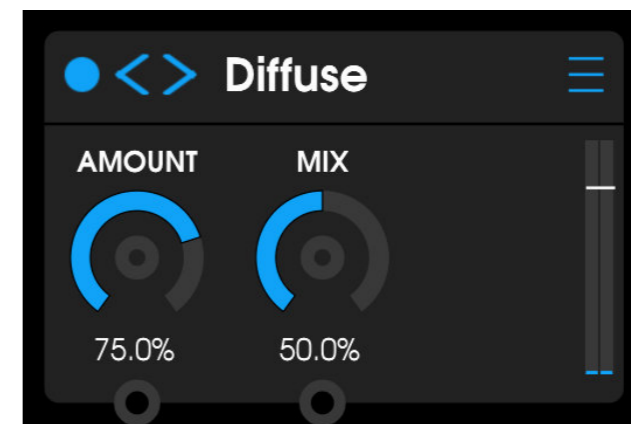
Decay: Sets the overall length of the reverb.

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

DESCRIPTION

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.

CONTROLS

Amount: Sets the amount of feedback for the internal diffusers. Higher values will lead to more echoes.
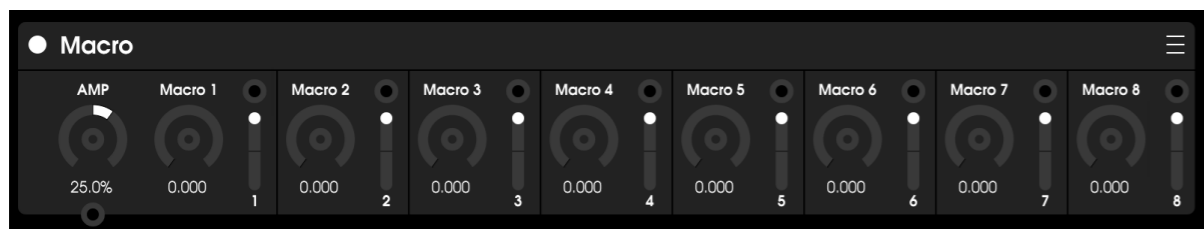
MODULATION

MACRO SYSTEM

Due to BYOME'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.
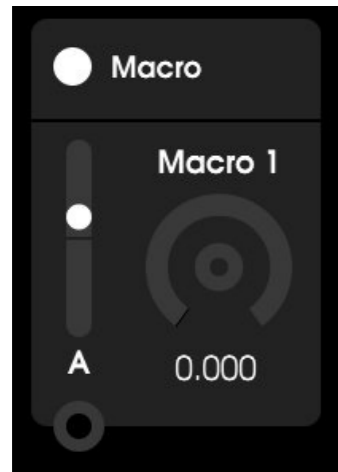
When creating a new patch, you will only see the Macro 1 knob. Double clicking the Macro header will expand the Macro Modulator into "patching view". As you connect modulation cables, more Macros will appear. There is a maximum of 32 Macros.

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.

MIDI IN
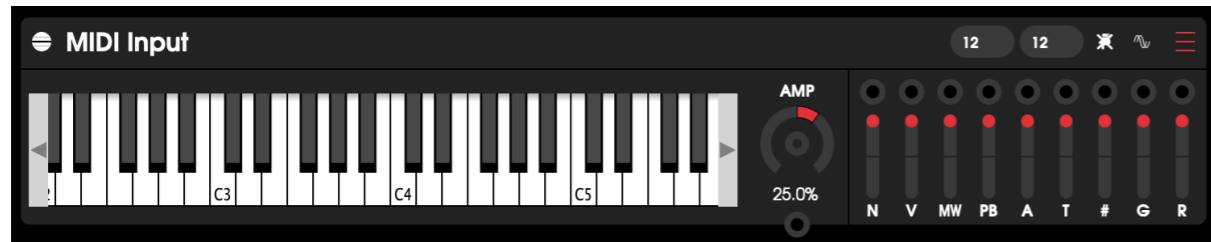
This modulator provides modulation outputs for all MIDI and MPE articulation sources.

The top two drop-down menus (that say 12 by default) control pitchbend depth, in semitones. In regular MIDI mode, this depth will affect the pitch wheel. In MPE mode, this will affect the pitch wheel, but not per-note bends. Per-note bends are +/- 48 semitones, which is the MPE standard. This will provide intuitive slides on MPE controllers like the ROLI Seaboard, the Sensel Morph, and the Linnstrument.

