



By

ADERN

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Introduction



Welcome to **FleXor**'s world of advanced modular synthesis and sound design. We would like to thank you for joining us in the everlasting quest for new and fascinating sonic landscapes. By integrating **FleXor** into your Creamware modular synthesizer system, you are now within reach of some of the best-sounding synthesis building blocks ever assembled in the digital domain, the result of years of hard development. We are proud to present you with this state-of-the-art package of exquisite filters, granular processors, unique oscillators, hyper-fast envelopes and accurate sequencers as well as our innovative one-of-a-kind wave shaper collection.

All **FleXor** components were made using our own original design, constructed from scratch with the most basic components such as adders, multipliers, dividers and buffers. This approach enables us to create brand new modules with a sound character totally different to the existing synthesis components. Functionally the modules explore new territories, creating new sound dimensions never before possible. We do not think about **FleXor** as a “Virtual Analogue” imitation. True to its origin, **FleXor** is nothing less than True Digital.

We hope you enjoy this powerful package and find it inspiring for your sonic passions. You are welcome to visit us at www.adern.com for news, updates, patches, presets, tutorials and information on additional products, and to join a growing community of modular **FleXorians**.

With best regards,
Your ADERN team

F.A.Q.

Q: I have a Scope Home card (3dsp's). What can I expect from flexor?

A: First, you have to acquire the SonicCore Modular II or III synthesizer system for your Scope Home card, and make sure you have the updated Scope software. Once this has been done, FleXor will work well with Scope or other 3 chips DSP cards. However, if you are a real modular freak, it is likely that in the future you would like to upgrade your hardware to have more DSP. Please be fully aware of each module's polyphony limits, by referring to the manual or popup help.

Q: Will flexor work on all modular versions?

A: FleXor has been thoroughly tested on Modular II and III, and it works very well in these systems. Regarding Modular I and FleXor, it is neither recommended nor supported by us, although it does work.

Q: Does flexor need any special resources?

A: All that FleXor needs is a functional SonicCore DSP card with Scope software version 5.0 and a running Modular II or III. To purchase a Creamware DSP card equipped with Modular II or III, or to purchase Modular III, check <http://www.soniccore.com> to find your nearest dealer.

Q: Will flexor work on all SFP versions?

A: FleXor works only on SFP software version 3.1c or later. The latest version is freely downloadable from <http://www.soniccore.com> for all SonicCore DSP cards.

Technical Issues

•How to get Flexor Envelopes to work

In order to operate any of the FleXor envelopes, the “E” (Envelope Sync or ‘E-sync’) and the Gate outputs on the MVC module must be connected to one another first. After this, all that is required is to connect the Gate output from the MVC module to the Gate input of the FleXor envelope. The cross-linking of E-sync and Gate on the MVC is unnecessary if you are using an additional Creamware envelope with E-sync in your patch.

•Some shapers result in a dc offset.

Many of the FleXor shapers are based on manipulation of amplitude and polarity, including splits and shifts at various amplitude ranges. Some of these processes eventually cause a DC shift at the output signal path. While this is considered to be a problem on “normal” audio processes such as mixing and mastering, it is a regular phenomenon on some synthesis processes.

If necessary, the DC offset can be eliminated by using the DC Blocking Filter available in Modular III, or with a 6db High-Pass Filter.

•Pattern Switching by ramp results in a small glitch

Pattern switching, unlike step switching, is an asynchronous function, meaning it responds slower than audio rate, causing glitching. However, using the NBL Phase Mod module may eliminate this phenomenon by shifting the preset-shifting ramp slightly forward. This is a limitation of the FleXor sequencers which has to be accepted for now. The issue may be resolved in the future, but currently we cannot offer a solution for this problem.

Polyphony

FleXor's program structure allows a module to be placed only on 1 DSP chip. This means that polyphony is limited by the number of times a module can fit on a DSP chip. The number of voices a module can handle is not dependent on the number of DSP chips you have on your system.

Modules that can handle a maximum of 7-voice polyphony or less are loaded in a monophonic state (with yellow side bars) by default, to avoid DSP overloads. To change a module to a polyphonic state (Blue side bars), right-click on the module and select "switch to polyphonic mode". Be sure to check the module's maximum polyphony, indicated in the reference section of this manual and in the popup help (click the "A" logo found on the surface of each module to access this feature).

While most of FleXor's modules allow enough polyphony, some of its modules like the filters, have stricter limits. This limitation can be overcome by using Creamware's PolyOut module in a smart way. Here is a demonstration patch:



The XF-2 filter, for instance, has a maximum of 2-voice polyphony. By converting the module to a monophonic state, and putting it after the PolyOut module, the filter can receive the polyphonic sum of the synthesizer, and process it all together. This way the synthesizer is capable of 10 or more voices!

Controls

Misc Controls



Fader

Click and drag the fader to change its value.



Text-Fader

Text-faders have custom parameters of control. Click and drag up/down to change a text-faders' value or setting.



Text-Field

A text-field is a place to input text or numbers. Click the text-field, type your desired input and press [enter] to set it.



Button

Toggles between on and off conditions. Click the button to toggle.



Draw

Draws fader positions with a single drag. Click and track the magic pencil on the draw view.

FleXor Knobs

Small Knob LED's



The smaller FleXor knobs have an inner circle of 12 LEDs, so each illuminated LED represents an increase at the value of 1/12th out of the total knob range.

Large Knob LED's



The bigger FleXor knobs have an inner circle of 16 LEDs, illuminating in purple light. While turning such knob, each illuminated LED represents an increase at the value of 1/16th out of the total knob range.

Knob Unipolar



A potentiometer with the range of 0 (extreme left) to 1 (extreme right). Click the knob and drag it around to change its value. The knob's outer led will flash softly at 0, 1/4, 1/2, 3/4 and 1.

Knob Bipolar



A potentiometer with the range of -1 (extreme left) to 1 (extreme right). Click the knob and drag to change its value. The knob's outer led will flash softly at -1, -1/2, 0, 1/2, and 1.

Position Indicator



Each FleXor knob has a position indicator which its outer end glows in pale yellow at exactly every 1/4 of the total knob range. These indicators make it easier to find those "in between" positions of 25%, 50% and 75% of the total knob range (as well as indicating both extremes).

Keyboard adjustment

Just as with all other standard SFP knobs and sliders, after a FleXor control has been selected (clicked) with the mouse, the keyboard's Left and Right arrow keys can be used to make fine adjustments while the Page Up and Page Down keys can also provide control for coarse adjustments.

Signal Connections

Audio I/O:

- Audio signals contain sound. They are signals that vary fast enough (i.e. with high frequency) to create hearable waveforms. They can, of course, be used as modulators too.

Frequency:

- Frequency signals are used to control the pitch of oscillators. Connect any “freq” output (from MVC, Pitch 2 CV, or note sequencer modules) to “freq” inputs on oscillators.

Control:

- Control signals contain sub-audio-rate varying values (low frequencies). The signals are used to modulate (control). These are generated by modules like LFOs, envelopes, sequencers, constant values and MVCs.

Gate:

- Gate signals are used to trigger envelopes, oscillators, LFOs and so on. There are two kinds of gate signals in FleXor: gate on (when note is pressed) and gate off (when note is released).

Oscillators

Oscillators are sound generators, generating a repetitive signal pattern resulting in a constant pitch. The pattern shape will determine the oscillator's timbre, while the frequency of repetition determines the pitch. Oscillators are usually used as the first building block in a synthesizer chain, but can also be used to modulate many of the synthesizer's functions.

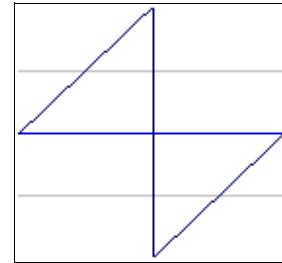
NBL Oscillators

All digital oscillators have a basic problem called "Aliasing". Aliasing occurs when the oscillator's harmonics are too high for the sample-rate to deal with, thus creating a side-effect called aliasing. Aliasing adds inharmonic frequencies to the oscillator making it sound dirty and grungy. Most oscillators employ aliasing filters to filter out those frequencies. The Alisaw oscillators use that "problem" as a musical flavor that can be an advantage at times.

FleXor's Non-Band-Limited oscillators have a linear ramp shape (Saw Up). The dropdown from their maximum to minimum takes 1 sample. The accurate linearity and the fast dropdown give these oscillators a rock-solid sound, ideal for tasks which need high precision. While having their advantages, NBL oscillators have their disadvantages. Therefore, FleXor has two kinds of NBL oscillators, each specialized for different things.

NBL Saw:

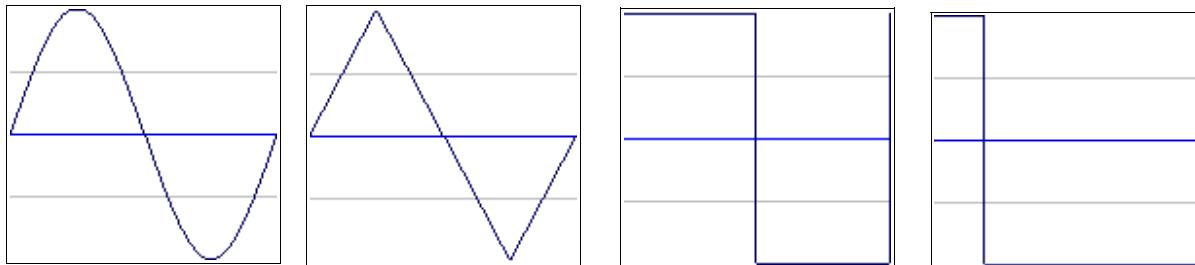
A perfect sawtooth with no aliasing. This oscillator plays only frequencies that do not alias. It has a wide spectrum resulting in a pure and clean sound. The side-effect is that high notes gradually drift out of tune. This oscillator is perfect for punchy bass sounds, and for higher detuned pads.



NBL Aliasaw:

A perfect aliasing sawtooth. This saw oscillator can track the frequency range accurately, leading to a side-effect of aliasing. The oscillator has a wide spectrum with a grungy and dirty sound. The dirt amount and its character can be determined by the "saw division" control on the oscillator. If shaped to triangle or sine, the aliasing frequencies will disappear. When shaped to sine or triangle aliasing will dramatically diminish.

FleXor also has a shaping section that allows you to shape the saw into the classic waveforms: triangle, sine and pulse.



FleXor's NBL oscillators are able to sync with sample accurate precision, making them ideal for sync sounds and for retriggering punchy bass sounds and percussive sounds. The oscillator's fast response makes them excellent for FM manipulation as well. Their accurate shape makes them perfect for use with wave-shapers, which may reduce the aliasing depending on the shape.

Aliasaw Simple



Aliasaw Simple Gate is a sawtooth-shaped Non-Band-Limited oscillator whose design results in output rich in high-frequency harmonic content and aliasing side-effects. Gate messages retrigger this module, causing it to start again from the initial point of its cycle.

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Gate:

This is an input for gate signals. Every time a signal is received at the Gate input, the generated wave-shape “jumps” back, and plays from the starting position.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

Aliasaw Simple Sync



Aliasaw Simple Sync is a sawtooth-shaped Non-Band-Limited oscillator whose design results in output rich in high-frequency harmonic content and aliasing side-effects. Sync messages retrigger this module, causing it to start again from the initial point of its cycle.

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Sync:

This is an input for audio signals. Every time a signal crosses the zero point while moving in a positive direction, the generated wave-shape “jumps” back, and plays from the oscillators starting point.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

Aliasaw Simple



Aliasaw Simple is a sawtooth-shaped Non-Band-Limited oscillator whose design results in output rich in high-frequency harmonic content and aliasing side-effects.

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

NBL Aliasaw Gate



Max. Polyphony: 13

This module is a sawtooth-shaped oscillator, featuring an adjustable non-interpolated “Hard” harmonic division. This function produces a unique sound, ranging from near-perfect ramp down to raw digital aliasing oscillator with various quantization errors. The Non-Band-Limited oscillator design results in a rich high-frequency harmonic content. The oscillator is retriggered by gate messages to start from the point indicated on the phase control.

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This function adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This function adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Gate:

This is an input for gate signals. Every time a signal is received at the Gate input, the generated waveshape “jumps” back, and plays from the starting position (set by the phase control).

Phase:

This function sets the point from which the generated wave cycle will start its playback every time a gate signal is been received.

Phase mod:

This is an input for a control signal (usually from another Oscillator or LFO), which is used to modulate the Phase (start-point) of the oscillator. The Phase Mod knob adjusts the amount of modulation applied to the Phase function, and is bipolar.

FM:

This is an input for control signals (usually from another Oscillator or LFO), which are used to modulate the Frequency of the oscillator. The FM knob adjusts the degree to which the control input modulates the Frequency.

Saw Division:

This function quantizes the number of cycles of the generated waveshape. This process is required for accurate pitch tracking. It creates an aliasing effect that can be varied by changing the saw division value. The values 1024, 2048, 4096 and 8192 are recommended.

Out:

This is the audio output of the Oscillator. The waveform emerges here.

NBL Aliasaw Sync



Max. Polyphony: 13

This module is a sawtooth-shaped oscillator, featuring an adjustable non-interpolated “Hard” harmonic division. This function produces a unique sound, ranging from near-perfect ramp down to raw digital aliasing oscillator with various quantization errors. The Non-Band-Limited oscillator design results in a rich high-frequency harmonic content. This oscillator can be synced to other audio sources to create new waveforms or to follow pitch.

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This function adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This function adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Sync:

This is an input for audio signals. Every time a signal crosses the zero point while moving in a positive direction, the generated wave shape “jumps” back, and plays from the oscillators starting point (set by the phase control).

Phase:

This function sets the point from which the generated wave cycle will start its playback every time a gate signal is been received.

Phase mod:

This is an input for a control signal (usually from another Oscillator or LFO), which is used to modulate the Phase (start-point) of the oscillator. The Phase Mod knob adjusts the amount of modulation applied to the Phase function, and is bipolar.

FM:

This is an input for control signals (usually from another Oscillator or LFO), which are used to modulate the Frequency of the oscillator. The FM knob adjusts the degree to which the control input modulates the Frequency.

Saw Division:

This function quantizes the number of cycles of the generated wave-shape. This process is required for accurate pitch tracking. It creates an aliasing effect that can be varied by changing the saw division value. The values 1024, 2048, 4096 and 8192 are recommended.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

NBL Saw Gate



This is a Non-Band-Limited oscillator which produces the most accurate sawtooth wave-shape digitally possible. The side-effect is that high notes gradually drift out of tune. The oscillator is re-triggered by gate messages to start from the point indicated by the Phase control.

Max. Polyphony: 13

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This function adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This function adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Gate:

This is an input for gate signals. Every time a signal is received at the Gate input, the generated waveshape “jumps” back, and plays from the starting position (set by the phase control).

Phase:

This function sets the point from which the generated wave cycle will start its playback every time a gate signal is been received.

Phase mod:

This is an input for a control signal (usually from another Oscillator or LFO), which is used to modulate the Phase (start-point) of the oscillator. The Phase Mod knob adjusts the amount of modulation applied to the Phase function, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

NBL Saw Sync



This is a Non-Band-Limited oscillator which produces the most accurate sawtooth wave-shape digitally possible. The side-effect is that high notes gradually drift out of tune. This oscillator can be synced to other audio sources to create new waveforms or to follow pitch.

Max. Polyphony: 13

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This function adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This function adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Sync:

This is an input for audio signals. Every time a signal crosses the zero point while moving in a positive direction, the generated wave-shape “jumps” back, and plays from the oscillators starting point (set by the phase control).

Phase:

This function sets the point from which the generated wave cycle will start its playback every time a gate signal is been received.

Phase mod:

This is an input for a control signal (usually from another Oscillator or LFO), which is used to modulate the Phase (start-point) of the oscillator. The Phase Mod knob adjusts the amount of modulation applied to the Phase function, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

NBL Transient



This module produces a single shot of the shortest possible bipolar sound. It generates a shot of 1 sample up and 1 sample down. It is useful for synthesizing percussive sounds, as well as excitation of feedback delay lines such as comb filters and resonators.

Max. Polyphony: 16

Gate:

This is an input for gate signals. Every time a signal is received at the Gate input, the module generates a transient of one sample up, one sample down.

Out:

This is the audio output of the Oscillator. The transient is generated here.

FR Oscillators

This is a collection of band limited oscillators. Band limiting oscillators dramatically reduces aliasing effects and sound much cleaner than aliasing oscillators. These oscillators have internal wave-shaping circuits that allow the creation of new and interesting shape modulation effects. These oscillators are especially nice for creating dense detuned leads and sequences.

FR Pulse



FR Pulse is a free running oscillator that generates a pulse shaped waveform whose width can be controlled and modulated. This oscillator module is Band-Limited to reduce aliasing and is capable of bi-polar FM (frequency modulation).

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

PW:

This parameter adjusts the Pulse Width (shape) of the oscillator.

Pwm(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Pulse Width. The PWM knob adjusts the amount of Pulse Width modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

FR Saw



FR Saw is a free running oscillator that generates a saw shaped waveform. This oscillator module is Band-Limited to reduce aliasing and is capable of bi-polar FM (frequency modulation).

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

FR Saw Width



(frequency modulation).

FR Saw Width is a free running oscillator that generates a saw shaped waveform whose width can be controlled and modulated. This oscillator module is Band-Limited to reduce aliasing and is capable of bi-polar FM

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

SW:

This parameter adjusts the Saw Width (shape) of oscillator.

SWm(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Saw Width. The SWm knob adjusts the amount of Saw Width modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

FR Saw Double



FR Saw Double is a free running oscillator that generates a waveform in the shape of two saws that over-ride each other. The width of the over-ride can be controlled and modulated. This oscillator module is

Band-Limited to reduce aliasing and is capable of bi-polar FM (frequency modulation).

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

SW:

This parameter adjusts the Saw Width (shape) of the oscillator.

SWm(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Saw Width. The SWm knob adjusts the amount of Saw Width modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

FR Saw Flip



FR Saw Flip is a free running oscillator that generates a waveform in the shape of two inverse saws that over-ride each other. The width of the over-ride can be controlled and modulated. This oscillator module is Band-Limited to reduce aliasing and is capable of bi-polar FM (frequency modulation).

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Flip:

This parameter Flips the Saw waveform. Changing the waveform of the oscillator.

Fmod(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Flip parameter. The Fmod knob adjusts the amount of Flip modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

FR Square



FR Square is a free running oscillator that generates a square shaped waveform. This oscillator module is Band-Limited to reduce aliasing and is capable of bi-polar FM (frequency modulation).

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

FR Square Double



FR Square Double is a free running oscillator that generates a waveform in the shape of two square waves whose relative widths can be controlled and modulated. This oscillator module is Band-Limited to reduce aliasing and is capable of bi-polar FM (frequency modulation).

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

PW:

This parameter adjusts the Pulse Width (shape) of the oscillator.

PWM(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Pulse Width. The PWM knob adjusts the amount of Pulse Width modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

FR Square Width



FR Square Width is a free running oscillator that generates a waveform in the shape of a square whose width can be controlled and modulated. This oscillator module is Band-Limited to reduce aliasing and is capable of bi-polar FM (frequency modulation).

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

SW:

This parameter adjusts the Square Width (shape) of the oscillator.

SWm(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Square Width. The SWm knob adjusts the amount of Square Width modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

FR Saw-Tri



FR Saw-Tri is a free running oscillator that generates a waveform whose variable shape ranges from saw to tri. The variable shape can be controlled and modulated. This oscillator module is Band-Limited

to reduce aliasing and is capable of bi-polar FM (frequency modulation).