To activate MPE mode, click the MIDI box in the bottom left of the interface (by default, it will probably say "MIDI: All Channes"). From the drop-down menu, select MPE.

OUTPUTS

N: Note number. This will output a 0.0 for MIDI note 0, and a 1.0 for MIDI note 127. This is useful for, say, mapping the filter's cutoff to the current note.

V: Velocity. This is the value of the Note One message's Velocity, or how hard the note was pressed. In MPE Mode, this is the "Strike" parameter.

MW: Mod Wheel. This outputs the current value of MIDI CC message 1, which is usually the value of the Mod Wheel on your controller keyboard.

PB: Pitch Bend. This is the value of the Pitch Wheel in MIDI/Monophonic mode, or the value of the per-note bend in MPE/Polyphonic mode.

**A: Aftertouch**. This is the value of the current channel's Aftertouch (or "Pressure" in MPE terms).

T: Timbre (MPE Only). This is the value of the current note's Y position on the keyboard. If you are using a ROLI Seaboard or Sensel Morph, for instance, this is the vertical position of your finger on the note.

#: Voice number. This is a nomalized value based on which voice the active note is assigned to compared to the maximum number of allowed voices (i.e. in Poly 8 mode, this is 8 voices). This is only useful if the MIDI In modulator is set to be polyphonic. This is great for creating variation between each active voice.

G: Gate. This outputs a positive value while the current voice is active.

R: Release Velocity (MPE Only). This modulator will output a value based on how quickly you remove your finger from the keyboard. This can be used to create release-stage articulations.

MASTER ADSR
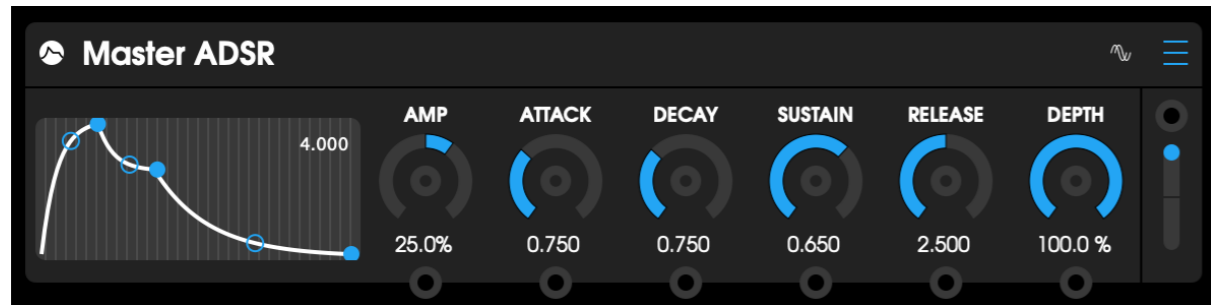
The Master ADSR is included by default in every patch. For a basic explanation of the ADSR and its controls, see the regular ADSR description in the Envelopes section.

The Master ADSR controls the amplitude of your synth patch. It is essentially always connected to an invisible GAIN control at the end of the synth voice (but before the BYOME row).

The Master ADSR is also available at each output on this modulator, allowing you to link other controls' modulation to the amplitude of the voice output. Each of these outputs is affected by the **AMP** control. The AMP control does not affect how much the ADSR controls the amplitude of the synth voice.

The **DEPTH** control decides how much the velocity of each note affects the amplitude of the assigned voice. At 100%, you will have the most expressive range, but softly pressed notes will be hard to hear. At 0%, the velocity does not affect the ADSR. By default, this is set to 50% for a good balance.

# ANALYZERS

COMPARATOR
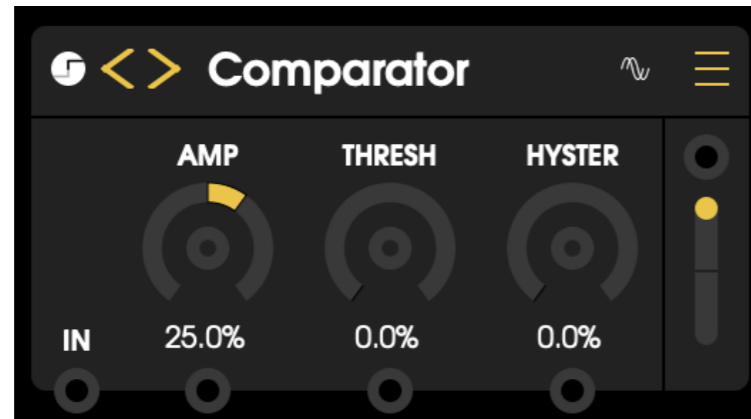
DESCRIPTION

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.

CONTROLS

Threshold: Sets the threshold for the comparator. If the input signal's amplitude exceeds this threshold, the output of the comparator will be true.

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

INPUT FOLLOWER
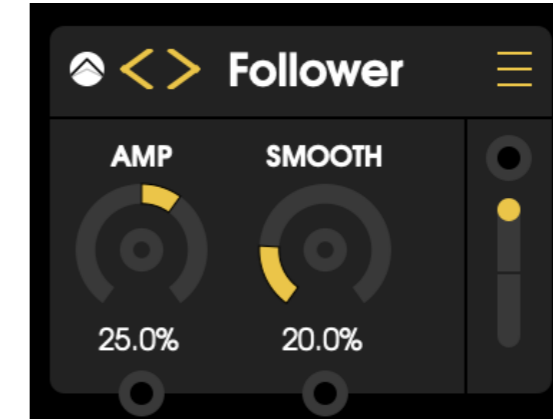
DESCRIPTION

This generates a modulation signal based off of the plugin's audio input. The modulation signal is based on the amplitude of the input. The Smooth parameter can help remove jitter from the modulation.

CONTROLS

Smooth: Determines how quickly the envelope follower responds to changes in amplitude. Increase smoothing if your modulation signal is too jumpy.
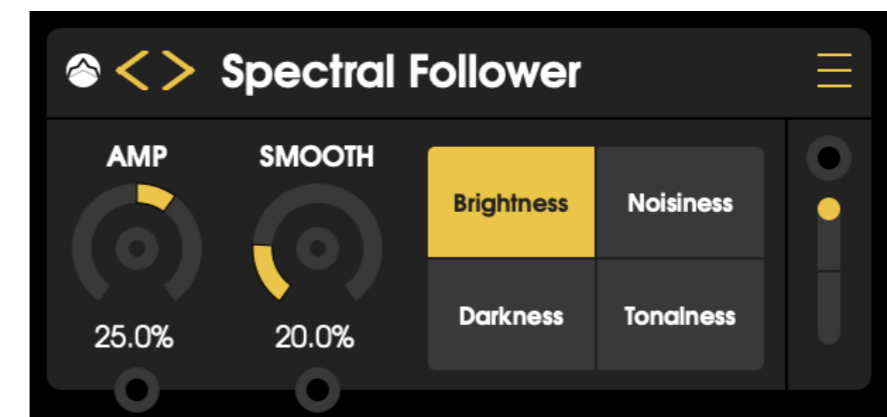
SCOPE
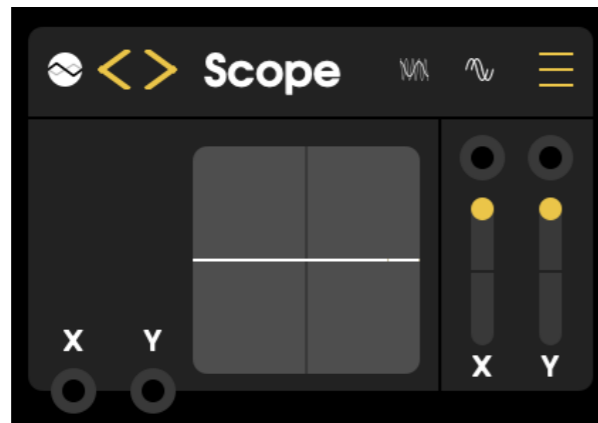
DESCRIPTION

This is a simple tool for visualizing your modulation signals.

CONTROLS

Lissajous Mode: When enabled, the signals will be plotted as XY coordinates on a Cartesian plane. Otherwise, the signals will be plotted on top of each other.

SPECTRAL FOLLOWER

DESCRIPTION

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.

CONTROLS

Smooth: Determines how quickly the envelope follower responds to changes in amplitude. Increase smoothing if your modulation signal is too jumpy.

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.