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Shape:

This parameter adjusts the Shape of the oscillator's waveform.

Smod(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the waveform's Shape. The Smod knob adjusts the amount of Shape modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

FR Trapezoid



FR Trapezoid is a free running oscillator that generates a waveform whose variable trapezoid shape ranges from saw through square to triangle. The variable shape can be controlled and modulated. This oscillator module is Band-Limited to reduce aliasing and is capable of bi-polar FM (frequency modulation).

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Clip:

This parameter adjusts the amount of waveform clipping (antialiased).

Tilt:

This parameter adjusts the amount of waveform tilt.

Cmod(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the waveform's Clip. The Cmod knob adjusts the amount of Clip modulation derived from the control signal, and is bipolar.

Tmod(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the waveform's Shape. The Tmod knob adjusts the amount of Tilt modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

Special

Fixed Formant



Fixed Formant is a fixed formant oscillator which is based on a synced noise architecture. This creates a palette rich in overtones that don't transpose when the fundamental frequency does, but rather 'stay in place'

(like in most acoustic instruments). The formant control is used for transposing the overtones, and controlling the timbre of the oscillator.

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Sync:

This is an input for audio signals. Every time a signal crosses the zero point while moving in a positive direction, the generated wave-shape "jumps" back, and plays from the oscillators starting point.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Formant:

This parameter adjusts the formant of the oscillator. The formant of the oscillator represents an offset of a set of fixed frequencies that are emphasized.

Fmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Formant. The Fmod knob adjusts the amount of Formant modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

String Simple



String Simple is a simple implementation of the Karplus-Strong string model. Designed to emulate the physical behavior of a string, this module does not generate sound on its own. Instead it needs some kind of 'excitation' (input) to produce the string's vibrations. Excitations might be short noise bursts, envelope transients, or any other audio source.

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

In:

This is the audio input of the resonator, where you connect the audio signal to be processed.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Res:

This parameter adjusts the amount of resonance the string model provides.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

Source:

When on, a dry version of the audio input is mixed with the resonator.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

String Variable



String Variable is a more elaborate implementation of the Karplus-Strong “plucked string” model, equipped with an all-pass filter in the feedback loop to facilitate the modeling of enharmonic sounding strings and mallets. Designed to emulate the physical behavior of a string, this module does not generate sound on its own.

Instead it needs some kind of ‘excitation’ (input) to produce the string’s vibrations. Excitations might be short noise bursts, envelope transients, or any other audio source.

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Coarse:

This parameter adjusts the pitch of its Oscillator up or down in semi-tones.

Fine:

This parameter adjusts the pitch of its Oscillator up or down in cents (1/100 of a semitone).

Res:

This parameter adjusts the amount of resonance the string model provides.

Phase:

this parameter adjusts the phase of the signal going through the feedback path. Turning the knob to the left will result in inverted phase, while turning it to the right will result in a positive phase. This control can add enharmonic overtones to the string.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

Pmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Phase. The Pmod knob adjusts the amount of Phase modulation derived from the control signal, and is bipolar.

HP:

Adjusts a high-pass filter cutoff frequency. The high-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the low frequencies of the signal.

LP:

Adjusts a low-pass filter cutoff frequency. The low-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the high frequencies of the signal.

Source:

When on, a dry version of the audio input is mixed with the resonator.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

Tools

Soft Sync



Soft Sync is a tool that enables you to create a soft sync effect, which reverses the phase of the oscillator according to the master oscillator's polarity by using FM. To operate, connect the output of the master oscillator to the sync input and connect a frequency input (from MVC, Pitch2CV etc) to Freq. Then connect the frequency output to the carrier oscillator's Freq input.

Freq:

This is an input for a frequency signal, which is used to control the pitch of the oscillator. Here you connect the frequency output of an MVC, Pitch2CV or note sequencer.

Sync:

This is an input for audio signals. Every time a signal crosses the zero point while moving in a positive direction, the generated wave-shape “jumps” back, and plays from the oscillators starting point.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

LFOs

An LFO is a Low Frequency Oscillator. In other words, it is a periodic waveform generator that operates at a slow rate. LFOs are used to modulate various parameters in a synthesizer in a cyclic manner to create vibrato effects, slow sweeps and even rhythmic patterns. In Flexor there are only a few LFO's since all oscillators can also be used as LFOs. The LFOs in flexor's collection are those which can't be used as oscillators.

We advise you to try and experiment with flexor's oscillators combined with flexor's shapers for creating multiple LFO shapes. Or use one of our three special LFOs in the following collection.

Draw LFO



Draw LFO is a self-looping device that generates low frequency control signals based on what you 'draw'. To use this module simply draw your own LFO patterns in real-time by dragging the Draw fader. Draw LFO is ideal for creating your own wave-shapes to modulate any parameter on another module (cutoff, resonance, amplitude and so on).

Time:

This parameter controls the length of the loop, and gives more time to record movements

BPM:

This parameter sets the Time of the LFO in BPM.

Div:

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds:

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Hold:

This control helps tune the time it takes from the moment the blue dot has stopped moving and the time the module stops recording the movement.

Tmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate Time. The Tmod knob adjusts the amount of modulation which will be applied to Time parameter, and is

bipolar.

Xmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the blue dot position anywhere along the horizontal axis. The Xmod knob adjusts the amount of modulation which will be applied to the blue dot position, and is bipolar.

Ymod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the blue dot position anywhere along the vertical axis. The Ymod knob adjusts the amount of modulation which will be applied to the blue dot position, and is bipolar.

Xout:

This is the X output of this module, where the LFO's signal emerges.

Yout:

This is the X output of this module, where the LFO's signal emerges.

X Slider:

This slider sets horizontal position of the blue dot along the X axis.

Y Slider:

This slider sets vertical position of the blue dot along the Y axis.

Graphic Display :

This window shows movement of the LFO (the yellow dot). Drag the blue dot around to record your wave shape.

Draw LFO XY



(cutoff, resonance, amplitude and so on).

Draw LFO XY is a self-looping device that generates 2 low frequency control signals. To use this module simply 'draw' the two dimensional LFO patterns in real-time by dragging the point on the XY surface. Draw LFO is ideal for creating your own wave-shapes to modulate two parameters at a time on other modules

Time:

This parameter controls the length of the loop, and gives more time to record movements

Draw:

Drag this control to draw your waveform.

Dmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the blue line position. The Dmod knob adjusts the amount of modulation which will be applied to the blue line position, and is bipolar.

Tmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate Time. The Tmod knob adjusts the amount of modulation which will be applied to Time parameter, and is bipolar.

BPM:

This parameter sets the Time of the LFO in BPM.

Div:

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds:

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Hold:

This control helps tune the time it takes from the moment the blue dot has stopped moving and the time the module stops recording the movement.

Out:

This is the audio output of the Oscillator. The generated waveform emerges here.

Pattern LFO



Pattern LFO is a low frequency oscillator that offers a selection of 128 different complex waveforms. It can be tuned by HZ or BPM and has a retrigging function. It can be used for doing pattern based modulation of various parameters such as a filter's cutoff & resonance, an oscillator's frequency, the amplitude of an envelope and so on.

BPM:

This parameter sets the Time of the LFO in BPM.

Div:

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Hertz:

Shows the time of the LFO in Hertz (Cycles per second).

Freq:

This parameter adjusts the frequency (speed/rate) of the LFO.

Fmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the Frequency. The Fmod knob adjusts the amount of modulation which will be applied to the Frequency, and is bipolar.

Pmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the Phase. The Pmod knob adjusts the amount of modulation which will be applied to the Phase, and is bipolar.

Phase:

This parameter adjusts the phase of the LFO. 180 deg shifts the oscillator by half a cycle, while 360 shifts it by a whole cycle.

SR:

In order for this module to work correctly, please tune this controller to your hardware's Sample-Rate state (as listed under the "Samplerate" menu in scope).

Shapers

Wave-shapers are processors that affect the sound's amplitude range. This can be done by amplification, attenuation, reallocation and division of the amplitude range, amongst other techniques. Simple examples of wave-shapers are distortion, ring modulator, bit crusher and amplitude modulation effects.

FleXor's array of Shapers ranges from simple wave-shapers such as distortion and saturation, to more complex types such as oscillator re-shapers, harmonic wave-shapers and formantors.

Saturation:

This section has various Distortion modules. The distortion modules are great for guitar effects, drums, and distorted synths. They all consist of a combination of filters and saturation modules.

Saturation:

This section has various overdrive and saturation modules. The overdrive modules are great for guitar effects, drums, and distorted synths. The saturation effects are great for giving a warm presence to tracks or even whole mixes.

Special:

This section has special and uncommon distortion modules providing a variety of ways to alter your sounds, from bit crusher through PWM to wave reallocation.

Harmonic:

This section drastically alters the harmonic structure of an incoming waveshape by cutting them into several amplitude ranges and reconstructing them. On oscillators the may create complex FM or sync like effects, while on live audio it creates new esoteric distortion types. Destroy your sound like never before!

Osc Shapers:

These Shapers are responsible for changing oscillators' waveshapes. For example, you can extract 4 octaves from a sine or triangle wave (4 Sin Oct / 4 Tri Oct), or shape a triangle to sine (Tri 2 Sin).

Formantors:

These modules shape FleXor's NBL oscillators into formant-enriched waveforms and allow modulation of the formants' frequencies. Great for creating vocal-like sounds and formant oscillators!

FleXor's Shapers can be audio-rate modulated, routed in serial and parallel, and fed back into themselves in order to create new more complex and exciting wave-shaping. This shaper collection is unique. It enables the creation of new and unheard oscillator waveshapes, and sound-demolition effects never before possible.

Distortion

Fuzz



Fuzz is a fuzz-styled distortion effect. It is composed of filters and shapers which are designed to sound like the classic fuzz pedals. Its rocky hard-edged sound is ideal as a guitar or bass effect and is great for using together with real guitar amps.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Drive:

This control sets the amount of overdrive (distortion) applied to the incoming signal. Turning the knob to the right will gradually increase the Distortion effect.

Tone:

This control sets the brightness of the distortion effect applied to the incoming signal. Turning the knob to the right will result in a brighter sound.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Drive amount applied to the incoming signal. The Mod knobs adjust how much the Drive parameter is modulated by the control signals, and are bipolar.

Tmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Tone. The Mod knobs adjust how much the Tone parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Grinder



Grinder is a digital sounding distortion effect. It is composed of filters and shapers which are designed to sound like a nasty accident. Its hard-edged sound is ideal for making your audio unrecognizable.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Drive:

This control sets the amount of overdrive (distortion) applied to the incoming signal. Turning the knob to the right will gradually increase the Distortion effect.

Tone:

This control sets the brightness of the distortion effect applied to the incoming signal. Turning the knob to the right will result in a brighter sound.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Drive amount applied to the incoming signal. The Mod knobs adjust how much the Drive parameter is modulated by the control signals, and are bipolar.

Tmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Tone. The Mod knobs adjust how much the Tone parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Heavy Distortion



Heavy Distortion is a heavy-metal style distortion effect. It is composed of filters and shapers which are designed to sound like hard edged transistor based pedals. It's ideal as a guitar or bass effect and is great for using together with real guitar amps.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Dist:

This control sets the amount of distortion (overdrive) applied to the incoming signal. Turning the knob to the right will gradually increase the Distortion effect.

Tone:

This control sets the brightness of the distortion effect applied to the incoming signal. Turning the knob to the right will result in a brighter sound.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Distortion amount applied to the incoming signal. The Mod knobs adjust how much the Dist parameter is modulated by the control signals, and are bipolar.

Tmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Tone. The Mod knobs adjust how much the Tone parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Neo Distortion



Neo Distortion is a screaming-distortion effect. It is composed of filters and shapers which are designed to sound like a futuristic tube-screamer guitar pedal. It's ideal as a guitar or bass effect and is great for using together with real guitar amps.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Dist:

This control sets the amount of distortion (overdrive) applied to the incoming signal. Turning the knob to the right will gradually increase the Distortion effect.

Tone:

This control sets the brightness of the distortion effect applied to the incoming signal. Turning the knob to the right will result in a brighter sound.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Distortion amount applied to the incoming signal. The Mod knobs adjust how much the Dist parameter is modulated by the control signals, and are bipolar.

Tmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Tone. The Mod knobs adjust how much the Tone parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Smokey Distortion



Smokey Distortion is a classic overdrive effect. It is composed of filters and shapers which are designed to sound like vintage tube based guitar pedals. It's ideal as a guitar or bass effect and is great for using together with real guitar amps.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Dist:

This control sets the amount of distortion (overdrive) applied to the incoming signal. Turning the knob to the right will gradually increase the Distortion effect.

Tone:

This control sets the brightness of the distortion effect applied to the incoming signal. Turning the knob to the right will result in a brighter sound.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Distortion amount applied to the incoming signal. The Mod knobs adjust how much the Dist parameter is modulated by the control signals, and are bipolar.

Timod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Tone. The Mod knobs adjust how much the Tone parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Vanilla Distortion



Vanilla Distortion is a thick-sounding distortion effect. It is composed of filters and shapers which are designed to sound lush and thick. It's ideal for a guitar or bass effect used on power-chords, and is great for using together with real guitar amps.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Dist:

This control sets the amount of distortion (overdrive) applied to the incoming signal. Turning the knob to the right will gradually increase the Distortion effect.

Tone:

This control sets the brightness of the distortion effect applied to the incoming signal. Turning the knob to the right will result in a brighter sound.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Distortion amount applied to the incoming signal. The Mod knobs adjust how much the Dist parameter is modulated by the control signals, and are bipolar.

Tmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Tone. The Mod knobs adjust how much the Tone parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Harmonic

Squash



Squash is a shaper that 'Squashes' the incoming audio signal. This is done by dividing the signal into several amplitude ranges and then smoothly folding them back to be aligned side by side. Squash starts its division at both

clipping points and the smoothness of the divisions avoids sharp amplitude changes. The Squash process sounds a bit like FM in that it changes the harmonic structure of the wave according to its amplitude.

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Squash:

This control (the largest knob) sets the amount of Squash effect applied to the incoming signal. At its zero position (extreme left), the incoming signal is slightly saturated. Turning the knob to the right will gradually increase the squash effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges

Crumble



This shaper 'Crumbles' the incoming audio signal: it divides the signal into several amplitude ranges (crumble factor), and aligns them side by side. Crumble starts its position "DC Shifted" (see DC Shifter). The amount knob sweeps from minimum to maximum

division as indicated by the Crumble factor. The crumble process sounds extremely aggressive, as it changes the harmonic structure of the wave according to its amplitude.

Max. Polyphony: 16

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control (the largest knob) sets the amount of Crumble effect applied to the incoming signal. At its zero position (extreme left), the incoming signal is DC-shifted (see DC Shift module) at the first Crumble stage. Turning the knob to the right will gradually increase the Crumble effect.

Crumb:

The value indicated by this text-fader sets the 'Crumb factor': the number of times the process will be multiplied (done again) internally. To set this value, click and drag up/down.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Harm



This shaper 'Harms' the incoming audio signal. This is done by dividing the signal into several amplitude ranges (Harm factor), and aligning them side by side. Harm starts its division at the zero crossing point. The amount knob sweeps from minimum to maximum

division as indicated by the harm factor. The Harm process sounds extremely aggressive, as it changes the harmonic structure of the wave according to its amplitude.

Max. Polyphony: 16

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control (the largest knob) sets the amount of Harm effect applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Harm effect.

Harm:

The value indicated by this text-fader sets the ‘Harm factor’: the number of times the process will be multiplied (done again) internally. To set this value, click and drag up/down.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Shred



This shaper 'Shreds' the incoming audio signal. This is done by dividing the signal into several amplitude ranges (Shred factor), and aligning them side by side. Shred starts its division in two points: the zero crossing point and the clipping point. The amount knob sweeps from minimum to maximum division as indicated by the shred factor. The Shred process sounds extremely aggressive, as it changes the harmonic structure of the wave according to its amplitude.

Max. Polyphony: 16

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control (the largest knob) sets the amount of Shred effect applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Shred effect.

Shred:

The value indicated by this text-fader sets the 'Shred factor': the number of times the process will be multiplied (done again) internally. To set this value, click and drag up/down.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Squeeze



This shaper 'Squeezes' the incoming audio signal. This is done by dividing the signal into several smooth amplitude ranges ('Squeeze factor' or Octave control), and folding them smoothly back inverted to be aligned side by side. The smoothness of the divisions avoids sharp amplitude changes. Squeeze starts its division at the negative clipping point. At its zero position, the Pulse Width relation between the incoming signal sides will be offset slightly. The amount knob sweeps from minimum to maximum division as indicated by the squeeze factor. The Squeeze process sounds a bit like FM, as it changes the harmonic structure of the wave according to its amplitude.

Max. Polyphony: 7

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control (the largest knob) sets the amount of Squeeze effect to be applied to the incoming signal. At its zero position (extreme left), the Pulse-Width relation between the incoming signal sides will be offset slightly. Turning the knob to the right will gradually increase the Squeeze effect.

Octave:

This is the 'Squeeze factor'. Squeeze has a special behaviour. It creates octaves out of sine oscillators when the amount control is at its extreme right. Click and drag up/down to determine which octave will be generated at the maximum amount of the process.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Squish



This shaper 'Squishes' the incoming audio signal. This is done by dividing the signal into several smooth amplitude ranges (Squish factor), and folding them smoothly back inverted to be aligned side by side. The smoothness of the divisions avoids sharp amplitude changes.

Squish starts its division at both clipping points. The amount knob sweeps from minimum to maximum division as indicated by the squish factor. The Squish process sounds a bit like FM, as it changes the harmonic structure of the wave according to its amplitude.

Max. Polyphony: 9

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control (the largest knob) sets the amount of Squish effect applied to the incoming signal. At its zero position (extreme left), the incoming signal is slightly saturated. Turning the knob to the right will gradually increase the squish effect.

Squish:

The value indicated by this text-fader sets the 'Squish factor' : the number of times the process will be multiplied (done again) internally. To set this value, click and drag up/down.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Warp



This shaper 'Warps' the incoming audio signal: it divides the signal into several amplitude ranges (Warp factor), and folds them back, inverted, to be aligned side by side. Warp starts its division at both clipping points. The amount knob sweeps from minimum to maximum division as indicated by the Warp factor. The Warp process sounds a bit like sync, as it changes the harmonic structure of the wave according to its amplitude.

Max. Polyphony: 12

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control (the largest knob) sets the amount of Warp effect applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Warp effect.

Warp:

The value indicated by this text-fader sets the ‘Warp factor’: the number of times in which the process will be multiplied (done again) internally. To set this value, click and drag up/down.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Cheby 16



Cheby 16 is a wave-shaper that uses a sine waveform to generate the first 16 harmonics of the fundamental frequency. The harmonics are aligned at 90 degrees which makes it ideal for harmonic shaping. It has one input and 16 outputs (one for each harmonic overtone). It can also be used on any complex audio signal as a wave-shaper.

In:

This is the audio input of this shaper, where you connect the audio signal (Usually a sine wave) to be processed.

Outs 1-16:

Each of the outputs represents a harmonic overtone derived from the input. If the input is fed with a normalized sine wave, each output will generate a sine with a frequency multiplied by the output number (for instance, output 3 will generate a sine which is an octave and a fifth higher than the input – harmonic no. 3). however experimenting with other shapes than sine, and live audio is also beneficial!