CONTROLLERS

META
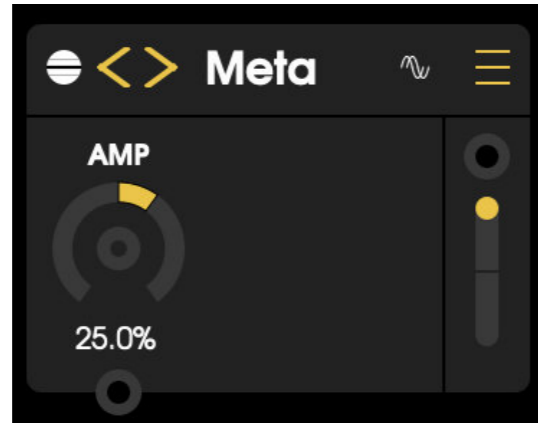
DESCRIPTION

This 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 input port, turning the Meta control into a summing modulation bus.

ROLI LIGHTPAD

DESCRIPTION

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 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.
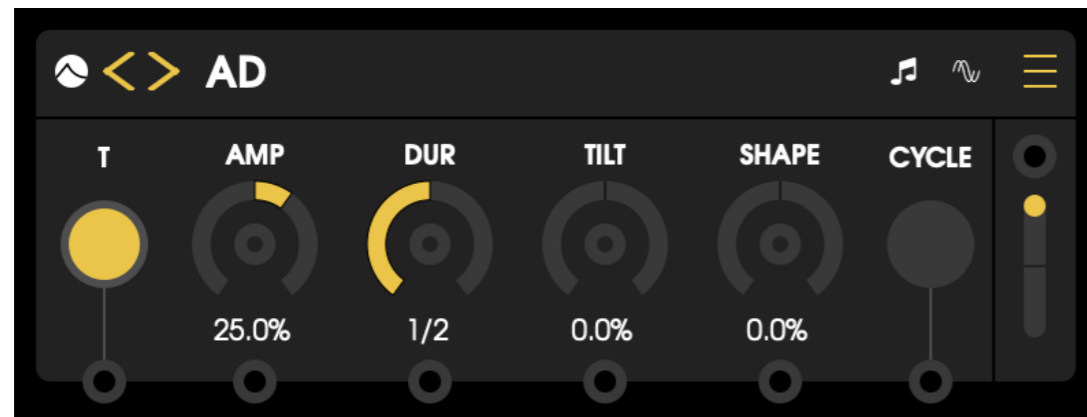
XY PAD
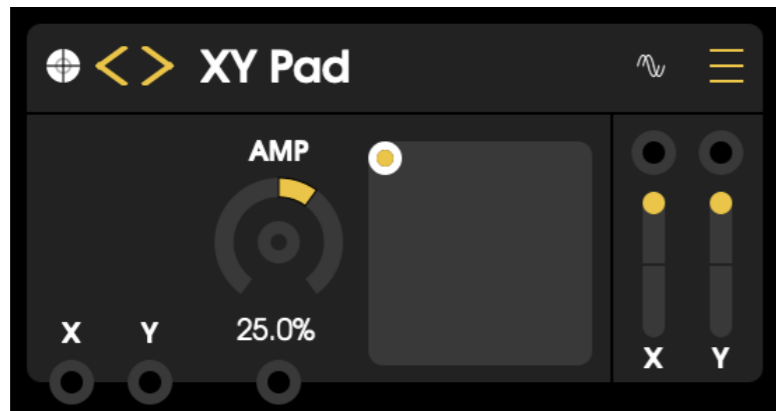
DESCRIPTION

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

ENVELOPES

AD ENVELOPE

DESCRIPTION

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.

CONTROLS

Duration: Sets the total duration on the envelope.

Tilt: Sets the attack-decay ratio of the envelope. At 100%, the envelope will have an instantaneous attack with a decay equal to the length of the envelope's duration. At -100%, the envelope will have an instantaneous decay. At 0%, the attack and decay lengths will be equal.

Shape: At 0%, the attack and decay stages will be linear. At 100%, the attack will be exponential while the decay will be logarithmic. At -100%, the decay will be exponential while the attack will be logarithmic.

ADSR ENVELOPE
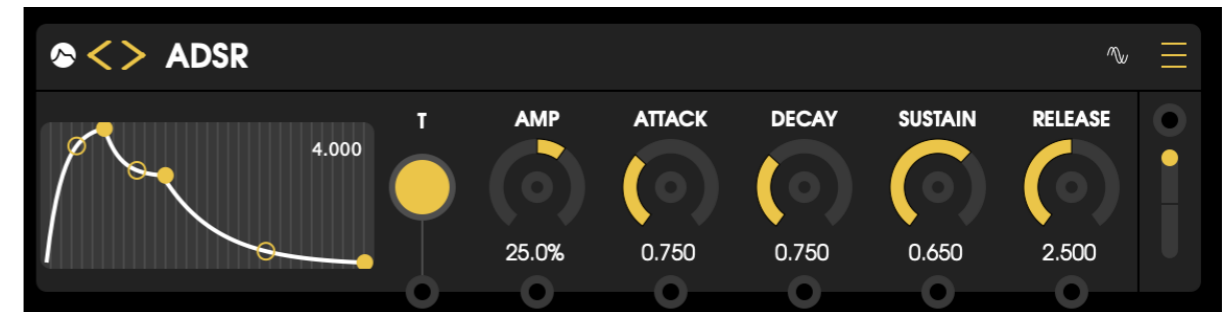
DESCRIPTION

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.

CONTROLS

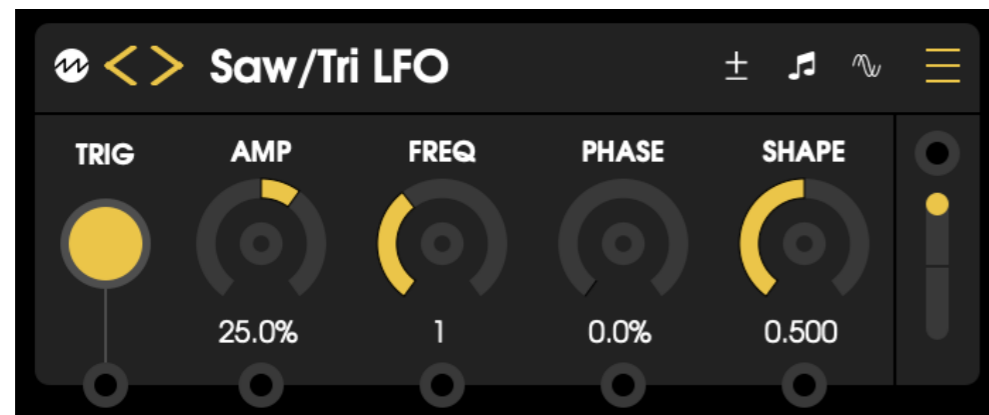**Attack**: How quickly the envelope rises to 1.0.

Decay: How quickly the envelope falls to the level set by Sustain.

Sustain: Set the level to decay to after the maximum envelope value is reached.

Release: How quickly the envelope falls to 0.0 after the input gate goes low.

LFOS

SAW/TRI LFO

DESCRIPTION

This is an LFO that generates a signal that can smoothly morph between a sawtooth, a triangle, and a ramp.

CONTROLS

Shape: This control determines the shape of the modulator's output, from ramp, to triangle, to saw, and then (at the very end) to ramp again. 0.0 = ramp, 0.5 = triangle, 0.98 = saw, 1.0 = ramp (again). It transitions smoothly to ramp at the very end to avoid abrupt direction flips when smoothly modulating this control.
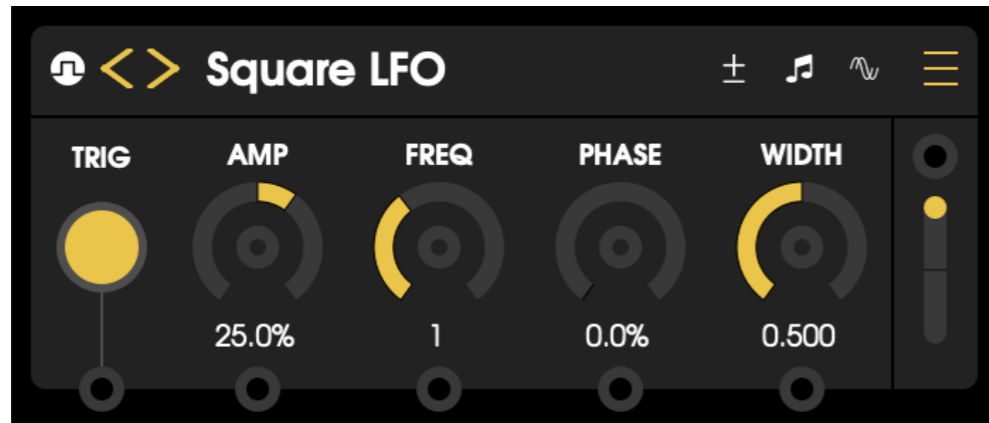
SINE LFO