Cheby 8



Cheby 8 is a wave-shaper that uses a sine waveform to generate the first 8 harmonics of the fundamental frequency. The harmonics are aligned at 90 degrees which makes it ideal for harmonic shaping. It has one input and 8 outputs (one for each harmonic overtone). It can also be used on any complex audio signal as a wave-shaper.

In:

This is the audio input of this shaper, where you connect the audio signal (Usually a sine wave) to be processed.

Outs 1-4:

Each of the outputs represents a harmonic overtone derived from the input. If the input is fed with a normalized sine wave, each output will generate a sine with a frequency multiplied by the output number (for instance, output 3 will generate a sine which is an octave and a fifth higher than the input – harmonic no. 3). however experimenting with other shapes than sine, and live audio is also beneficial!

Cheby 4



outputs (one for each harmonic overtone). It can also be used on any complex audio signal as a wave-shaper.

In:

This is the audio input of this shaper, where you connect the audio signal (Usually a sine wave) to be processed.

Outs 1-4:

Each of the outputs represents a harmonic overtone derived from the input. If the input is fed with a normalized sine wave, each output will generate a sine with a frequency multiplied by the output number (for instance, output 3 will generate a sine which is an octave and a fifth higher than the input – harmonic no. 3). however experimenting with other shapes than sine, and live audio is also beneficial!

Additive 16



It has two inputs (sine and cosine – sine at 90 deg) and 16 outputs (one for each harmonic overtone).

Sin:

To receive the desired results (a series of sine waves at the output) connect a sine oscillator here.

Cos:

To receive the desired results (a series of sine waves at the output) connect a 90deg phase shifted version of the sine oscillator (cosine) here. You may use a cosine oscillator.

Outs 1-16:

Each of the outputs represents a harmonic overtone derived from the input. If the inputs are fed with normalized sine and cosine waves, each output will generate a sine with a frequency multiplied by the output number (for instance, output 3 will generate a sine which is an octave and a fifth higher than the input – harmonic no. 3). however experimenting with other shapes than sine, and live audio is also beneficial!

Additive 8



Additive 8 is a wave-shaper that uses sine and cosine waveforms to generate the first 8 harmonics of the fundamental frequency. The harmonics are aligned at 0 degrees which makes it ideal for additive synthesis. It has two inputs (sine and cosine – sine at 90 deg) and 8 outputs (one for each harmonic overtone).

Sin:

To receive the desired results (a series of sine waves at the output) connect a sine oscillator here.

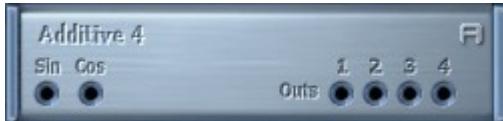
Cos:

To receive the desired results (a series of sine waves at the output) connect a 90deg phase shifted version of the sine oscillator (cosine) here. You may use a cosine oscillator.

Outs 1-8:

Each of the outputs represents a harmonic overtone derived from the input. If the inputs are fed with normalized sine and cosine waves, each output will generate a sine with a frequency multiplied by the output number (for instance, output 3 will generate a sine which is an octave and a fifth higher than the input – harmonic no. 3). however experimenting with other shapes than sine, and live audio is also beneficial!

Additive 4



Additive 4 is a wave-shaper that uses sine and cosine waveforms to generate the first 4 harmonics of the fundamental frequency. The harmonics are aligned at 0 degrees which makes it ideal for additive synthesis. It has two inputs (sine and cosine – sine at 90 deg) and 4 outputs

(one for each harmonic overtone).

Sin:

To receive the desired results (a series of sine waves at the output) connect a sine oscillator here.

Cos:

To receive the desired results (a series of sine waves at the output) connect a 90deg phase shifted version of the sine oscillator (cosine) here. You may use a cosine oscillator.

Outs 1-4:

Each of the outputs represents a harmonic overtone derived from the input. If the inputs are fed with normalized sine and cosine waves, each output will generate a sine with a frequency multiplied by the output number (for instance, output 3 will generate a sine which is an octave and a fifth higher than the input – harmonic no. 3). however experimenting with other shapes than sine, and live audio is also beneficial!

Osc Shapers

4 Sin Oct



oscillator.

Max. Polyphony: 16

In:

This is the audio input of this shaper. Connect a sine-shaped oscillator here in order to generate 4 upper octaves from it.

Out 1:

A sine wave, tuned 1 octave higher than the original sine oscillator, is generated here.

Out 2:

A sine wave, tuned 2 octaves higher than the original sine oscillator, is generated here.

Out 3:

A sine wave, tuned 3 octaves higher than the original sine oscillator, is generated here.

Out 4:

A sine wave, tuned 4 octaves higher than the original sine oscillator, is generated here.

4 Tri Oct



use more than one oscillator.

Max. Polyphony: 16

In:

This is the audio input of this shaper. Connect a triangle-shaped oscillator here in order to generate 4 upper octaves from it.

Out 1:

A triangle shape tuned 1 octave higher than the original triangle oscillator, is generated here.

Out 2:

A triangle shape tuned 2 octaves higher than the original triangle oscillator, is generated here.

Out 3:

A triangle shape tuned 3 octaves higher than the original triangle oscillator, is generated here.

Out 4:

A triangle shape tuned 4 octaves higher than the original triangle oscillator, is generated here.

This shaper, when fed with a sine shaped oscillator, generates 4 additional upper octaves at 4 separate outputs. This makes it easy to create impressive multi-layered sounds without the need to use more than one oscillator.

Freq Divider



known as a ‘flip-flop’). The result is a pulse shaped signal, tuned exactly one octave below the incoming signal. This shaper can be used to create the classic “sub-oscillator” effect, without actually using an additional oscillator.

Max. Polyphony: 14

In:

This is the audio input of this shaper, where you connect the audio signal to be processed (usually LFOs or oscillators).

Out:

This is the audio output of this shaper, where the processed signal emerges, transposed an octave lower in the form of a pulse wave.

Quad Shaper



This shaper divides the incoming signal into four amplitude sections, providing control over the amplitude and polarity of each section. Mixing between the four sections constructs new waveshapes, and the amplitude of each section can be modulated, enabling the creation of dynamically evolving sound textures.

Max. Polyphony: 2

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Faders 1-4:

These bipolar faders set the volume of their corresponding amplitude sections. At the centered position the sections will be muted. Dragging the faders up will result in positive amplification of the amplitude sections. Dragging them down will result in negative (inverted) amplification of the amplitude section. Varying the positions of the faders will create different oscillator shapes.

mod 1-4:

These are inputs for control signals (usually an Envelope or LFO). They are used to modulate the gain of the four stages. The Mod knobs adjust the amount of modulation applied to each fader, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Tri 2 Sin



This module shapes triangular waveforms into sine ones. The Amount knob controls the transition between triangle and sine wave-shapes. This module is useful for FM and Phase Modulation tasks.

Max. Polyphony: 16

In:

This is the audio input of this shaper. Connect a triangle oscillator here.

Amount:

This control sets the amount of shaping applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually shape the triangle into sine.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this Shaper, where the processed signal emerges.

NBL Phase Mod



This module shapes any FleXor NBL sawtooth oscillator into a phase shifted version of itself, in a dynamic manner (not just the phase start point). It is useful for creating massive detuned saw patches with only one NBL sawtooth oscillator. This module is

optimized for audio rate modulation, which makes it very useful for Phase Modulation (FM-like) operations.

Max. Polyphony: 16

In:

This is the audio input of this Shaper, where you connect the audio signal to be processed. This input is optimized for the output of the NBL Saw oscillator series.

Phase:

This function adjusts the phase of the incoming NBL oscillator . Turning the knob to the left will result in a phase of zero degrees, while turning it to the right will gradually shift the phase up to 360 degrees.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the phase of the incoming NBL oscillator. The Mod knobs adjust how much the Phase function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed sound emerges.

NBL Saw 2 PWM



wide pads, or an ensemble-like effect of several oscillators playing together. Max. Polyphony: 16

This module shapes any FleXor NBL sawtooth oscillator into a powerful-sounding pulse, with a variable Pulse-Width function to create classic Pulse waveforms from fat and symmetric to piercing and razor-thin. Modulate the Pulse-Width for evolving and

In:

This is the audio input of this shaper, where you connect the audio signal to be processed. This input is optimized for the output of the NBL Saw oscillator series.

PW:

This function (Pulse Width) adjusts the length relation between the positive and negative halves of a pulse waveshape. Turning the PW knob to the right will shorten the length of the positive half while lengthening the negative half accordingly, so the overall length of the complete pulse waveshape cycle remains unchanged. Turning the PW knob to the left will shorten the length of the negative half while lengthening the positive half accordingly.

PWM:

This is an input for a control signal (usually from an LFO or Envelope), which is used to modulate the Pulse-Width of the generated waveshape. The PWM knob adjusts the amount of Pulse Width Modulation, and is bipolar.

Out:

This is the audio output of this shaper, where the processed sound emerges.

NBL Saw 2 Tri-Sin



sound character of its own. Max. Polyphony: 16

This module shapes any FleXor NBL sawtooth oscillator into a triangular or near-perfect sine wave-shape, which makes it very useful for FM operations. The near-perfect sine wave-shape has a slight harmonic coloration (3rd harmonic at -50db), giving it a unique

In:

This is the audio input of this Shaper, where you connect the audio signal to be processed. This input is optimized for the output of the NBL Saw oscillator series.

Tri-Sin:

This function adjusts the waveform to which the input will be shaped. Turning the knob to the left will result in a triangular wave-shape, while turning it to the right will gradually shape it into a near-perfect sine wave.

mod:

This is an input for control signal (usually from an LFO or Envelope), which is used to modulate the Tri-Sin function. The Mod knob adjusts the degree of modulation.

Out:

This is the audio output of this shaper, where the processed sound emerges.

Saturation

Germanium Drive



produces a rich, warm and lush overdrive effect. Higher values of Drive yield a more drastic “near-square” saturation.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Drive:

This control sets the amount of overdrive (Saturation) applied to the incoming signal. Turning the knob to the right will gradually increase the Saturation effect.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Drive amount applied to the incoming signal. The Mod knobs adjust how much the Drive parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Soft Sat



This shaper gently saturates the incoming signal, boosting the perceived volume, and enhancing the overall presence of the tone, without clipping it. Its effect is mild and suitable for gentle tasks such as adding presence to vocals, solo instruments and drums,

and can even be used for mastering tasks.

Max. Polyphony: 16

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of Soft Sat applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Soft Sat effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Fat Sat



Max. Polyphony: 16

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of Fat Sat applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Fat Sat effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Warm Sat



Max. Polyphony: 16

This shaper gently saturates the incoming signal, boosting the perceived volume and warming up the tone without clipping it. Its effect is ideal for adding fatness and roundness to incoming signals such as vocals, drums and instruments.

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of Warm Sat applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Warm Sat effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Valve Drive



Max. Polyphony: 16

This shaper creates a mild tube-like overdrive effect. At lower values, a soft and rounded saturation occurs. Higher values will create deeper and fatter saturation, producing that rich and musical presence often associated with tube-based overdrive effects.

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of Valve Drive applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Valve Drive effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Tape Drive



“near-square” saturation, producing a rich, warm and lush overdrive effect.

Max. Polyphony: 15

This shaper creates mild to aggressive saturation effects. At lower values, an incoming triangle-shaped signal will be rounded off at its edges, creating a distinctive saturation of a hot recorded analogue tape machine. Higher values will create a far more drastic

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of Tape Drive applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Tape Drive effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Guitar Drive



This shaper clips sound according to its ballistic behaviour (speed and direction of movement). This makes it ideal for guitars as it interacts with the signal musically. It is excellent for connecting to real guitar amplifiers. Rolling-off some high frequencies with a

Low-Pass filter or EQ after this module can simulate a guitar speaker cabinet, obtaining a great direct-to-disk guitar solution. Obviously, this shaper also works well in all situations where a good-sounding distortion effect is required.

Max. Polyphony: 16

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of Guitar Drive applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Guitar Drive effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Hard Drive



Hard Drive is a shaper that creates aggressive overdrive effects. The saturation that occurs on the positive axis is asymmetric while the negative axis hard-clips. This module produces a hard, harsh and edgy overdrive effect.

Higher values of Drive yield a more drastic “near-square” saturation.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Drive:

This control sets the amount of overdrive (Saturation) applied to the incoming signal. Turning the knob to the right will gradually increase the Saturation effect.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Drive amount applied to the incoming signal. The Mod knobs adjust how much the Drive parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Hexa-Drive



Hexa-Drive is a shaper that creates mild to aggressive saturation effects. It uses linear interpolation to create polygon-like saturation, and it is possible to select the number of sides for the polygon. This module produces a

warm yet digital sounding overdrive effect. Higher values of Drive yield a more drastic “near-square” saturation.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Drive:

This control sets the amount of overdrive (Saturation) applied to the incoming signal. Turning the knob to the right will gradually increase the Saturation effect.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Drive amount applied to the incoming signal. The Mod knobs adjust how much the Drive parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Ripple Drive



Ripple Drive is a shaper that creates mild to aggressive saturation effects. It is designed to imitate low sample-rate digital devices and the ripple effect they create when being overdriven. You can control the amount of overdrive as well as the amount of ripple effect. This module produces a low-fi digital sounding overdrive effect. Higher values of Drive

yield a more drastic “near-square” saturation.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Drive:

This control sets the amount of overdrive (Saturation) applied to the incoming signal. Turning the knob to the right will gradually increase the Saturation effect.

Ripple:

This control sets the amount of ripple applied to the incoming signal. More ripple will result in more higher pitched overtones.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Drive amount applied to the incoming signal. The Mod knobs adjust how much the Drive parameter is modulated by the control signals, and are bipolar.

Rmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Ripple amount applied to the incoming signal. The Mod knobs adjust how much the Ripple parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Thick Drive



a thick, warm and lush overdrive effect. Higher values of Drive yield a more drastic “near-square” saturation.

In:

This is the audio input of this distortion module, where you connect the audio signal to be processed.

Drive:

This control sets the amount of overdrive (Saturation) applied to the incoming signal. Turning the knob to the right will gradually increase the Saturation effect.

Dmod(s):

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the Drive amount applied to the incoming signal. The Mod knobs adjust how much the Drive parameter is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Special

Crush



and raspy.

Max. Polyphony: 16

This shaper applies a bit-crushing (bit-reduction) effect. Amplitude changes of the incoming signal are quantized to bigger steps, downgrading the signal to lower bit depths. As a result, additional harmonics are generated, and the overall sound becomes more brittle

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of bit crushing effect applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the bit crushing effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator). They are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

DC Shift



modifies the incoming signal, transforming even the simplest waveshapes (such as sine or triangle) into bright and complex signals.

Max. Polyphony: 16

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of DC shift applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the DC shifting effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Flip n Clip



a wide spectrum of sound creation.

Max. Polyphony: 16

This shaper, instead of clipping, inverts the polarity of the peaks of the incoming signal, folds them back, and clips them to the zero-crossing point. This produces a special kind of distortion that generates a rich amount of additional harmonics, making it useful for

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of the Flip n Clip effect applied to the incoming signal. At its zero position (extreme left), the incoming signal is not affected at all. Turning the knob to the right will gradually increase the Flip n Clip effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Hard PWM



Max. Polyphony: 16

This shaper creates a Pulse Width Modulation effect, by offsetting the centered zero-crossing point and normalizing each side of the signal. The result of this process resembles Pulse Width Modulation when used on almost any waveshape except pulse.

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Amount:

This control sets the amount of the Hard PWM effect applied to the incoming signal. At its zero position (middle), the incoming signal is not affected at all. Turning the knob left or right will gradually increase the Hard PWM effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Amount function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

NBL WaveDraw



Monophonic.

This module shapes any FleXor NBL Saw-shaped oscillator into a new waveform drawn on-screen with the included virtual pencil. The virtual pencil can draw new waveshapes with the mouse, or it can be modulated to draw by its horizontal axis (input X) as well as vertical axis (input Y). WaveDraw has a 64 sample resolution, and it is Non-Band-Limited. The resulting sound has a raw and edgy digital character, reminiscent of the first generation of digital synthesizers. WaveDraw may also be used for wave reallocation to make special bit crushing effects.

In:

This is the audio input of this shaper, where you connect the audio signal to be processed. This input is optimized for the output of the NBL Saw oscillator series.

Virtual Pencil:

The virtual pencil allows free drawing of any waveshape with the mouse.

X:

This control sets the virtual pencil position anywhere along the horizontal axis.

X mod:

This is an input for a control signal (from the FleXor WaveDraw Assist module, or any LFO or Envelope), which is used to modulate the virtual pencil position anywhere along the horizontal axis. The X Mod knob adjusts the amount of modulation which will be applied to the pencil's position, and is bipolar.

Y:

This control sets the virtual pencil position anywhere along the vertical axis.

Y mod:

This is an input for a control signal (from the FleXor WaveDraw Assist module, or any LFO or Envelope), which is used to modulate the virtual pencil position anywhere along the vertical axis. The Y Mod knob adjusts the amount of modulation which will be applied to the pencil's position, and is bipolar.

Out:

This is the audio output of the shaper, where the processed sound emerges.

Gfx on/off:

Turns the graphic representation off to save CPU power

WaveDraw Assist



This module generates synchronized control signals, optimized especially for the FleXor NBL WaveDraw shaper. Combine these two modules, to transform an NBL saw oscillator in to almost infinite amount of complex and evolving waveshapes. Various

formants can be generated, as well as audio-rate modulation of waveshape drawing, which produces an aggressive sound character.

Monophonic.

Shape:

This control sets the waveshape of the control signal sends from output Y.

Harmonic:

This text-fader sets the frequency multiplication of the Y output in sync to the X output.

Formantor:

This control shapes the harmonic content to a formant-enhanced wave.

Hold:

Activating this switch will freeze the values of the control signals sent from outputs Y and X causing the "Virtual Pencil" on the connected NBL WaveDraw module to stop its drawing.

HiSpeed:

Activating this switch will cycle the X and Y control signals 25 times per second (closer to audio rate), allowing fast drawing of waves. With HiSpeed deactivated, the controls cycle every 4 seconds.

X:

The control signal for the X modulation input on the NBL WaveDraw module is generated here.

Y:

The control signal for the Y modulation input on the NBL WaveDraw module is generated here.

Curve



Curve is a shaper that creates an S shaped curve for bi-polar audio, or an exponential to logarithmic curve for uni-polar audio. At an extreme left setting the curve of the shaper is exponential, and on the extreme right a

logarithmic curve is applied. It is ideal for using with envelopes and control signals, or on oscillators before harmonic shapers.

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Curve:

This control sets the amount of the Curve effect applied to the incoming signal. At its zero position (middle), the incoming signal is not affected at all. Turning the knob left or right will gradually increase the Curve effect.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Curve function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

Soft PWM



Soft PWM is a shaper that creates a Pulse Width Modulation effect by applying a logarithmic to exponential curve. The result of this process resembles Pulse Width Modulation when used on almost any oscillator wave-

shape except pulse.

In:

This is the audio input of this shaper, where you connect the audio signal to be processed.

Curve:

This control sets the amount of the Soft PWM effect applied to the incoming signal. At its zero position (middle), the incoming signal is not affected at all. Turning the knob left or right will gradually increase the Soft PWM effect.