DESCRIPTION

This simple LFO generates a cyclical sine wave at a given frequency.
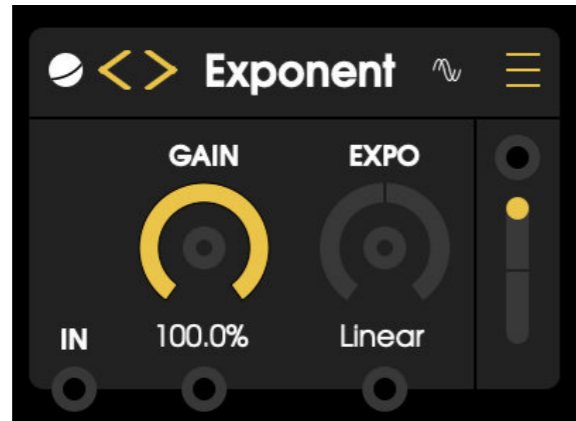
SQUARE LFO

DESCRIPTION

This LFO generates a pulse wave, often known as a square wave. The width of this pulse wave can be modified.

CONTROLS

Width: This control determines the pulse width of the modulator. At 50%, the output will be high and low for equal amounts of time.

MODIFIERS

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.

CONTROLS

Exponent: Sets the exponentiation of the input signal. At 12 o'clock, no exponentiation will occur. Clockwise, the signal will be made more exponential. Counter-clockwise, the signal will be made more logarithmic.

LOGIC MIX

DESCRIPTION

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.

OUTPUTS

+: Outputs the sum of both inputs.

-: Outputs the difference of both inputs.

X: Output the product of both inputs.

>: Outputs the value of the input with the greater value.

<: Outputs the value of the input with the lesser value.

RANDOM SOURCES

CHAOS
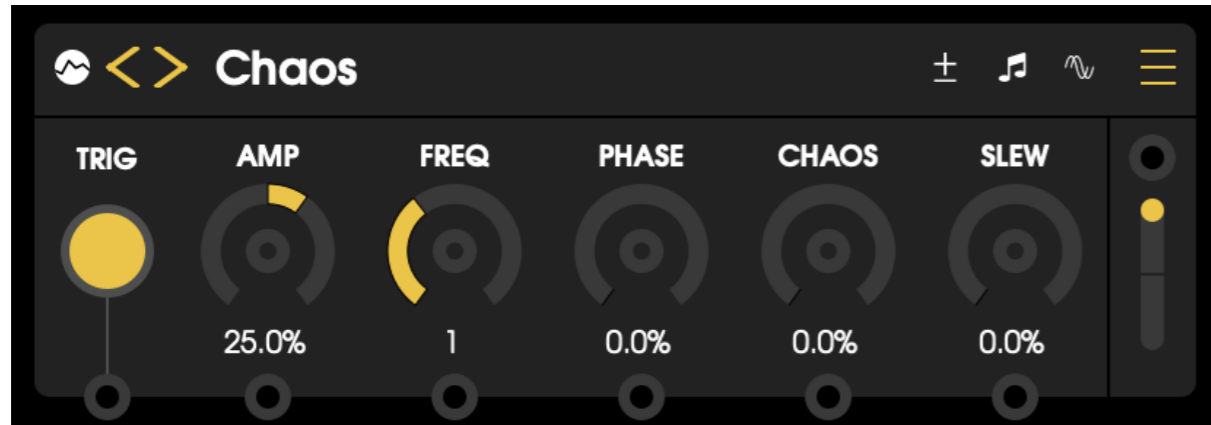
DESCRIPTION

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.

CONTROLS

Chaos: This control determines how unpredictable the output of this modulator is. At 0, there will be a stable triangle oscillation. At full, the output will be very unpredictable.

Slew: This control determines how quickly the modulator transitions between new values. With no slew, the new value is reached immediately (creating a random, stepped sequence). When slew is 100%, the time required to reach the new step is equal to the generation rate, meaning that the output will always be smooth.