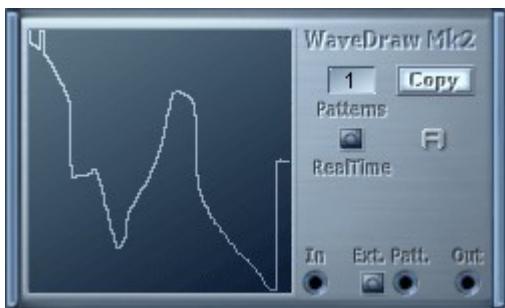
mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the shaping amount applied to the incoming signal. The Mod knobs adjust how much the Soft PWM function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this shaper, where the processed signal emerges.

WaveDraw MK2



WaveDraw MK2 shapes any audio signal using the wave-shapes you draw on the module's screen. WaveDraw Mk2 has a 128 sample resolution and it uses linear-interpolation. This makes it possible to use it as a wave-shaper on any audio source without any bit crushing side-effects. Save the different wave-shapes you create with the module as 128 patterns. WaveDraw Mk2 can also be used for modifying control signals such as LFOs and envelopes.

In:

This is the audio input of this shaper, where you connect the audio signal to be processed. This input is optimized for the output of the NBL Saw oscillator series.

Graphics Square:

Allows free drawing of any wave-shape with the mouse. Just click and drag.

RealTime:

Realtime comes to help draw the wave more efficiently, since the drawing action takes a lot of resources from the pc, turning it off just before drawing will make the process offline, and faster. Turning realtime on again will apply the waveform to the shaper.

Patterns:

The pattern of wave now presented.

Copy/Paste:

When the button is on copy state, pressing it, will copy the pattern to the clipboard. Once pressed it will change to “Paste” which will enable you to browse the patterns. Once you found the number of pattern you want to use, click paste, and the pattern will be duplicated to the new pattern slot.

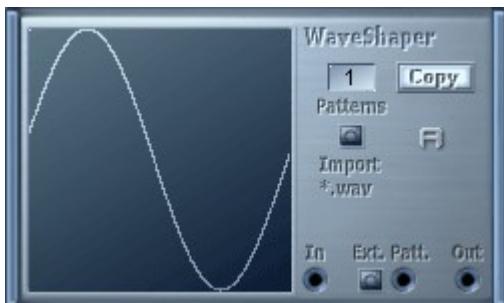
Ext. Patt:

When on, the input next to the button, allows a ramp signal to switch between the 128 patterns.

Out:

This is the audio output of the shaper, where the processed sound emerges.

WaveShaper



WaveShaper shapes any audio signal using a wave-shape based on an imported wave file. Import wave files into the dedicated panel and save the different shapes with the module as patterns. Wave-Shaper has a 128 sample resolution and it uses linear-interpolation. This makes it possible to use it as a wave-shaper on any audio source without any bit crushing side-effects. Wave-Shaper can also be used for modifying control signals such as LFOs and envelopes.

In:

This is the audio input of this shaper, where you connect the audio signal to be processed. This input is optimized for the output of the NBL Saw oscillator series.

Wave View:

Here you can view the waveform attached to the shaper.

Import Wave:

Opens a wave editor where you can load a wave file, and cut it for the shaper.

Patterns:

The pattern of wave now present.

Copy/Paste:

When the button is on copy state, pressing it, will copy the pattern to the clipboard. Once pressed it will change to “Paste” which will enable you to browse the patterns. Once you found the number of pattern you want to use, click paste, and the pattern will be duplicated to the new pattern slot.

Ext. Patt:

When on, the input next to the button, allows a ramp signal to switch between the 128 patterns.

Out:

This is the audio output of the shaper, where the processed sound emerges.

NBL 4mantor I



This shaper shapes any FleXor NBL saw-shaped oscillator into a saw with a single strong formant-emphasis. The sound resembles a very narrow and resonant band-pass filter, following the pitch of the oscillator. Fixed formants are possible by connecting

the note output of the MVC to the key follow input of NBL 4mantor and setting the key follow knob fully to the left. This shaper is also effective on other wave-shapes although the results may vary. It is excellent for making punchy bass sounds as well as filter-like sweeps.

Max. Polyphony: 7

In:

This is the audio input of this module, where you connect oscillators (FleXor NBL oscillators are recommended).

Formant:

This control sets the frequency of the formant. At its zero position (extreme left), no sound is heard at all (the formant is much lower than the fundamental). Turning the knob clockwise will gradually sweep the formant frequency upwards.

mod:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the formant frequency. The Mod knob adjusts the amount of formant frequency modulation, and is bipolar.

key f:

This knob sets the key follow amount and direction of the formant, and is bipolar. Setting it to the fully negative position creates fixed formants.

Out:

This is the audio output of this module, where the processed signal emerges.

NBL 4mantor II



This shaper shapes any FleXor NBL saw-shaped oscillator into a saw with two strong formant-emphasis frequencies. The sound resembles two very narrow and resonant band-pass filters, following the pitch of the oscillator. Fixed formants are possible by connecting the note output of the MVC to the note input of 4mantor and setting the key follow knobs to the extreme left. By modulating both of the formants at once, vocal-like vowels can be generated. The shift function will

modulate the formants in parallel. This shaper is also effective on other wave-shapes although the results may vary.

Max. Polyphony: 3

In:

This is the audio input of this module, where you connect oscillators (FleXor NBL oscillators are recommended).

Form 1-2:

These controls set the frequency of the formants. At their zero position (extreme left), no sound is heard at all (the formant is much lower than the fundamental). Turning the knobs clockwise will gradually sweep the formant frequencies upwards.

m 1-2:

These are inputs for control signals (usually an Envelope or LFO), which are used to modulate the formant frequencies. The Mod knobs adjust the amount of formant frequency modulation, and are bipolar.

Shift:

This function sets the global frequency shifting for all of the formants.

Sm:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the shifting function. The Ms knob adjusts the amount of formant shift modulation derived from the input, and is bipolar.

note in:

This is an input for the Note signal from an MVC module. It is necessary in order to track the formant frequency values with MIDI notes, according to the position of the Key F knobs. Turning the Key F knobs to the fully negative position will result in fixed formants.

key f 1-2:

These knobs set the key follow amount and direction of the formants, and are bipolar. Setting them to the fully negative position creates fixed formants.

Out:

This is the audio output of this module, where the processed signal emerges.

NBL 4mantor III



This shaper shapes any FleXor NBL saw-shaped oscillator into a saw with three strong formant-emphasis frequencies. The sound resembles three very narrow and resonant band-pass filters, following the pitch of the oscillator. Fixed formants are possible by connecting the note output of the MVC to the note input of 4mantor and setting the key follow knobs fully to the left. By modulating all of the formants at once, vocal-like vowels can be generated. The shift function will modulate the formants in parallel. This shaper is also effective on other wave-shapes although the results

may vary.

Max. Polyphony: 2

In:

This is the audio input of this module, where you connect oscillators (FleXor NBL oscillators are recommended).

Form 1-3:

These controls set the frequency of the formants. At their zero position (extreme left), no sound is heard at all (the formant is much lower than the fundamental). Turning the knobs clockwise will gradually sweep the formant frequencies upwards.

m 1- 3:

These are inputs for control signals (usually an Envelope or LFO), which are used to modulate the formant frequencies. The Mod knobs adjust the amount of formant frequency modulation, and are bipolar.

Shift:

This function sets the global frequency shifting for all of the formants.

Sm:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the shifting function. The Ms knob adjusts the amount of formant shift modulation derived from the input, and is bipolar.

Note In:

This is an input for the Note signal from an MVC module. It is necessary in order to track the formant frequency values with MIDI notes, according to the position of the Key F knobs. Turning the Key F knobs to the fully negative position will result in fixed formants.

key f 1-3:

These knobs set the key follow amount and direction of the formants, and are bipolar. Setting them to the fully negative position creates fixed formants.

Out:

This is the audio output of this module, where the processed signal emerges.

NBL 4mantor IV



This shaper shapes any FleXor NBL saw-shaped oscillator to a saw with four strong formant-emphasis frequencies. The sound resembles four very narrow and resonant band-pass filters, following the pitch of the oscillator. Fixed formants are possible by connecting the note output of the MVC into the note input of 4mantor and setting the key follow knobs to negative. By modulating all of the formants at once, vocal like vowels can be generated. This shaper is also effective on other wave-shapes yet the results may vary. This shift function will modulate the formants in a parallel

manner.

Monophonic.

In:

This is the audio input of this module, where you connect oscillators (FleXor NBL oscillators are recommended).

Form 1-4:

These controls set the frequency of the formants. At their zero position (extreme left), no sound is heard at all (the formant is much lower than the fundamental). Turning the knobs clockwise will gradually sweep the formant frequencies upwards.

m 1-4:

These are inputs for control signals (usually an Envelope or LFO), which are used to modulate the formant frequencies. The Mod knobs adjust the amount of formant frequency modulation, and are bipolar.

Shift:

This function sets the global frequency shifting for all of the formants.

Sm:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the shifting function. The Ms knob adjusts the amount of formant shift modulation derived from the input, and is bipolar.

Note In:

This is an input for the Note signal from an MVC module. It is necessary in order to track the formant frequency values with MIDI notes, according to the position of the Key F knobs. Turning the Key F knobs to the fully negative position will result in fixed formants.

key f 1-4:

These knobs set the key follow amount and direction of the formants, and are bipolar. Setting them to the fully negative position creates fixed formants.

Out:

This is the audio output of this module, where the processed signal emerges.

Sequencers

Sequencers are pattern generators. FleXor's sequencers generate patterns of 16 steps, and come in three versions:

Note Sequencer:

Generates a frequency output to control oscillator frequencies (i.e. for generating melodies).

Control Sequencer:

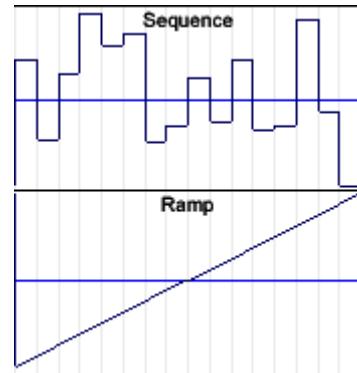
Generates bipolar control signals to modulate parameters such as filter cutoff, envelope stages, etc.

Gate Sequencer:

Generates gate messages to retrigger envelopes, oscillators and LFOs.

Using FleXor's Ramp module in conjunction with the sequencers, enables consecutive playing of the steps with sample-accurate timing and synchronization. Using other oscillators will result in different playing orders. When playing at high speeds, they can also be used as wave-table oscillators.

All of FleXor's sequencers have pattern management that allows you to save an infinite number of patterns and switch between 128 patterns per bank. In order to select patterns from the sequencer's modules, it is necessary to index the required bank to 0 and auto-index all the presets within that bank. The pattern management comes in the form of presets, allowing you to save, copy and paste patterns in a comfortable manner. It is also possible to switch patterns with an external source like the Ramp module.



FleXor's Ramp module may be used to synchronize any number of sequencer modules (including modules from the Granular section). It can be controlled by BPM, frequency, or an external frequency source. It can also be used as an LFO when combined with FleXor's array of Shapers.

The PolyRhythmer module allows the creation of polyrhythmic patterns, by limiting the number of sequencer steps, while not changing the time needed to progress through them.

Algorithmic composition can be achieved by modulating the signal, by controlling sequencers with other sequencers, or by using logic gates.

All FleXor sequencer modules are now equipped with Swing function in v1.5, which shifts specific sequencer steps forward or backward in time, to create various degrees of those classic up or down beat shuffling grooves

The intuitive interfaces of FleXor's sequencers make it easy to create complete compositions within minutes, totally within the modular environment.

Ramp Tools

Gate 2 Ramp



Gate 2 Ramp converts gate messages into a stepped Ramp signal with a specified number of steps. Each time a gate on message is received the ramp will progress to the next step, until the maximum is reached. It is then reset to the first step and starts over from there. It also is possible to reset the ramp using another gate signal input to Ext Reset or by using the reset button. This module is ideal for using an external midi source to drive control and pitch sequencers.

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. Each time a gate signal is received the ramp will progress in one step.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional SonicCore envelope with Esync in your patch, or use FleXor's Esync module.

Reset:

When pressed the ramp re-sets to its starting position(step1).

Ext. Reset:

This input accepts gate messages. When a gate message is received, the ramp is reset to its starting position (step 0). This input is only active when the button next to it is set to On.

Steps:

This parameter adjusts the number of steps to take per ramp cycle. Drag up and down to change the number of steps per ramp cycle.

Indicator:

Here you can see the the ramp's current step state. This is not a text fader.

Out:

This is the output of the module. The generated ramp signal emerges here, which can be used to run FleXor's sequencers and granular processors, or it can also act as a very accurate LFO when combined with the Shapers.

Ramp Distributor 2



Ramp Distributor 2 divides one ramp signal into two ramps and distributes them consecutively. It can be used expand the number of steps in sequencers, or just to switch between ramp driven modules.

In:

This is the input of the module, where the ramp signal should be connected for division.

Outs:

These outputs are where the ramp signals emerge consecutively. The first ramp is generated in the 1st output, the second ramp in the 2nd output and so on....

Ramp Distributor 4



Ramp Distributor 4 divides one ramp signal into four ramps and distributes them consecutively. It can be used expand the number of steps in sequencers, or just to switch between ramp driven modules.

In:

This is the audio input of this module, where you connect the ramp to be divided. This input is optimized for the output of the NBL Saw oscillator series.

Outs:

These outputs are where the ramp signals emerge consecutively. The first ramp is generated in the 1st output, the second ramp in the 2nd output and so on....

Ramp Ranger



Ramp Ranger allows an incoming ramp signal to be scaled to a specified range of steps. The start and end position of the ramp can be set, and the number of steps in the ramp can be set to match the device being controlled so that the start and end values are correct. This module is useful for making polyrhythmic patterns as well as controlling pattern selection.

Ramp:

This is the audio input of this module, where you connect the ramp (saw-up oscillator) to be ranged. This input is optimized for the output of the NBL Saw oscillator series.

Start:

This parameter sets the starting step of the ramp.

End:

This parameter sets the ending step of the ramp.

Steps:

The parameter shows the number of steps left in a ranged ramp cycle. This is not a text fader.

Max:

This parameter shows the number of overall steps for a complete un-ranged ramp.

Out:

This is where the ranged Ramp signal emerges.

Ramp MK2



Ramp MK2 generates an accurate Ramp (saw-up) signal which is similar to the FleXor Aliasaw. Its output is typically connected to the Ramp input of a FleXor Sequencer, Switch, or granular device to dictate playing that module's steps in sequential order. Use Gate signals from an MVC to re-trigger Ramp Mk2, and use the Phase

knob to set the point the ramp signal is re-triggered from. The frequency of the generated ramp signal can be specified in Hz or in BPM..note: Ramp MK2 can also be used as an LFO when combined with FleXor's NBL Shapers.

BPM:

This parameter sets the Time of the LFO in BPM.

Div:

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Hertz:

Shows the time of the LFO in Hertz (Cycles per second).

SR:

In order for this module to work correctly, please tune this controller to your hardwares Sample-Rate state (as listed under the "Samplerate" menu in scope).

Freq:

This parameter adjusts the frequency (speed/rate) of the LFO.

Fmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the Frequency. The Fmod knob adjusts the amount of modulation which will be applied to the Frequency, and is bipolar.

Pmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the Phase. The Pmod knob adjusts the amount of modulation which will be applied to the Phase, and is bipolar.

Phase:

This parameter adjusts the phase of the LFO. 180 deg shifts the oscillator by half a cycle, while 360 shifts it by a whole cycle.

Out:

This is the output of the module. The generated ramp signal emerges here, which can be used to run FleXor's sequencers and granular processors, or it can also act as a very accurate LFO when combined with the Shapers.

Ramp



This module generates an accurate ramp (saw-up) signal, which is very similar to the FleXor NBL Saw Gate oscillator. Its output is usually connected to the Ramp input of a FleXor sequencer module, dictating consecutive playing order of the sequencer steps. Retriggering the Ramp module is done by sending gate signals from an MVC module. It will retrigger from the phase point set by the Start Phase knob. The frequency of the generated ramp signal can be set by Hz, by BPM (Beats-Per-Minute) with clock division, or by an external frequency source. Only one of these modes can be used at the same time: when any one is selected, the functions of the other two modes are disabled. The Ramp module can also be used as an LFO when combined with FleXor's NBL Shapers.

Max. Polyphony: 12

Mode:

Click and drag up/down to determine whether the speed of the generated ramp signal will be set by the internal Frequency knob, the internal BPM controls or by an external frequency generator connected to the Ext Freq input. When any one of these three modes is selected, the functions of the other two modes are disabled.

Frequency:

This function is activated only when the Mode is set to Knob, enabling the Frequency knob to control the speed of the generated ramp signal, in Hz.

BPM:

This function is activated only when the Mode is set to BPM, allowing the speed of the generated ramp signal to be set by the BPM value selected by clicking and dragging up/down on this text-fader.

Div:

This function is activated only when the Mode is set to BPM. Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 1, 1/2, 1/4, 1/8, 1/16 and so on. Choosing a Div value of 1 will play the pattern on a connected sequencer at the BPM indicated by the BPM control, while choosing the beat value 1/2 will play the pattern twice as fast.

Start Phase:

This function sets the point from which the generated ramp cycle starts its playback, every time a gate signal is received. At the centered position (0 degrees), The ramp will start its cycle from the zero crossing. In order to trigger the sequencers from their first step, Start Phase should be set to the extreme left.

Ext. Freq:

This input is activated only when the Mode is set to Ext Freq, enabling a connected external frequency generator to control the speed of the generated ramp signal.

Gate:

This is an input for gate signals, usually from a Gate output of an MVC module. Each time a signal is received at the gate input, the generated ramp is retrigged from the phase position indicated by Start Phase.

FM:

This is an input for control signals (usually from another Oscillator or LFO), which are used to modulate the frequency of the Ramp. The FM knob adjusts the degree to which the control input modulates the Ramp frequency.

Out:

This is the output of the module. The generated ramp signal emerges here, which can be used to run FleXor's sequencers and granular processors, or it can also act as a very accurate LFO when combined with the Shapers.

Poly Rhythmer



This module sets the number of steps which will be played on any of the FleXor sequencer modules. It receives the frequency output from a ramp generator, and sends it to the Ramp input of any FleXor sequencer module. The sequencer can then be limited to play any

range of its steps, by the ability to set the start and end of the sequence. The Poly Rhythmer adjusts the tempo of the sequencer, so it will always take the same time to complete the playback of a pattern, regardless of how many steps it is constructed of. This module is ideal for creation of polyrhythmic grooves and algorithmic music composition.

Max. Polyphony: 16

In:

This is the input of the module. It should usually be connected to the output of the FleXor Ramp module.

Start:

This function sets the step from which the pattern will start its playback on the connected sequencer. Every time the pattern reaches the last step (indicated by the End value), it will restart its sequence from the step indicated by the Start value. Click and drag this text-fader up/down to set the Start step to a value from 1 to 16.

End:

This function sets the step at which the pattern will end its playback on the connected sequencer. Every time the pattern reaches the step number indicated by the End value, it will restart its sequence from the step number indicated by the Start value. Click and drag this text-fader up/down to set the End step to a value from 1 to 16.

Steps:

This display shows how many steps will be played on a connected sequencer, according to the difference between the step numbers indicated at the Start and End windows. It is not directly adjustable, it is merely an indicator based on the Start and End values.

Out:

This is the output of this module. It should be connected to the Ramp input of any FleXor sequencer module.

Seq Joiner 32



them together.

Ramp:

This is the audio input of this module, where you connect the ramp (saw-up oscillator) to be ranged. This input is optimized for the output of the NBL Saw oscillator series. Here you connect the master Ramp signal.

Ins:

This is an input for the sequencers outputs. Connect your 16 step sequencers consecutively from left to right. The first sequencer to in 1, the second into in 2 and so on.

Ramp Outs:

Here is the split ramp control. This is what you connect to the two 16 step sequencers consecutively.

Out:

This is the output of this module, here emerges a 32step sequence combined from the two sequencers connected to it.

Seq Joiner 64



also switches between the outputs of the individual sequencers to make it easy to join them together.

Ramp:

This is the audio input of this module, where you connect the ramp (saw-up oscillator) to be ranged. This input is optimized for the output of the NBL Saw oscillator series. Here you connect the master Ramp signal.

Ins:

This is an input for the sequencers outputs. Connect your 16 step sequencers consecutively from left to right. The first sequencer to in 1, the second into in 2 and so on.

Ramp Outs:

Here is the split ramp control. This is what you connect to the two 16 step sequencers consecutively.

Out:

This is the output of this module, here emerges a 64step sequence combined from the four sequencers connected to it.

Ramp Divider



external pattern selections.

Max. Polyphony: 5

This module divides a ramp cycle into several ramps, thus multiplying its frequency. It is useful for synchronizing several sequencers that operate at different tempo divisions, enabling the creation of complex polyrhythmic patterns, or for controlling

In:

This is the audio input of this module, where you connect the ramp (saw-up oscillator) to be divided. This input is optimized for the output of the NBL Saw oscillator series.

Division:

This control sets the amount of division effect applied to the incoming ramp signal. At its zero position (extreme left), the incoming signal is not divided at all. Turning the knob to the right will increase division amount of each incoming ramp cycle into shorter ramp cycles.

Out:

This is the audio output of this module, where the divided ramp emerges.

Ramp State Indicator



This module displays a series of sixteen LEDs that indicate the position of sequencers that are being driven by the Ramp module.

Max. Polyphony: 5

In:

This is the input of this module, where the output of a ramp generator (usually the FleXor Ramp module) should be connected.

LED 1 through 16 :

These LEDs indicate the step position of any 16-steps sequencer module, which is driven in parallel with the same ramp signal connected to this module.

Sequencers

Hyper Control Sequencer Mk2



Hyper Control Sequencer Mk2 is a sequencer module that outputs a 16 step sequence of control values that are determined by each step's fader. To play the 16 steps in sequential order connect this module's Ramp input to a FleXor Ramp module's output. More complex playing orders can be created by using other control sources like LFOs and step sequencers. Up to 128 different patterns can be defined and stored with each instance of the

module used inside a patch. This module is ideal for sending sequenced control values to any parameter that can be modulated--like envelope decay, filter cutoff and so on.

Unipolar:

This switch shifts the control signal output range of the Hyper Control Seq from bipolar to unipolar.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO wave-shape cycle.

Faders 1-16:

These faders set the value of the control signal which will be generated at each step. The control range of the faders is bipolar, so at the centered zero position no signal will be generated. Setting the faders up or down enables the generation of positive and negative control signal values. This controller allows freely drawing the position of the faders in one mouse movement.

Pattern:

The pattern of sequence now presented.

Ext. Patt:

When on, the input next to the button, allows a ramp signal to switch between the 128 patterns.

Copy/Paste:

When the button is on copy state, pressing it, will copy the pattern to the clipboard. Once pressed it will change to "Paste" which will enable you to browse the patterns. Once you found the number of pattern you want to use, click paste, and the pattern will be duplicated to the new pattern slot.

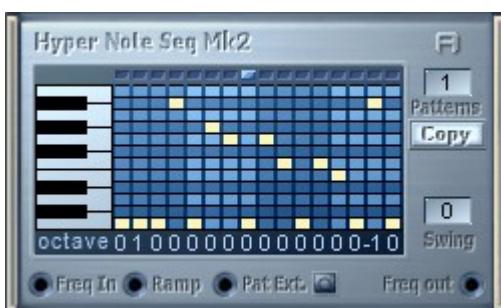
Swing:

This function sets the swing amount applied to the sequencer pattern. Any desired value between 100 to -100 can be entered at the swing display, either by typing or by clicking and dragging the mouse up / down.

Out:

This is the output of the sequencer, from which the sequence of control signal values emerges. This output can be connected into any modulatable function, such as VCA, filter cutoff and so on.

Hyper Note Sequencer Mk2



Hyper Note Sequencer Mk2 is a sequencer module that outputs a 16 step sequence of note values that are based off of the frequency input then transposed as specified for each step. To play the 16 steps in sequential order connect this module's Ramp input to a FleXor Ramp module's output. More complex playing orders can be created by using other control sources like LFOs and step sequencers. Up to 128 different patterns can be defined and stored with each instance of the module used inside a patch. This module is ideal for generating melodic patterns

to be used for algorithmic music composition with modules such as oscillators, resonators and so on.

Freq in:

This is an input for an external frequency signal, used to transpose the sequenced pattern up or down. Usually the Freq output from an MVC module will be used, so that MIDI notes will control the transposition of the pattern. This input must be connected in order for the module to work.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO wave-shape cycle.

Piano Roll Display:

Here you can compose a 16 step melody. The vertical columns represent the pitch (according to the keyboard image), while the horizontal rows represent the placement in time, according to each step. Click and drag to arrange the notes. This controller allows freely drawing the position of notes in one mouse movement.

Octave:

This function sets the transposition of each note by whole octaves. Setting the desired value is made by clicking and dragging the octave number indicator beneath each note.

Pattern:

The pattern of sequence now presented.

Ext. Patt:

When on, the input next to the button, allows a ramp signal to switch between the 128 patterns.

Copy/Paste:

When the button is on copy state, pressing it, will copy the pattern to the clipboard. Once pressed it will change to "Paste" which will enable you to browse the patterns. Once you found the number of pattern you want to use, click paste, and the pattern will be duplicated to the new pattern slot.

Swing:

This function sets the swing amount applied to the sequencer pattern. Any desired value between 100 to -100 can be entered at the swing display, either by typing or by clicking and dragging the mouse up / down.

Freq out:

This is the frequency output of the sequencer, where a sequence of frequency values is generated. This output should be connected into the frequency input of oscillators, to generate melodies.

Hyper Gate Sequencer Mk2



Hyper Gate Sequencer Mk2 is a sequencer module that outputs a 16 step sequence of Gate messages. To play the 16 steps in sequential order connect this module's Ramp input to a FleXor Ramp module's output. More complex playing orders can be created by using other control sources like LFOs and step sequencers. Up to 128 different patterns can be defined and stored with each instance of the module used inside a patch. This module is ideal for triggering envelopes, LFOs, oscillators and so on.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO waveshape cycle.

Switches 1-16:

These are the on/off switches for each one of the 16 steps. In the on state the step will generate on and off gate messages, while at the off state it will not generate any gate messages.

Length:

This function globally sets the length of the steps. At its zero position (extreme left), the steps are not active at all, while higher values will make the steps longer. At its maximum position (extreme right), neighboring activated steps will be tied together to become one longer step.

Lmod:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the Length of the sequencer steps. The Lmod knob adjusts the amount of modulation, and is bipolar.

Pattern:

The pattern of sequence now presented.

Ext. Patt:

When on, the input next to the button, allows a ramp signal to switch between the 128 patterns.

Copy/Paste:

When the button is on copy state, pressing it, will copy the pattern to the clipboard. Once pressed it will change to "Paste" which will enable you to browse the patterns. Once you found the number of pattern you want to use, click paste, and the pattern will be duplicated to the new pattern slot.

Swing:

This function sets the swing amount applied to the sequencer pattern. Any desired value between 100 to -100 can be entered at the swing display, either by typing or by clicking and dragging the mouse up / down.

Out:

This is the output of the sequencer, from which the sequence of control signal values emerges. It can be connected to any gate-triggered devices such as LFOs, envelopes, oscillators and so on.

Interpolated Control Sequencer



Interpolated Control Sequencer is a sequencer module that is based off of 16 steps which are then interpolated linearly so that the output sequence of control values has smooth transitions between the steps. The module also interpolates between the first and last steps so that the transition back to the beginning of the sequence is seamless. To play the 16 steps in sequential order connect this module's Ramp input to a FleXor Ramp module's

output. More complex playing orders can be created by using other control sources like LFOs and step sequencers. Up to 128 different patterns can be defined and stored with each instance of the module used inside a patch. This module is ideal for sending smoothly sequenced control values to any parameter that can be modulated--like envelope decay, filter cutoff and so on. It can also be used as a wave-shaper by connecting an audio signal to the Ramp input (rather than a Ramp signal).

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO wave-shape cycle.

Faders 1-16 (Graphic display):

These faders set the value of the control signal which will be generated at each step. The control range of the faders is bipolar, so at the centered zero position no signal will be generated. Setting the faders up or down enables the generation of positive and negative control signal values. This controller allows freely drawing the position of the faders in one mouse movement.

Pattern:

The pattern of sequence now presented.

Ext. Patt:

When on, the input next to the button, allows a ramp signal to switch between the 128 patterns.

Copy/Paste:

When the button is on copy state, pressing it, will copy the pattern to the clipboard. Once pressed it will change to "Paste" which will enable you to browse the patterns. Once you found the number of pattern you want to use, click paste, and the pattern will be duplicated to the new pattern slot.

Swing:

This function sets the swing amount applied to the sequencer pattern. Any desired value between 100 to -100 can be entered at the swing display, either by typing or by clicking and dragging the mouse up / down.

Out:

This is the output of the sequencer, from which the sequence of control signal values emerges. This output can be connected into any modulatable function, such as VCA, filter cutoff and so on.

Hyper Control Seq



This is a control sequencer module with 16 steps, each of which sends out its own control value. It is ideal for sending sequenced control values to any modulatable function, such as envelope decay, filter cutoff, and so on. Consecutive playing order is achieved with the FleXor Ramp module connected to the Ramp input. Different playing orders can be achieved with other control inputs, such as LFOs and step sequencers. Up to 128 different patterns can be stored in each bank of presets, and scrolling through the patterns is possible by clicking and dragging the

pattern select text-fader, or with a ramp generator or any other modulation source through the external input.

Monophonic.

Unipolar:

This switch shifts the control signal output range of the Hyper Control Seq from bipolar to unipolar.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO waveshape cycle.

Faders 1-16:

These faders set the value of the control signal which will be generated at each step. The control range of the faders is bipolar, so at the centered zero position no signal will be generated. Setting the faders up or down enables the generation of positive and negative control signal values.

Draw:

Enabling the Draw function will activate the Virtual Pencil. This tool allows freely drawing the position of the faders in one mouse movement.

Manage Patterns:

This switch will open the preset window, allowing saving and loading of pattern presets. The preset bank should be indexed as bank 0 with the presets inside auto-indexed (right-click on the preset numbers) in order for this function to work.

Pattern Select:

Click and drag up/down to scroll thru the preset patterns. The preset bank should be indexed as bank 0 with the presets inside auto-indexed (right click on the preset numbers) in order for this function to work.

Ext. (pattern select):

This is an input for a control signal, used for scrolling through the preset patterns in the bank selected with the Manage Patterns function. A ramp generator such as saw-up LFO (the FleXor Ramp module is the most recommended) will make the Hyper Control Seq to scroll thru its preset

patterns one-after-another as numbered. Other LFO shapes will scroll through the preset patterns in various orders, according to the amplitude changes in the LFO waveshape cycle.

No. of patterns:

Click and drag up/down to set the number of patterns to be scrolled either by control signal modulating the Ext input, or by clicking and dragging at the Pattern Select text-fader.

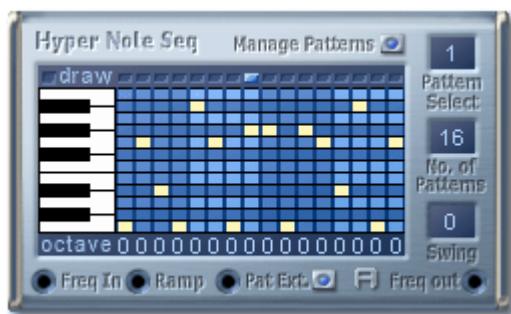
Swing:

This function sets the swing amount applied to the sequencer pattern. Any desired value between 100 to -100 can be entered at the swing display, either by typing or by clicking and dragging the mouse up / down.

Out:

This is the output of the sequencer, from which the sequence of control signal values emerges. This output can be connected into any modulatable function, such as VCA, filter cutoff and so on.

Hyper Note Seq



This is a note sequencer module with 16 steps, each of which sends out its own frequency value in parallel to the frequency input. It is ideal for algorithmic music composition, generating melodic patterns with modules such as oscillators, resonators and so on. Consecutive playing order is achieved with the FleXor's Ramp module connected to the Ramp input. Different playing orders can be achieved with other control signals, such as LFOs and step sequencers. Up to 128 different patterns can be stored in each bank of presets, and scrolling through the patterns is possible by

clicking and dragging the Pattern Select text-fader, or with a ramp generator or any other modulation source through the external input.

Monophonic.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO waveshape cycle.

Freq in:

This is an input for an external frequency signal, used to transpose the sequenced pattern up or down. Usually the Freq output from an MVC module will be used, so that MIDI notes will control the transposition of the pattern. This input must be connected in order for the module to work.

Piano Roll Display:

Here you can compose a 16 step melody. The vertical columns represent the pitch (according to the keyboard image), while the horizontal rows represent the placement in time, according to each step. Click and drag to arrange the notes.

Octave:

This function sets the transposition of each note by whole octaves. Setting the desired value is made by clicking and dragging the octave number indicator beneath each note.

Draw:

Enabling the Draw switch will activate the Virtual Pencil. This tool allows freely drawing the position of notes in one mouse movement.

Manage Patterns:

This switch will open the preset window, allowing saving and loading of pattern presets. The preset bank should be indexed as bank 0 with the presets inside auto-indexed (right-click on the preset numbers) in order for this function to work.

Pattern select:

Click and drag up/down to scroll thru the preset patterns. The preset bank should be indexed as bank 0 with the presets inside auto-indexed (right click on the preset numbers) in order for this function to work.

Ext. (pattern select):

This is an input for a control signal, used for scrolling through the preset patterns in the bank selected with the Manage Patterns function. A ramp generator such as saw-up LFO (the FleXor Ramp module is the most recommended) will make the Hyper Control Seq to scroll thru its preset patterns one-after-another as numbered. Other LFO shapes will scroll through the preset patterns in various orders, according to the amplitude changes in the LFO waveshape cycle.

No. of patterns:

Click and drag up/down to set the number of patterns to be scrolled either by control signal modulating the Ext input, or by clicking and dragging at the Pattern Select text-fader.

Swing:

This function sets the swing amount applied to the sequencer pattern. Any desired value between 100 to -100 can be entered at the swing display, either by typing or by clicking and dragging the mouse up / down.

Freq out:

This is the frequency output of the sequencer, where a sequence of frequency values is generated. This output should be connected into the frequency input of oscillators, to generate melodies.

Hyper Gate Seq



This is a gate sequencer module with 16 steps, each of which sends out its own gate value. It is ideal for triggering envelopes, LFOs, oscillators and so on. Consecutive playing order is achieved with the FleXor's Ramp module connected to the Ramp input. Different playing orders can be achieved with other control signals, such as LFOs and step sequencers. Up to 128 different patterns can be stored in each bank of presets, and scrolling through the patterns is possible by

clicking and dragging the pattern select text-fader, or with a ramp generator or any other modulation source through the external input.

Monophonic.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO waveshape cycle.

Switches 1-16:

These are the on/off switches for each one of the 16 steps. In the on state the step will generate on and off gate messages, while at the off state it will not generate any gate messages.

Length:

This function globally sets the length of the steps. At its zero position (extreme left), the steps are not active at all, while higher values will make the steps longer. At its maximum position (extreme right), neighboring activated steps will be tied together to become one longer step.

Lmod:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the Length of the sequencer steps. The Lmod knob adjusts the amount of Length modulation, and is bipolar.

Manage Patterns:

This switch will open the preset window, allowing saving and loading of pattern presets. The preset bank should be indexed as bank 0 with the presets inside auto-indexed (right-click on the preset numbers) in order for this function to work.

Pattern select:

Click and drag up/down to scroll thru the preset patterns. The preset bank should be indexed as bank 0 with the presets inside auto-indexed (right click on the preset numbers) in order for this function to work.

Ext. (pattern select):

This is an input for a control signal, used for scrolling through the preset patterns in the bank selected with the Manage Patterns function. A ramp generator such as saw-up LFO (the FleXor Ramp module is the most recommended) will make the Hyper Control Seq to scroll thru its preset patterns one-after-another as numbered. Other LFO shapes will scroll through the preset patterns in various orders, according to the amplitude changes in the LFO waveshape cycle.

No. of patterns:

Click and drag up/down to set the number of patterns to be scrolled either by control signal modulating the Ext input, or by clicking and dragging at the Pattern Select text-fader.

Swing:

This function sets the swing amount applied to the sequencer pattern. Any desired value between 100 to -100 can be entered at the swing display, either by typing or by clicking and dragging the mouse up / down.

Out:

This is the output of the sequencer, from which the sequence of control signal values emerges. It can be connected to any gate-triggered devices such as LFOs, envelopes, oscillators and so on.

Tools

Besides the important sound generators modulators and processors, FleXor offers a wide array of useful and time-saving tools to enhance and simplify your modular experience.

Control

Control Smoother



This module smoothes out any control signal fed to it. It is ideal for smoothing steps from low-resolution midi controllers, or for the creation of glides between notes generated by note/control sequencer, and it can also round the edges of a square or saw shaped LFOs.

This module is optimized for non audio-rate, control signals.

Max. Polyphony: 16

In:

This is the module's input. Here you connect any control signal intended to be smoothed.

Smooth:

This control sets the amount of smoothing to be applied. Turning the knob towards the right will result in more smoothing of the signal.

mods:

These are inputs for control signals (usually from an Envelope, LFO or other oscillator) which are used to modulate the amount of smoothing applied to the incoming signal. The mod knobs adjust how much the Smooth function is modulated by the control signals, and are bipolar.

Out:

This is the audio output of this module, where the processed signal emerges.

Half Inverter



This module is useful when dealing with unipolar signals such as envelopes and constant values. It inverts only the positive amplitude range of an incoming signal, so a value of 0 amplitude becomes 1 (maximum amplitude) and a value of 1 becomes 0 (minimum amplitude). It is ideal for creation of inverted envelopes and other unipolar control signals.

Max. Polyphony: 16

In:

This is the module's input where you connect any unipolar signal you want to half-invert.

Out:

This is the audio output of this module, where the half-inverted signal emerges.

Val Freezer



This module samples and holds the input signal when fed with a gate signal. When a gate on message is received, the module will freeze the incoming signal's value on its current state, until a gate off message is received, and the effect is bypassed. When connected between a FleXor Ramp module and a sequencer module, it can be used as a Pause function. It can also transform an incoming LFO waveshape into various new shapes, by feeding the gate input with an additional, slower or faster, LFO.

Max. Polyphony: 16

In:

This is the module's input where you connect any signal you want to freeze.

Gate:

This is the gate input of the module. Connect a gate signal in order to toggle between the bypass and freeze states of this module.

Freeze:

This button is used to freeze the incoming signal manually.