LOOPING RANDOM
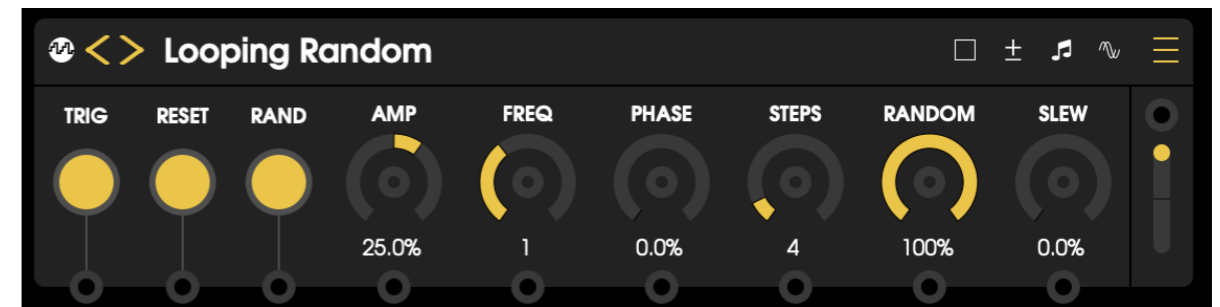
DESCRIPTION

This is an advanced version of the S+H Noise modulator where random values can be trapped and looped as a sequence. These values can then evolve slowly over time depending on the probability settings.

CONTROLS

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.

PROBABILITY GATE
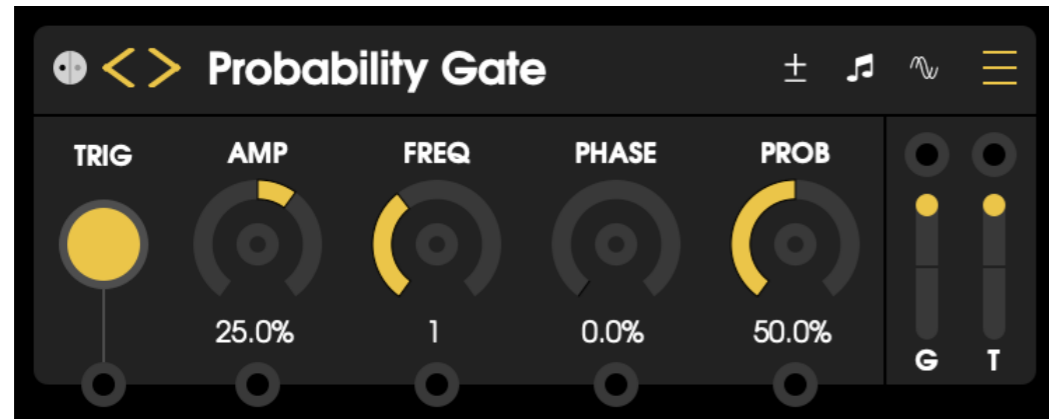
DESCRIPTION

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.

CONTROLS

Probability: This control determines how likely the gate will be high during each cycle. At 100%, the gate will always be high. At 0%, the gate will always be low.

S+H NOISE
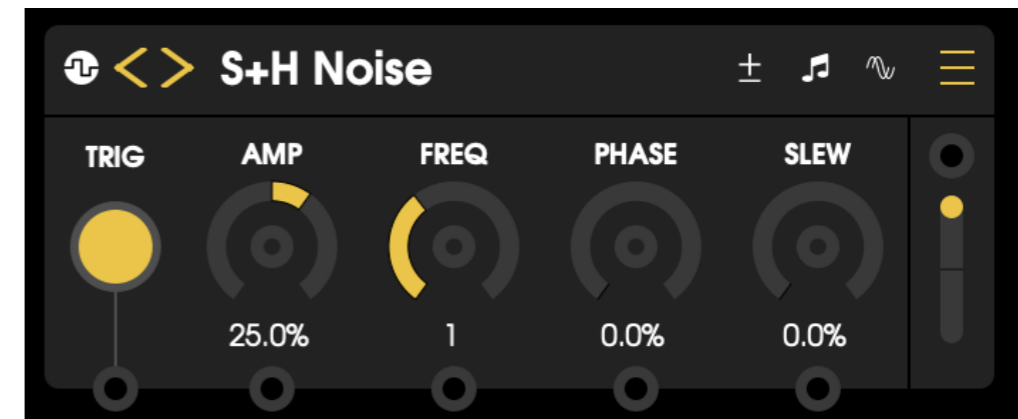
DESCRIPTION

This is similar to the LFOs, but it generates random, unpredictable values instead of repeating waveforms. Depending on the amount of slew, these sequences can be completely smooth or stepped.

CONTROLS

Slew: This control determines how quickly the modulator transitions between new values. With no slew, the new value is reached immediately (creating a random, stepped sequence). When slew is 100%, the time required to reach the new step is equal to the generation rate, meaning that the output will always be smooth.