Out:

This is the module's output, where the processed signal emerges

Val Monitor



This module is a high-precision VU meter showing the exact value of the signal it receives (-1 to 1). It can also be used as a precise constant-value module, by entering numbers in the value field, or dragging the meter's indicator.

Monophonic.

In:

This is the module's input, where the signal to be monitored should be connected.

Value Field:

This window shows the value of the incoming signal. When no signal is connected to the module, it can produce a constant value of control signal, according to the value entered on the value field.

Meter Window:

This window is a simple graphical meter view of the monitored or generated control signal. The indicator represents the level and polarity of the incoming signal. A MIDI controller can be assigned to the fader, to control external devices. It can also be moved manually with the mouse in order to set a custom constant value, if nothing is connected to the input.

Out:

This is the module's control output, where a control signal is generated (if the module is being used as a constant-value module).

Absolute Val Monitor



Monophonic.

This module is a high-precision VU meter showing the exact value of the signal it receives (-2³¹ to 2³¹). It can also be used as a precise constant-value module, by entering numbers in the value field, or dragging the meter's indicator.

In:

This is the module's input, where the signal to be monitored should be connected.

Value Field:

This window shows the value of the incoming signal. When no signal is connected to the module, it can produce a constant value of control signal, according to the value entered on the value field.

Meter Window:

This window is a simple graphical meter view of the monitored or generated control signal. The indicator represents the level and polarity of the incoming signal. A MIDI controller can be assigned to the fader, to control external devices. It can also be moved manually with the mouse in order to set a custom constant value, if nothing is connected to the input.

Out:

This is the module's control output, where a control signal is generated (if the module is being used as a constant-value module).

Multi Val Monitor



This module is an efficient control surface for modular patches. It provides 8 faders that can be toggled between uni-polar and bi-polar modes. It can also be used for sending multiple MIDI control signals by assigning the faders to different control numbers. With its inputs for control signal, this module can be used also as a highly precise 8-channel value monitor.

Max. Polyphony: 5

In 1 through 8:

These are the inputs of this module, where the output of various control signals (such as LFOs and Envelopes) can be connected to monitor their values.

Text Display 1 through 8:

In these text displays, user can type an identifiable title for each of the 8 channels.

Uni / bi Polar Switch 1 through 8 :

A mouse click over the U or B symbols sets whether the outputted value will be unipolar or bipolar.

Meter View 1 through 8:

This meter view indicates the value of a possible incoming control signal. The purple cursor can also be dragged and dropped, in order to produce a desired value at the output stage.

Value Display 1 through 8 :

These displays shows the value of incoming signal. When no signal is connected to the module's inputs, each value display can produce a constant value of control signal, according to the value typed in.

Out 1 through 8

These are the outputs of this module where the control signals emerge.

Gfx:

Turns the graphic representation off to save CPU power

Elastic Val



This module applies elasticity and friction properties to control signals. It is useful for changing the behavior of LFOs, envelopes or any other control signal. This module is optimized for non audio-rate control signals.
Max. Polyphony: 5

In:

This is the module's input stage, where you connect any control signal you want to modify.

Elasticity:

This control sets the amount of Elasticity effect applied to the incoming signal. Turning the knob to the right will increase the Elasticity effect, resulting in a longer time and higher distance of spring-like movements.

Emod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Elasticity time. The Emod knob adjusts the amount of Elasticity modulation derived from the control signal, and is bipolar.

Friction:

This control sets the amount of Friction effect applied to the incoming signal. Turning the knob to the right will increase the Friction effect, resulting in a slower rate and longer time of spring-like movements.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Friction time. The Fmod knob adjusts the amount of Friction modulation derived from the control signal, and is bipolar.

Graphic Display :

this window shows the applied elasticity effect (the blue cursor), applied around its center position (the purple cursor) with a spring-like motion. The purple cursor indicates the value of incoming signal at the module's input stage, and can also be dragged and dropped with the mouse along the graphic display.

Out:

This is the output of this module, where the applied elastic motion emerges.

Elastic X/Y



This module applies elasticity and friction properties to the control signals of an X/Y surface. It is useful for changing the behavior of LFO, and envelope pairs or any other control signal. This module is optimized for non audio-rate, control signals.

Max. Polyphony: 5

In:

This is the module's input stage, where you connect any control signal you want to modify.

Elasticity:

This control sets the amount of Elasticity effect applied to the incoming signal. Turning the knob to the right will increase the Elasticity effect, resulting in a longer time and higher distance of spring-like movements.

Emod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Elasticity time. The Emod knob adjusts the amount of Elasticity modulation derived from the control signal, and is bipolar.

Friction:

This control sets the amount of Friction effect applied to the incoming signal. Turning the knob to the right will increase the Friction effect, resulting in a slower rate and longer time of spring-like movements.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Friction time. The Fmod knob adjusts the amount of Friction modulation derived from the control signal, and is bipolar.

Xmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the blue dot position anywhere along the horizontal axis. The X Mod knob adjusts the amount of modulation which will be applied to the blue dot position, and is bipolar.

Ymod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the blue dot position anywhere along the vertical axis. The Y Mod knob adjusts the amount of modulation which will be applied to the blue dot position, and is bipolar.

Xout:

This is the X output of this module, where the applied elastic motion over the horizontal axis emerges.

Yout:

This is the Y output of this module, where the applied elastic motion over the vertical axis emerges.

X Slider:

This slider sets horizontal position of the blue dot along the X axis.

Y Slider:

This slider sets vertical position of the blue dot along the Y axis.

Graphic Display :

this window shows movement of the elasticity effect (the yellow dot), applied around its center position (the blue dot) with a two dimensional spring-like motion. The blue dot indicates the value of the horizontal (X) and vertical (Y) sliders, and can also be dragged and dropped with the mouse over the graphic display.

Out:

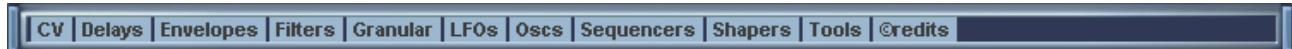
This is the output of this module, where the applied elastic motion emerges.

Gfx:

Turns the graphic representation off to save CPU power

Misc

Flexor Menu



The **FleXor Menu** is a module menu that allows the user to browse easily within the FleXor module pack. Just place it in your patch and click and drag the modules you need.

Text Bar:



On this module you can write notes, thoughts, credits and information regarding your patch. To write, click the text-field, type your text and press [enter].

Text Board



On this module you can write notes, thoughts, credits and information about your patch. To write click on any line, type your text and press [enter]. Moving to another line is done simply by clicking it.

M/S Encoder



M/S Encoder converts a stereo signal (left / right) into a m/s signal (middle sides). This is done by placing the Left channel at the center (middle), adding the right channel in the left and subtracting it on the right (sides) and attenuating everything by half.

L/R inputs:

This is the audio input of the module. Here you connect your stereo audio for encoding to Middle Sides.

M/S outputs:

This is the audio output of the module. Here the M/S (Middle/Sides) signal emerges.

M/S Decoder



M/S Decoder converts an m/s signal (middle sides) into a stereo signal (left / right). This is done by placing the Left channel at the center (middle), adding the right channel in the left and subtracting it on the right (sides).

L/R inputs:

This is the audio input of the module. Here you connect your M/S (Middle/Sides) audio for decoding to Stereo.

M/S outputs:

This is the audio output of the module. Here the Stereo signal emerges.

Michu's Avoider

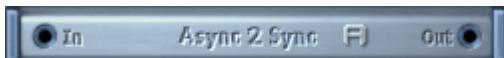


Michu's Avoider is a module designed to combine 2 CV signals in such a way that each signal's strength attenuates the other signal (which 'Avoids' clipping). In other words, as each of the input signals gets further away from 0 amplitude it attenuates the other input signal more. When both inputs have a signal, the A+B output contains both signals combined plus the individual outputs A and B provide a form of ring modulation (where both outputs are unique). If only one input has signal present--the other input has 0 amplitude--then Avoider allows the signal that is present to pass through unchanged to the output that corresponds to its input (A or B) as well as output A+B.

from 0 amplitude it attenuates the other input signal more. When both inputs have a signal, the A+B output contains both signals combined plus the individual outputs A and B provide a form of ring modulation (where both outputs are unique). If only one input has signal present--the other input has 0 amplitude--then Avoider allows the signal that is present to pass through unchanged to the output that corresponds to its input (A or B) as well as output A+B.

I/O – please read above description.

Async 2 Sync



Async 2 Sync converts asynchronous signals (such as constant value, note, velocity, gate and so on...) to synchronous signals (audio-rate signals). This is useful for connecting modules that modular refuses to connect.

In:

This is an Async input for asynchronous signals.

Out:

This is a Sync output generating synchronous signals.

Note: For more information about Async vs Sync, please refer to the Sonic-Core Modular manual.

Sync 2 Async



Sync 2 Async converts synchronous signals (audio-rate signals) to asynchronous signals (such as constant value, note, velocity, gate and so on). This is useful for connecting modules that modular refuses to connect.

In:

This is a Sync input for synchronous signals.

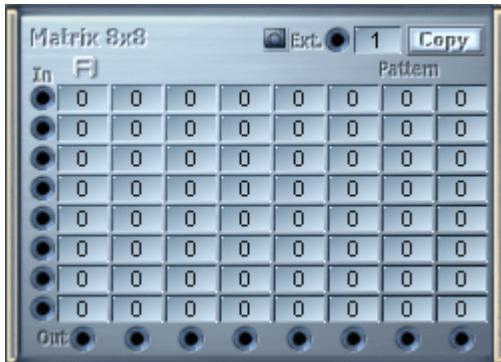
Out:

This is an Async output generating asynchronous signals.

Note: For more information about Async vs Sync, please refer to the Sonic-Core Modular manual.

Mix n Route

Matrix 8X8



Matrix 8X8 is a routing matrix with 8 inputs that are routed and mixed to any of its 8 outputs. This makes this module ideal for implementing very complex routing schemes. Matrix 8X8 also has 128 editable routing patterns that you can customize to instantly switch between different routing schemes.

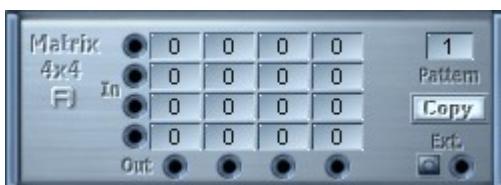
Ins:

These are the audio inputs for the matrix, here you connect signals to be routed to the different outputs of the matrix.

Outs:

These are the audio outputs of the matrix, here the routed and mixed signals emerge.

Matrix 4X4



Matrix 4X4 is a routing matrix with 4 inputs that are routed and mixed to any of its 4 outputs. This makes this module ideal for implementing very complex routing schemes. Matrix 4x4 also has 128 editable routing patterns that you can customize to instantly switch between different routing schemes.

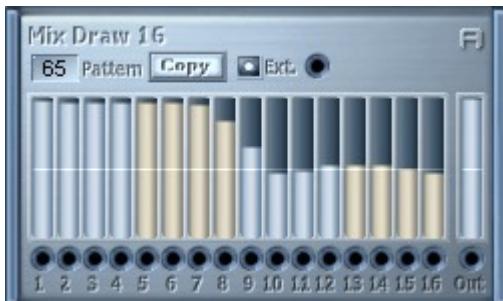
Ins:

These are the audio inputs for the matrix, here you connect signals to be routed to the different outputs of the matrix.

Outs:

These are the audio outputs of the matrix, here the routed and mixed signals emerge.

Mix Draw 16



Mix Draw 16 is a simple fader driven mixer that mixes 16 inputs to one master fader out. It is possible to drag the mouse smoothly across the faders to adjust them. This module is ideal for additive synthesis, harmonic distortion and any other mixing task involving up to 16 signals.

Ins:

These are the audio inputs for the mixer, here you connect signals to be mixed together.

Faders 1-16:

These faders set the value of the control signal which will be generated at each step. The control range of the faders is bipolar, so at the centered zero position no signal will be generated. Setting the faders up or down enables the generation of positive and negative control signal values. This controller allows freely drawing the position of the faders in one mouse movement.

Pattern:

The pattern of sequence now presented.

Ext. Patt:

When on, the input next to the button, allows a ramp signal to switch between the 128 patterns.

Copy/Paste:

When the button is on copy state, pressing it, will copy the pattern to the clipboard. Once pressed it will change to "Paste" which will enable you to browse the patterns. Once you found the number of pattern you want to use, click paste, and the pattern will be duplicated to the new pattern slot.

Out:

This is the audio output of the mixer, here a mixed signal emerges.

ModMix 2



ModMix 2 is a simple mixer with 2 inputs mixed to one output. It is possible to modulate the level of either input for dynamic mixing tasks.

Ins:

These are the audio inputs for the mixer, here you connect signals to be mixed together.

Mods:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the amplitude of the incoming audio. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Out:

This is the audio output of the mixer, here a mixed signal emerges.

ModMix 4



ModMix 4 is a simple mixer with 4 inputs mixed to one output. It is possible to modulate the level of any or all of the inputs for dynamic mixing.

Ins:

These are the audio inputs for the mixer, here you connect signals to be mixed together.

Mods:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the amplitude of the incoming audio. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Out:

This is the audio output of the mixer, here a mixed signal emerges.

ModMix 8



ModMix 8 is a simple mixer with 8 inputs mixed to one output. It is possible to modulate the level of any or all of the inputs for dynamic mixing.

Ins:

These are the audio inputs for the mixer, here you connect signals to be mixed together.

Mods:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the amplitude of the incoming audio. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Out:

This is the audio output of the mixer, here a mixed signal emerges.

Pitch n Tune

Freq 2 Note



glide and fm information.

Freq 2 Note converts a Frequency signal to a Note signal. This module is ideal for filter tracking that includes

Freq:

This is the frequency input of this module. It accepts normalized frequency values (usually from the MVC).

Note:

This is the note output of this module, here the pitch scale emerges.

Note 2 Freq



Note 2 Freq converts a Note signal to a Frequency signal. This module is ideal for oscillator pitch modulation.

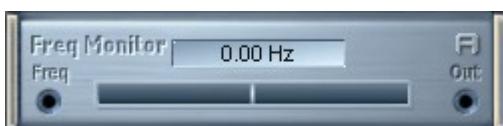
Note In:

This is the note input of this module. It accepts pitch values (usually from the MVC).

Freq Out:

This is the frequency output of this module. Here the frequency signal emerges (normalized frequency)

Freq Monitor



Freq Monitor is a high-precision VU meter that shows the exact value (in Hertz) for the frequency signal it is receiving. It can also be used as the source of a precise constant frequency by entering the desired frequency in the value field, or by dragging the meter's indicator.

Freq:

This is the frequency input of this module. It accepts normalized frequency values (usually from the MVC).

Freq Out:

This is the frequency output of this module. Here the frequency signal emerges (normalized frequency)

Scale Tuning



Scale Tuning is a microtonal pitch processor that can alter the pitch of individual notes to create custom tunings. This makes it ideal for creating various exotic scales, symmetric scales, pentatonic scales, Arabic maqamat, Indian ragas and so on. It is equipped with dozens of scale presets to act as starting points for your microtonal explorations.

Freq:

This is the frequency input of this module. It accepts normalized frequency values (usually from the MVC).

Note:

This is the note input of this module. It accepts pitch values (usually from the MVC).

Important: in order for this module to work, both freq and note values have to be connected. You may use the Note 2 Freq and Freq 2 Note converters if you have only one of the signals.

Keyboard:

Each key on the keyboard has two text faders. The top controls the pitch shifting in semitones, and the bottom controls the pitch shifting in cents. Use this keyboard to create new interesting scale tunings!

Presets:

Scale tuning comes with a huge list of scales to choose from

Base Note:

Since not all scales are symmetric you may choose the starting note for the scale. The starting note will take its position from C on the keyboard.

Switches

ModSwitch 2 to 1



ModSwitch 2 to 1 is an externally controlled switch with 1 output that is switched between 2 inputs. The switch is controlled by an external Ramp signal (or any bipolar signal) and is sample accurate.

Ins:

These are the audio inputs of this module. Connect the signals you want to be switched here.

Ramp:

This is an input for a control signal, which dictates the playing order of 2 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (2 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO wave-shape cycle.

Out:

This is the audio output of this module, where the processed sound emerges.

ModSwitch 4 to 1



ModSwitch 4 to 1 is an externally controlled switch with 1 output that is switched between 4 inputs. The switch is controlled by an external Ramp signal (or any bipolar signal) and is sample accurate.

Ins:

These are the audio inputs of this module. Connect the signals you want to be switched here.

Ramp:

This is an input for a control signal, which dictates the playing order of 4 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (4 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO wave-shape cycle.

Out:

This is the audio output of this module, where the processed sound emerges.

ModSwitch 8 to 1



ModSwitch 8 to 1 is an externally controlled switch with 1 output that is switched between 8 inputs. The switch is controlled by an external Ramp signal (or any bipolar signal) and is sample accurate.

Ins:

These are the audio inputs of this module. Connect the signals you want to be switched here.

Ramp:

This is an input for a control signal, which dictates the playing order of 8 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (8 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO waveshape cycle.

Out:

This is the audio output of this module, where the processed sound emerges.

ModSwitch 16 to 1



ModSwitch 16 to 1 is an externally controlled switch with 1 output that is switched between 16 inputs. The switch is controlled by an external Ramp signal (or any bipolar signal) and is sample accurate.

Ins:

These are the audio inputs of this module. Connect the signals you want to be switched here.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO wave-shape cycle.

Out:

This is the audio output of this module, where the processed sound emerges.

ModSwitch 1 to 2



ModSwitch 1 to 2 is an externally controlled switch with 1 input that is switched between 2 outputs. The switch is controlled by an external Ramp signal (or any bipolar signal) and is sample accurate.

In:

This is the audio input of the switch. Connect the signal you wish to distribute to several inputs here.

Ramp:

This is an input for a control signal, which dictates the playing order of 2 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (2 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO wave-shape cycle.

Outs:

These are the outputs of the switch. The signal will only emerge in one of the outputs, depending on the ramp state.

ModSwitch 1 to 4



ModSwitch 1 to 4 is an externally controlled switch with 1 input that is switched between 4 outputs. The switch is controlled by an external Ramp signal (or any bipolar signal) and is sample accurate.

In:

This is the audio input of the switch. Connect the signal you wish to distribute to several inputs here.

Ramp:

This is an input for a control signal, which dictates the playing order of 4 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (4 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO wave-shape cycle.

Outs:

These are the outputs of the switch. The signal will only emerge in one of the outputs, depending on the ramp state.

ModSwitch 1 to 8



ModSwitch 1 to 8 is an externally controlled switch with 1 input that is switched between 8 outputs. The switch is controlled by an external Ramp signal (or any bipolar signal) and is sample accurate.

In:

This is the audio input of the switch. Connect the signal you wish to distribute to several inputs here.

Ramp:

This is an input for a control signal, which dictates the playing order of 8 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (8 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO wave-shape cycle.

Outs:

These are the outputs of the switch. The signal will only emerge in one of the outputs, depending on the ramp state.

ModSwitch 1 to 16



ModSwitch 1 to 16 is an externally controlled switch with 1 input that is switched between 16 outputs. The switch is controlled by an external Ramp signal (or any bipolar signal) and is sample accurate.

In:

This is the audio input of the switch. Connect the signal you wish to distribute to several inputs here.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO wave-shape cycle.

Outs:

These are the outputs of the switch. The signal will only emerge in one of the outputs, depending on the ramp state.