SEQUENCERS

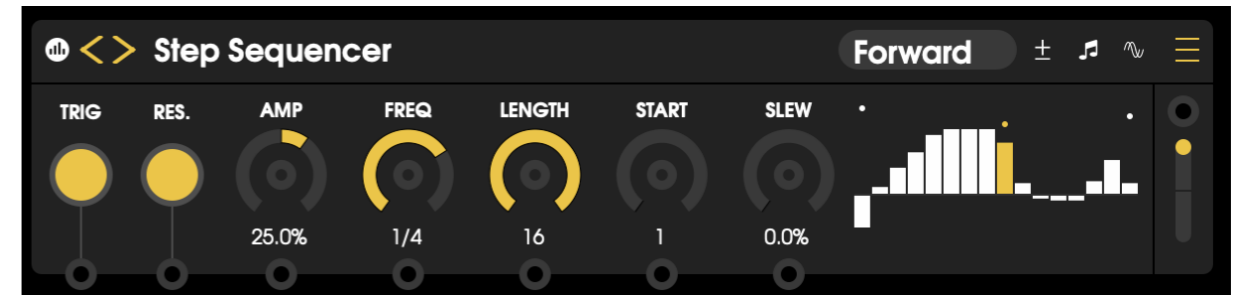## GATE SEQUENCER
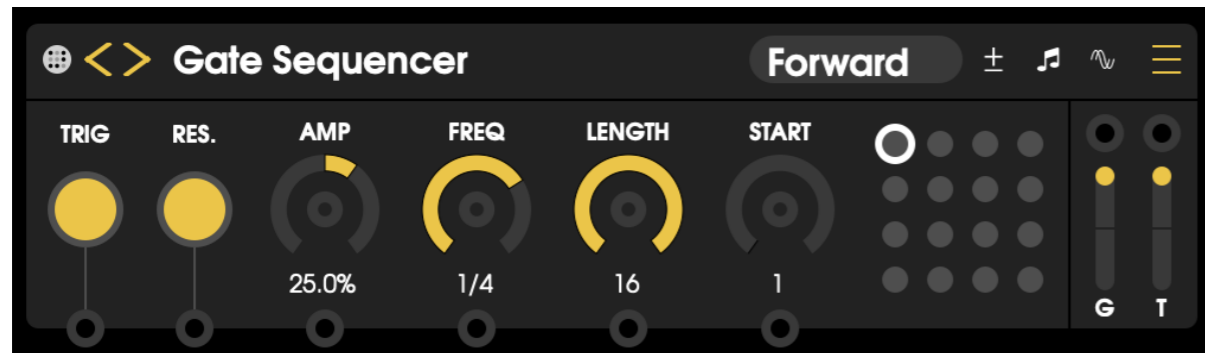
DESCRIPTION

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).

CONTROLS

Length: This control determines how many active steps are available in the sequence, from 1 to 16.

Start: This control determines which step of the sequence is the starting step. This step is selected when playback restarts or when a trigger is received on the reset input.

STEP SEQUENCER

DESCRIPTION

This is a bank of up to 16 values that can be accessed in sequential or random order. The sequence can run automatically or be manually triggered.

CONTROLS

Length: This control determines how many active steps are available in the sequence, from 1 to 16.

Start: This control determines which step of the sequence is the starting step. This step is selected when playback restarts or when a trigger is received on the reset input.

Slew: This control determines how quickly the sequencer transitions between steps. With no slew, the new step is reached immediately. As you increase slew, you will hear a more prominent slide between values.

CREDITS

LION was written by Joshua Dickinson and Michael Hetrick.

The visual theme of BYOME, Triad, and LION was created by Wes Milholen.

Thank you:

Many thanks to Emilie Gillet of Mutable Instruments for her many contributions to synthesizer design. LION was inspired by elements of Braids, Plaits, Warps, and Shruthi.

Stephen McCaul for his implementation of Summation Synthesis.

Lance Putnam for creating Gamma, the foundational DSP library that we use.

Aaron Anderson for allowing us to use his Wave Terrain research for our Terrain mixer algorithms.

Randy Jones at Madrona Labs for letting us use his intuitive microtuning organization system. Check out his great plugins!

Our spectacular preset creators:

Hans Besselink (Oddiction)

EmptyVessel

ignatius

John "Skippy" Lehmkuhl (PluginGuru.com)

Drew Madden (Audilepsy)

Liam Mclean (Joker)

Peter "Glaca" Mohamed

Alex Retsis

John Tejada

Tomavatars