Threshold Switch 1 to 2



Threshold Switch 1 to 2 is an externally controlled switch--with a threshold--that switches the module's 1 input between 2 outputs. The switch's Threshold is compared to the external bipolar signal connected to the Switch input. The switching occurs at the selected

Threshold point and is sample accurate.

In:

This is the audio input of this module. Connect the signal you want to distribute to different outputs here.

Threshold:

This parameter sets the threshold of the signal switch. If the switch signal is lower than the threshold value, the in signal will emerge through out 1. If the switch signal is higher than the threshold value, the in signal will emerge through out 2 .

Tmod:

These is an input for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the Threshold. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Switch:

This is an audio input, by which the module switches between the outputs.

Outs:

These are the outputs of the switch. The signal will only emerge in one of the outputs, depending on the switch state.

Threshold Switch 2 to 1



Threshold Switch 2 to 1 is an externally controlled switch--with a threshold--that switches the module's 1 output between 2 inputs. The switch's Threshold is compared to the external bipolar signal connected to the

Switch input. The switching occurs at the selected Threshold point and is sample accurate.

Ins:

These are the audio inputs of this module. Connect the signals you want to be switched here.

Threshold:

This parameter sets the threshold of the signal switch. If the switch signal is lower than the threshold value, the output will use input 1. If the switch signal is higher than the threshold value, the output will use input 2.

Tmod:

These are an input for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the Threshold. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Switch:

This is an audio input, by which the module switches between the inputs.

Out:

This is the audio output of this module, where the processed sound emerges.

Note Driven Switch



Note Driven Switch is a note controlled switch that compares the input notes/values to the notes/values on the four text faders. When the note/value selected on one of the faders is received from MVC, that text fader's

corresponding input will be routed to the module's output. This is useful for switching between parameters using the keyboard.

Note:

Connect a note signal (usually from the MVC) to tell the module which note is playing.

Ins:

These are the audio inputs of this module. Connect the signals you want to be switched here.

Note Text Faders:

These text faders set the note on which the corresponding input is switched on.

Out:

This is the audio output of this module, where the processed sound emerges.

CV

C.V. stands for Control Voltage, a term commonly used in the world of analogue synthesizers in order to describe one of the first control signal standards for interconnection between different synthesizers and synthesis modules.

In the digital domain of FleXor, the term “CV” represents numbers on a 31-bit scale (positive side of 32-bit wave) which are used as a control signal just like voltage in the real world. CV signals come in different types, such as gate signals, modulation signals, frequency signals and so on. These signals can be used to control all the parameters (cutoff, pitch, amplitude, time and whatever your patch contains) of the synthesizer or effect, providing they have the right inputs (modulation inputs, gate inputs, frequency inputs and so on)

FleXor's Audio 2 CV section is dedicated to generating CV signals from incoming audio by analyzing its sound characteristics, like its amplitude and pitch. It enables analyzing elements of an incoming sound, and using the resulting control signals to control synthesis and effect parameters.

Here is a combined example:

Hook a monophonically-playing guitar to Pitch 2 CV to extract its pitch, and apply it on an oscillator, by connecting it to its frequency input.

Hook the same source to Hyper Follower to extract the guitar's amplitude. Connect the oscillator to a VCA and the Hyper Follower to the VCA's mod input to apply the guitar's dynamic behavior on the oscillator.

Connect the same guitar source to Audio 2 Gate to trigger an envelope that will control the synthesizer's filter. Add a filter after the oscillator, and connect an envelope to Gate output of Audio 2 Gate so it will be triggered according to the guitar. Connect the envelope to the filter's modulation input to control its cutoff.

Now you have a monophonic synthesizer controlled by a guitar, which translates some of its expression characteristics to the implementation of dynamics and timbre of the synth.

Follower



chains), auto-filter effects and so on.
Poly15

In:

This is the audio input of the module. It is optimized for audio signals which have distinguishable changes of amplitude, like drum loops, vocal lines, guitar solos, and so on.

Attack:

This knob controls the smoothness of positive amplitude changes (signals going from quiet to loud). Turning the knob to the right will result in smoother, slower amplitude transitions in the attack stage, while turning it towards the left will result in faster and tighter amplitude changes, until it reaches the loudest peak.

Release:

This knob controls the smoothness of negative amplitude changes (signals going from loud to quiet). Turning the knob to the right will result in smoother, slower amplitude transitions in the release stage, while turning it towards the left will result in faster and tighter amplitude changes, until a new peak is detected.

Level:

This parameter allows you to fine tune the level of the envelope's signal.

Out:

This is the envelope output of the module, where the generated envelope emerges. This output can be connected to any modulation input on modules such as filters, VCAs, switches, oscillators and so on.

Follower is a classic envelope follower that analyzes an incoming audio signal's amplitude to produce the control signal at its output. The behavior of the envelope is set with the provided attack and release controls. Use Follower for making dynamic effects (compressors, expanders, side-

Follower Mod



Follower Mod is a classic envelope follower with modulation inputs that analyzes an incoming audio signal's amplitude to produce the control signal at its output. The behavior of the envelope is set with the provided attack and release controls. The envelope's attack & release controls can also be modulated, resulting in very dynamic changes

to the envelope's response. Use Follower Mod for making dynamic effects (compressors, expanders, side-chains), auto-filter effects and so on.

Poly15

In:

This is the audio input of the module. It is optimized for audio signals which have distinguishable changes of amplitude, like drum loops, vocal lines, guitar solos, and so on.

Attack:

This knob controls the smoothness of positive amplitude changes (signals going from quiet to loud). Turning the knob to the right will result in smoother, slower amplitude transitions in the attack stage, while turning it towards the left will result in faster and tighter amplitude changes, until it reaches the loudest peak.

Release:

This knob controls the smoothness of negative amplitude changes (signals going from loud to quiet). Turning the knob to the right will result in smoother, slower amplitude transitions in the release stage, while turning it towards the left will result in faster and tighter amplitude changes, until a new peak is detected.

Amod:

This is an input for a control signal (usually from a sequencer module, LFO, or even another envelope), which is used to modulate the Attack time. The Amod knob adjusts the amount of Attack modulation, and is bipolar.

Rmod:

This is an input for a control signal (usually from a sequencer module or LFO, or even another envelope), which is used to modulate the Release time. The Rmod knob adjusts the amount of Release modulation, and is bipolar.

Level:

This parameter allows you to fine tune the level of the envelope's signal.

Out:

This is the envelope output of the module, where the generated envelope emerges. This output can be connected to any modulation input on modules such as filters, VCAs, switches, oscillators and so on.

Threshold Follower



Threshold Follower is an envelope generator with a threshold that is compared to the incoming audio signal's amplitude. When the signal's amplitude exceeds the selected threshold, the envelope's attack stage is triggered. When the signal falls below the threshold the envelope

enters the release stage. The envelope's behaviour is linear. Use Threshold Follower to make dynamic effects (compressors, expanders, side-chains), auto-filter effects and so on.
Poly16

In:

This is the audio input of the module. It is optimized for audio signals which have distinguishable changes of amplitude, like drum loops, vocal lines, guitar solos, and so on.

Threshold:

This parameter sets the threshold on which the audio switches from attack or release. If the audio input is louder than the threshold, the envelope will switch to attack stage and rise. If it is lower than the threshold, the envelope will switch to the release stage and fall.

Attack:

This knob controls the smoothness of positive amplitude changes (signals going from quiet to loud). Turning the knob to the right will result in smoother, slower amplitude transitions in the attack stage, while turning it towards the left will result in faster and tighter amplitude changes, until it reaches the loudest peak.

Release:

This knob controls the smoothness of negative amplitude changes (signals going from loud to quiet). Turning the knob to the right will result in smoother, slower amplitude transitions in the release stage, while turning it towards the left will result in faster and tighter amplitude changes, until a new peak is detected.

Level:

This parameter allows you to fine tune the level of the envelope's signal.

Out:

This is the envelope output of the module, where the generated envelope emerges. This output can be connected to any modulation input on modules such as filters, VCAs, switches, oscillators and so on.

Threshold Follower Mod



Threshold Follower Mod is an envelope generator with a threshold that is compared to the incoming audio signal's amplitude and modulation inputs. When the signal's amplitude exceeds the selected threshold, the envelope's attack stage is triggered. When the signal falls below the threshold the envelope enters the release stage. The envelope's behaviour is linear. The envelope's threshold, attack and release controls can also be modulated resulting

in very dynamic changes to the envelope's response. Use Follower Mod for making dynamic effects (compressors, expanders, side-chains), auto-filter effects and so on.

Poly16

In:

This is the audio input of the module. It is optimized for audio signals which have distinguishable changes of amplitude, like drum loops, vocal lines, guitar solos, and so on.

Threshold:

This parameter sets the threshold on which the audio switches from attack or release. If the audio input is louder than the threshold, the envelope will switch to attack stage and rise. If it is lower then the threshold, the envelope will switch to the release stage and fall.

Attack:

This knob controls the smoothness of positive amplitude changes (signals going from quiet to loud). Turning the knob to the right will result in smoother, slower amplitude transitions in the attack stage, while turning it towards the left will result in faster and tighter amplitude changes, until it reaches the loudest peak.

Release:

This knob controls the smoothness of negative amplitude changes (signals going from loud to quiet). Turning the knob to the right will result in smoother, slower amplitude transitions in the release stage, while turning it towards the left will result in faster and tighter amplitude changes, until a new peak is detected.

Tmod:

This is an input for a control signal (usually from a sequencer module, LFO, or even another envelope), which is used to modulate the Threshold. The Amod knob adjusts the amount of Threshold modulation, and is bipolar.

Amod:

This is an input for a control signal (usually from a sequencer module, LFO, or even another envelope), which is used to modulate the Attack time. The Amod knob adjusts the amount of Attack modulation, and is bipolar.

Rmod:

This is an input for a control signal (usually from a sequencer module or LFO, or even another envelope), which is used to modulate the Release time. The Rmod knob adjusts the amount of Release modulation, and is bipolar.

Level:

This parameter allows you to fine tune the level of the envelope's signal.

Out:

This is the envelope output of the module, where the generated envelope emerges. This output can be connected to any modulation input on modules such as filters, VCAs, switches, oscillators and so on.

Audio 2 Gate



This module analyzes the amplitude changes of the audio input, in order to produce gate messages at its output. A Threshold function sets the minimum input level which the incoming signal should exceed in order to trigger a gate signal, while the Time function sets the

time gap between one gate message and the next. This module is especially suitable for extracting grooves from live drums, in high precision.

Monophonic.

In:

This is the audio input of the module. It is optimized for audio signals which have distinguishable changes of amplitude, like drum loops, vocal lines, guitar solos, and so on.

Time:

This function sets the time gap between one gate message and the next. It is important to set this parameter to a time value which is suited to the transients of the incoming audio signal. With proper use of the threshold function, the module's LED helps to set the right Time value.

Threshold:

This function sets the minimum level which the amplitude of the incoming audio signal needs to exceed in order to produce a gate message.

LED indicator:

This LED blinks every time a gate signal is produced by this module.

Gate:

This is the output of the module, where the generated gate messages emerge. It can be connected to the Gate input on various modules, in order to trigger envelopes, ramps, VCAs, LFOs and so on.

Hyper Follower



This module is a fast-responding envelope follower. It analyzes the amplitude changes of the incoming audio signal, in order to produce an envelope signal at its output. The behavior of the envelope is set with the provided attack and release controls. These can also be modulated, resulting in dynamic changes in the envelope's response. It is useful for making dynamic effects (compressors, expanders, side-chains), as well as auto-filter effects.

Max. Polyphony: 15

In:

This is the audio input of the module. It is optimized for audio signals which have distinguishable changes of amplitude, like drum loops, vocal lines, guitar solos, and so on.

Attack:

This knob controls the smoothness of positive amplitude changes (signals going from quiet to loud). Turning the knob to the right will result in smoother, slower amplitude transitions in the attack stage, while turning it towards the left will result in faster and tighter amplitude changes, until it reaches the loudest peak.

Release:

This knob controls the smoothness of negative amplitude changes (signals going from loud to quiet). Turning the knob to the right will result in smoother, slower amplitude transitions in the release stage, while turning it towards the left will result in faster and tighter amplitude changes, until a new peak is detected.

Amod:

This is an input for a control signal (usually from a sequencer module, LFO, or even another envelope), which is used to modulate the Attack time. The Amod knob adjusts the amount of Attack modulation, and is bipolar.

Rmod:

This is an input for a control signal (usually from a sequencer module or LFO, or even another envelope), which is used to modulate the Release time. The Rmod knob adjusts the amount of Release modulation, and is bipolar.

Out:

This is the envelope output of the module, where the generated envelope emerges. This output can be connected to any modulation input on modules such as filters, VCAs, switches, oscillators and so on.

Pitch 2 CV



This module analyzes monophonic audio signals of a melodic nature, and produces a corresponding frequency and gate signal at its output. It is ideal for extracting the pitch from external sources, and applying it on modular oscillators in real-time. The Low Note and Range controllers are used to tune the frequency spectrum of the analysis. Different tracking methods are available for different source types. The gate output is optional. It helps in blocking notes under the indicated

Low Note. This module is optimized to track the pitch of instruments.

Monophonic.

In:

This is the audio input of the module, where you connect the incoming audio signal in order to extract its pitch.

Low Note:

Click and drag up/down on this text-fader to set the lowest note which will be recognized by the Pitch to CV module. Pitch to CV will not recognize notes lower than the displayed note.

Range:

This function sets the range in which the Pitch to CV module recognizes pitch. The Range knob adjusts the range up from the note indicated by the Low note window.

Portamento:

This function sets the time for a pitch glide between different notes. At its zero position (extreme left), pitch changes from one note to the next will be produced as they originally appear at the incoming audio signal. Turning the knob clockwise will add a gradual transition of pitch between the notes, creating a smooth pitch glide.

Tracking Method:

Click and drag up/down on this text-fader to set the pitch-recognition method which will be used to track the fundamental frequency of the incoming audio signal. Accurate is the fastest, while the Smart- Track methods will detect the pitch dynamically. Each Smart-Track method has a different feel, with 1 having the widest and fastest response, and 3 having the tightest pitch detection.

Monitor:

This switch activates the monitoring function for the Hz and Samples Per Cycle displays.

Hz:

This window is activated only when the Monitor switch is enabled. It shows the frequency of the generated signal, in Hertz (cycles per second), as recognized by the Pitch to CV module. Samples Per Cycle: This window is activated only when the Monitor switch is enabled. It displays the number of samples in each cycle of the incoming audio signal, as recognized by the Pitch to CV module.

Freq:

This is the frequency output of the module, where the generated frequency signal emerges. It is usually connected to the frequency input of an oscillator, in order to make the oscillator play the melody extracted from the incoming audio signal.

Gate:

This is the gate output of this module, where generated gate messages emerge. These gate messages are generated every time the Pitch to CV module recognizes an incoming audio signal at a frequency higher than the note indicated at the Low note window. This gate output is optional, needed only on certain occasions. It is useful to help eliminate occasional glitches of pitch recognition, caused by sub - harmonic content in the incoming audio signal. When used, it is usually connected to the gate input of an envelope which controls a VCA module that handles the audio output of the played oscillator, just like a standard noise gate circuit.

Pitch 2 CV II



This module analyzes monophonic audio signals of a melodic nature, and produces a corresponding frequency and gate signal at its output. It is ideal for extracting the pitch from external sources, and applying it on modular oscillators in real-time. The Low Note and Range controllers are used to tune the frequency spectrum of the analysis. Different tracking methods are available for different source types. The gate output is optional. It helps in blocking notes under the indicated

Low Note. This module is optimized to track the pitch of vocals.

Monophonic.

In:

This is the audio input of the module, where you connect the incoming audio signal in order to extract its pitch.

Low Note:

Click and drag up/down on this text-fader to set the lowest note which will be recognized by the Pitch to CV II module. Pitch to CV II will not recognize notes lower than the displayed note.

Range:

This function sets the range in which the Pitch to CV II module recognizes pitch. The Range knob adjusts the range up from the note indicated by the Low note window.

Portamento:

This function sets the time for a pitch glide between different notes. At its zero position (extreme left), pitch changes from one note to the next will be produced as they originally appear at the incoming audio signal. Turning the knob clockwise will add a gradual transition of pitch between the notes, creating a smooth pitch glide.

Tracking Method:

Click and drag up/down on this text-fader to set the pitch-recognition method which will be used to track the fundamental frequency of the incoming audio signal. Accurate is the fastest, while the Smart- Track methods will detect the pitch dynamically. Each Smart-Track method has a different feel, with 1 having the widest and fastest response, and 3 having the tightest pitch detection.

Monitor:

This switch activates the monitoring function for the Hz and Samples Per Cycle displays.

Hz:

This window is activated only when the Monitor switch is enabled. It shows the frequency of the generated signal, in Hertz (cycles per second), as recognized by the Pitch to CV II module.

Samples Per Cycle:

This window is activated only when the Monitor switch is enabled. It displays the number of samples in each cycle of the incoming audio signal, as recognized by the Pitch to CV II module.

Freq:

This is the frequency output of the module, where the generated frequency signal emerges. It is usually connected to the frequency input of an oscillator, in order to make the oscillator play the melody extracted from the incoming audio signal.

Gate:

This is the gate output of this module, where generated gate messages emerge. These gate messages are generated every time the Pitch to CV II module recognizes an incoming audio signal at a frequency higher than the note indicated at the Low note window. This gate output is optional, needed only on certain occasions. It is useful to help eliminate occasional glitches of pitch recognition, caused by sub-harmonic content in the incoming audio signal. When used, it is usually connected to the gate input of an envelope which controls a VCA module that handles the audio output of the played oscillator, just like a standard noise gate circuit.

Filters

A filter is a sound processor that blocks a certain portion of frequencies in a signals spectrum, and lets the rest pass. Filters are very effective for creating extreme timbre changes, and yield best results on signals rich in overtones.

Usually, a filter's name reflects its filtering characteristic. For example, a "low-pass" filter allows only low frequencies to pass through to its output, while filtering out the high frequencies.

In the audio and sound synthesis world, most filters have a variable frequency cutoff point, which sets the range of frequencies to be filtered out; i.e. the frequency from which the filtering will apply.

Modifying the filter's cutoff point can produce dramatic effects, as long as the incoming signal contains enough frequencies to be filtered.

Some filters can emphasize the cutoff frequency. This function is called resonance, as it creates a resonating peak at the cutoff point. Other common names for this function are Q (used in EQs) and Emphasis (used in old analog synthesizers). Resonance is achieved by feeding the filter's output back into its input. All resonant FleXor filters are now equipped with resonance modulation input (Rmod)

FleXor has 7 primary filter types:

Low-pass filters:

These filter frequencies higher than the cutoff point and let the lower frequencies pass, creating a darker and fatter filter character.

High-pass Filters:

These filter frequencies below the cutoff point and let high frequencies pass, creating a sharp and thin filter character.

Band-pass Filters:

These filter high and low frequencies either side of the cutoff point, and let a narrow band of frequencies pass through, in order to achieve soft and round or sharp and edgy characters.

Band-reject Filters:

These filter frequencies at the cutoff point, and leaving all but a narrow band of frequencies to pass through, in order to achieve phasing and suction style characters.

Ridge Filters:

These implement a complex filtering of high, low and mid frequencies to create peaks and notches, resulting in a special fusion between a band-pass filter and a phaser.

Comb Filters:

These filter frequencies at a harmonic structure, and but a narrow band of frequencies to pass through, to create peaks and notches very much like a flanger.

All-Pass Filters:

These filters alter the phase content of the signal allowing to create varous filters ranging from resonators, band rejects, to phasers and so on...

A lot of care has been taken to create the FleXor filter collection. Each module is designed and optimized to achieve a unique character of its own, with extreme attention to detail

Low Pass

LP-1



This is a one-pole (6db) low-pass filter. It is the basic building block of all the FleXor filters. It is ideal for creating your own filter combinations or when a mild filtering effect is needed. Its modulation inputs are optimized to receive audio-rate modulation sources.

Max. Polyphony: 16

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter: frequencies above this point will be filtered. Turning the knob to the left will result in less high frequencies.

mods:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

LP-4 Legend



This is a four-pole (24db) low-pass filter. Its classic, warm character is ideal for creating classic leads, bass sounds, pads and more. Its modulation inputs are optimized to receive audio-rate modulation sources (like oscillators) to create amazing filter modulation effects.

Max. Polyphony: 5

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

LP-4 Multistage



This is a four-pole (24db) low-pass filter with a unique feature: it enables free routing from any stage (1 to 4) within the internal filter structure, to be sent to the final audio output of the filter. The resonance has a unique sound character, as it is always fed-back from the fourth filter stage, no matter which stage is selected to be heard. This filter's harmonic character is more

complex than standard low-pass filters, making it ideal for interesting pad and lead sounds.

Max. Polyphony: 7

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Stage:

This text fader (click and drag up/down) allows selection of any one of the four internal filter stages to be routed to the final audio output. You can select between a 6db filter (stage 1), 12db filter (stage 2), 18db filter (stage 3) and 24db filter (stage 4). The resonance feedback is always fed from the last stage, creating new and interesting filter types.

LP-8 Butter



Max. Polyphony: 4

This is an eight-pole (48db) low-pass filter. Its slope is twice as steep as the more common four-pole or 24db filter type. The deep slope produces an extreme result of filtered sound, and is great for quacking filters, wah-wahs, throaty bass-lines and other funky sounds. The unique FaT feature allows a rich and warm resonance saturation.

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

FaT:

This function alters the internal gain structure of the filter, in order to boost fatness and warmth. When fully engaged, a desirable analogue-like soft saturation is obtained.

LP-3 Rubber



LP-3 Rubber is a three-pole (18dB) Low-Pass filter. It has internal filter saturation and a screaming resonance. The “rubbery” and “juicy” sound it imparts is great for classic acid type synthesizer riffs and basses.

Poly 4

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above this point will be filtered, passing through only the low frequencies lower than the cutoff frequency.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

LP-6 Aqua



Poly 4

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above this point will be filtered, passing through only the low frequencies lower than the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Stage:

This text fader (click and drag up/down) allows selection of any one of the four internal filter stages to be routed to the final audio output. You can select between a 6db filter (stage 1), 12db filter (stage 2), 18db filter (stage 3), 24db filter (stage 4), 30db filter (stage 5) and 36db filter (stage 6). The resonance feedback is always fed from the last stage, creating new and interesting filter types.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

High Pass

HP-1



Max. Polyphony: 16

This is a one-pole (6db) high-pass filter. It is the basic building block of all the FleXor filters. It is ideal for creating your own filter combinations or when a mild filtering effect is needed. Its modulation inputs are optimized to receive audio-rate modulation sources.

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter: frequencies below this point will be filtered.

Turning the knob to the right will result in less low frequencies.

mods:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

HP-4 Legend



This is a four-pole (24db) high-pass filter. Its sharp yet silky character is ideal for creating classic leads, pads and riffs. Its modulation inputs are optimized to receive audio-rate modulation sources (like oscillators) to create amazing filter modulation effects.

Max. Polyphony: 3

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

HP-4 Multistage



This is a four-pole (24db) high-pass filter with a unique feature: it enables free routing from any stage (1 to 4) within the internal filter structure, to be sent to the final audio output of the filter. The resonance has a unique sound character, as it is always fed-back from the fourth filter stage, no matter which stage is selected to be heard. This filter's harmonic character is more

complex than standard High-Pass filters, making it ideal for interesting pad and lead sounds.

Max. Polyphony: 3

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Stage:

This text fader (click and drag up/down) allows selection of any one of the four internal filter stages to be routed to the final audio output. You can select between a 6db filter (stage1), 12db filter (stage2), 18db filter (stage 3) and 24db filter (stage 4). The resonance feedback is always fed from the last stage, creating new and interesting filter types.

HP-8 Knife



This is an eight-pole (48db) high-pass filter. Its slope is twice as steep as the more common four-pole or 24db filter type. The deep slope produces an extreme result of filtered sound, with a clean yet powerful, edgy character. It is great for aggressive sharp filter sweeps, filtering drums and more.

Max. Polyphony: 2

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

HP-8 Razor



HP-8 Razor is an eight-pole (48db) High-Pass filter. It has an extra steep slope for producing extreme results. It is great for aggressively sharp filter sweeps, filtering drums and more with a very powerful, sharp and edgy character to the filtered sound
Poly 3

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies below this point will be filtered, passing through only the frequencies higher than the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Band Pass

BP-1



Max. Polyphony: 16

This is a one-pole (6db) band-pass filter. It is the basic building block of all the FleXor filters. It is ideal for creating your own filter combinations or when a mild filtering effect is needed. Its modulation inputs are optimized to receive audio-rate modulation sources.

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

mods:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

BP-4 Legend



This is a four-pole (24db) band-pass filter. Its classic, warm character is ideal for creating classic leads, bass sounds, pads and more. Its modulation inputs are optimized to receive audio-rate modulation sources (like oscillators) to create amazing filter modulation effects.

Max. Polyphony: 4

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

BP-4 Multistage



This filter is a four-pole (12db) Band-pass type with a unique feature: it enables free routing from any stage (1 to 4) within the internal filter structure, to be sent to the final audio output of the filter. The resonance has a unique sound character, as it is always fed-back from the fourth filter stage, no matter which stage is selected to be heard. This filter's harmonic character is more

complex than standard Band-Pass filters, making it ideal for interesting pad and lead sounds.

Max. Polyphony: 5

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Stage:

This text fader (click and drag up/down) allows selection of any one of the four internal filter stages to be routed to the final audio output. You can select between a 6db low-pass filter (stage 1), 6db band-pass filter (stage 2), 12db asymmetric band-pass filter (stage 3) and 12db band-pass filter (stage 4). The resonance feedback is always fed from the last stage, creating interesting filter types.

BP-8 Butter



Max. Polyphony: 3

This is an eight-pole (48db) band-pass filter. Its slope is twice as steep as the more common four-pole or 24db filter type. The deep slope produces an extreme result of filtered sound, and is great for quacking filters, wah-wahs, throaty bass-lines and other funky sounds. The unique FaT feature allows rich and warm resonance saturation.

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

FaT:

This function alters the internal gain structure of the filter, in order to boost fatness and warmth. When fully engaged, a warm filter is obtained.

BP-2 Granite



relationship along the audio frequency spectrum
Poly 12

BP-2 Granite is a two-pole (12dB) Band-Pass filter. It has a mellow, smooth and musical filtering effect that is ideal for pre & post-distortion, and for drum loop processing. The filtering effect also changes the phase

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a band of frequencies at the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

BP-4 Granite



the audio frequency spectrum.

Poly 6

BP-4 Granite is a four-pole (24dB) Band-Pass filter. It has a smooth and musical filtering effect that is ideal for pre & post-distortion, and for drum loop processing. The filtering effect also changes the phase relationship along

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a band of frequencies at the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

BP-8 Granite



the audio frequency spectrum.
Poly 3

BP-8 Granite is an eight-pole (48dB) Band-Pass filter. It has a thin, smooth and musical filtering effect that is ideal for extreme filtering and for drum loop processing. The filtering effect also changes the phase relationship along

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a band of frequencies at the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Band Reject

BR-2 Granite



the audio frequency spectrum.
Poly 11

BR-2 Granite is a two-pole (12dB) Band-Reject filter. It has a mild, smooth and musical filtering effect that is ideal for using on pads, drum loops, and post-distortion. The filtering effect also changes the phase relationship along

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point will be filtered, passing high and low frequencies around the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

BR-4 Granite



the audio frequency spectrum
Poly 6

BR-4 Granite is a four-pole Band-Reject filter. It has a deep, smooth and musical filtering effect that is ideal for using on pads, drum loops, and post-distortion. The filtering effect also changes the phase relationship along

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point will be filtered, passing high and low frequencies around the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

BR-8 Granite



the audio frequency spectrum
Poly 3

BR-8 Granite is an eight-pole Band-Reject filter. It has a heavy, smooth and musical filtering effect that is ideal for using on sequences, drum loops, and post-distortion. The filtering effect also changes the phase relationship along

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point will be filtered, passing high and low frequencies around the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

All Pass

AP-1 Basic



AP-1 Basic is a one-pole (6db) All-Pass filter. It forms the basic building block of some of the other FleXor filters. All audio passes through but changes its phase across the frequency range. This filter is great for creating phaser-effects and phase modulation effects. Its modulation inputs are optimized to receive audio-rate modulation sources.

Poly 16

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point will be filtered, passing through a phase altered signal.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

AP-2 Resonator



AP-2 Resonator is a two-pole All-Pass filter. It creates a resonant peak at the cutoff frequency, and does not block any frequencies. It is handy for boosting certain frequencies in a sound, or for creating interesting resonating sweeping sounds.

Poly 9

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point will be filtered, passing through a phase altered signal.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

AP-3 Formant



AP-3 Formant is a block of 3 multi-mode filters selectable between HighPass, BandPass, LowPass, AllPass and Band-Reject. The filter type control changes all 3 filters together. There are controls and mod inputs for all three filters individually plus all 3 combined, so that they can be controlled separately or all at once. This filter can be used to create custom formant effects or complex filtering interactions for added flavor.

Poly 3

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Type:

This controller allows a filter type selection of HighPass, BandPass, LowPass, AllPass or Band- Reject.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Form 1-3:

These controllers set the cutoff frequencies individually for each of the three filters.

Offset:

This controller sets the offset for the cutoff frequencies for all of the three filters combined.

Fmods:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Omod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the offset of cutoff frequencies of the three filters. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of this Filter, where the processed sound emerges.

AP-3 Vowel A



wish.

Poly 3

AP-3 Vowel A is vowel filter with an X-Y vector control that allows smooth cross-fading between 4 vowel types at once. Each vowel is assignable to any of 11 types (IY, IH, EH, AE, AA, AO, OH, OO, UY, AH, AX). The filter is built from a special array of All-Pass filters yields 3 multi-mode filters which can be switched between High-Pass, Band-Pass, Low-Pass, All-Pass and Band-Reject via the text fader (which switches all 3 to the same type). This filter is great for making vocal effects on any sound source you

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Type:

This controller allows a filter type selection of HighPass, BandPass, LowPass, AllPass or Band-Reject.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Graphic Display:

Drag the dot on the vector controller to change the vowels of the filter. For each corner there is a different vowel represented by the text fader next to it.

Vowel Text Faders:

Each of the text faders allows the selection of 11 vowels: IY, IH, EH, AE, AA, AO, OH, OO, UY, AH or AX. The text faders will affect the corresponding corner of the graphic display.

Xmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the blue dot position anywhere along the horizontal axis. The Xmod knob adjusts the amount of modulation which will be applied to the yellow dot position, and is bipolar.

Ymod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the blue dot position anywhere along the vertical axis. The Ymod knob adjusts the amount of modulation which will be applied to the yellow dot position, and is bipolar.

X Slider:

This slider sets horizontal position of the yellow dot along the X axis.

Y Slider:

This slider sets vertical position of the yellow dot along the Y axis.

Form:

This controllers sets the formant offset of the three internal filters. Use this to change the character of the vocal effect from male to female.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Fmod:

This is an input for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of this Filter, where the processed sound emerges.

AP-3 Vowel B



AP-3 Vowel B is a vector controlled vowel filter with an X-Y vector control that allows smooth cross-fading between 4 vowel types at once. Each vowel is selectable to any of 11 vowel types (IY, IH, EH, AE, AA, AO, OH, OO, UY, AH, AX). It is built from 3 special one-pole All-Pass filters that give a very pronounced vocal character to any sound source that passes through it.

Poly 4

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Graphic Display:

Drag the dot on the vector controller to change the vowels of the filter. For each corner there is a different vowel represented by the text fader next to it.

Vowel Text Faders:

Each of the text faders allows the selection of 11 vowels: IY, IH, EH, AE, AA, AO, OH, OO, UY, AH or AX. The text faders will affect the corresponding corner of the graphic display.

Xmod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the blue dot position anywhere along the horizontal axis. The Xmod knob adjusts the amount of modulation which will be applied to the yellow dot position, and is bipolar.

Ymod:

This is an input for a control signal (such as LFO or Envelope), which is used to modulate the blue dot position anywhere along the vertical axis. The Ymod knob adjusts the amount of modulation which will be applied to the yellow dot position, and is bipolar.

X Slider:

This slider sets horizontal position of the yellow dot along the X axis.

Y Slider:

This slider sets vertical position of the yellow dot along the Y axis.

Form:

This controllers sets the formant offset of the three internal filters. Use this to change the character of the vocal effect from male to female.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Fmod:

This is an input for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

Out:

This is the audio output of this Filter, where the processed sound emerges.

AP-6 Phaser



AP-6 Phaser is a phase shifting filter based on up to 6 dual All-Pass filter stages. The number of stages is selectable from 1-6 with the text fader for up to 6 peaks or notches (selected using the invert button). Its special glassy sound is excellent for using on pads, spacey sequences and to create out of this world effects.

Poly 2

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Stages:

This controller sets the number of stages (peaks and notches) of the phaser.

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point will be filtered, passing through a phase altered signal.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

AP-8 Additive



AP-8 Additive consists of 8 All-Pass filters arranged in parallel as 8 harmonics. This unique filter blends subtractive synthesis with concepts of additive synthesis to create exotic effects ranging from simple resonators to quasi-comb filters and more.

Poly 1

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Cutoff:

This function sets the cutoff frequency of the filters. Frequencies at harmonic intervals will be filtered and emphasized, passing through an altered signal.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Type:

This controller sets the type of filters to be used in this module. It allows the selection between a bell type filter, and a peak type filter.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Comb

CM-1 Flanger



CM-1 Flanger is a comb filter with a classic flanging sound. It can be tuned to have 4 different flavors (depending on the mix amount and invert switch position). This flanger is ideal for modulation effects as well as for use like a normal filter on pads, leads, and sequences.

Poly 6

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Fat:

The parameter can be used to add low frequency content to the flanger and make the flanging effect deeper.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point and up will be filtered, passing through a comb shaped frequency spectrum.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the

keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

CM-1 Phase



CM-1 Phase is a non-recursive phase inverted comb filter. It imparts a quasi-PWM effect when modulated. It also acts as a low-cut filter and is ideal for pads, drum loop mangling and sequences.

Poly 6

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point and up will be filtered, passing through a comb shaped frequency spectrum.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

CM-1 Variable



CM-1 Variable is a classic comb filter with a twist: it has an All-Pass filter in its feedback path. This is useful for creating enharmonic comb filtering, which sounds quite exotic. It also changes the phase relationship along the frequency spectrum of the audio.

Poly 3

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point and up will be filtered, passing through a comb shaped frequency spectrum.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Phase:

this parameter adjusts the phase of the signal going through the feedback path. Turning the knob to the left will result in inverted phase, while turning it to the right will result in a positive phase. This control can add enharmonic overtones to the string.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

Pmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Phase. The Pmod knob adjusts the amount of Phase modulation derived from the control signal, and is bipolar.

CM-2 Chorus



CM-2 Chorus is a classic chorus effect. There is control over the length of the delay lines which makes it useful for creating fresh new chorus effects with different LFO shapes or even envelopes.
Poly 3

Ins:

These are the audio inputs of the module where you connect the audio signal to be processed.

Del L/R:

These parameters control the delay time (max 100 ms).

Dmods:

These are inputs for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay times – and thus achieve a chorus effect. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar. Use a sine LFO to achieve classic chorus effects.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Outs:

These are the audio outputs of this module. Where the processed sound emerges.

CM-2 Cross



CM-2 Cross is a cross-comb filter. It has two delay lines setup in cross feedback to create a very strong and distinct comb filtering effect. It may also be used to create a stereo effect using its two outputs (each output is fed from one of the delay lines).

Poly 3

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

OL/ OR:

These are the audio outputs of this Filter (Left and Right), where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point and up will be filtered, passing through a comb shaped frequency spectrum.

Feedback:

This function sets the feedback amount of the filter. The feedback emphasizes a narrow peaks at the cutoff frequency of the filter and higher. Turning the knob to the right will result in more resonance.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Fmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

CM-2 Detune



flavor types.
Poly 10

CM-2 Detune is a modulated double comb filter. It creates a detuned effect by tripling and detuning any incoming signal (with 2 additional detuned delay lines mixed to the output with the original signal). This filter has 3

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Amount:

This controller sets the amount of detune effect to be applied. Turning the knob to the right will result in more detune effect.

Amod(s):

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Detune Amount. The Amod knob adjusts the amount of Detune modulation derived from the control signal, and is bipolar.

Type:

This text fader allows the selection of 3 detune flavors.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Mono to stereo

M2S Comb



M2S Comb is a mono to stereo filter that uses the MS technique together with a comb filter. The inverted comb filtering effect for each side creates the pseudo stereo effect, and on settings up to 50% wet it is mono compatible.

Poly 6

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out (L/R):

These are the audio outputs of this Filter, where the processed sound emerges.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point and up will be filtered, passing through a comb shaped frequency spectrum.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

M2S Phase



M2S Phase is a mono to stereo filter that uses the MS technique together with a series of All-Pass filters. An inverted phase filtering effect (6 notches) for each side creates the pseudo stereo effect, and on settings up to 50% wet it is mono compatible.
Poly 8

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out (L/R):

These are the audio outputs of this Filter, where the processed sound emerges.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Stages:

This controller sets the number of notches the phase effect will have.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies at this point and up will be filtered, passing through a phase altered frequency spectrum with peaks and notches.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Multi Filters

MF-2 Spice



MF-2 Spice is a common two-pole (12db) state variable filter with access to 4 filter types at once. The filter types are: High-Pass, Band-Pass, Low-Pass and Band-Reject. Each filter type has its own output and is capable of self oscillation.
Poly 8

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies below or above this point will be filtered, depending on the filter type selection.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance. This filter is capable of self resonating.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

HP / BP / LP / BR:

These are the audio output of this Filter, where the processed sound emerges respectively: HP = High Pass, BP = Band Pass, LP= Low Pass, BR = Band Reject.

MF-2 Swiss



MF-2 Swiss is a multi-pole State variable filter selectable between 6 filter types. The 6 filter types are: High-Pass 1 (6dB), High-Pass 2 (12dB), Band-Pass (6dB), Low-Pass 1 (6dB), Low-Pass (12dB) and All-Pass. This filter is capable of self oscillation and has a sharp filtering character that can be used for a variety of filtering operations.

Poly 7

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies below or above this point will be filtered, depending on the filter type selection.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance. This filter is capable of self resonating.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Type:

This text fader allows the selection of 6 different filter types: High-Pass 1 (6dB), High-Pass 2 (12dB), Band-Pass (6dB), Low-Pass 1 (6dB), Low-Pass (12dB) and All-Pass.

Out:

This is the audio output of this Filter, where the processed sound emerges.

MF-4 Swiss



MF-4 Swiss is a multi-pole State variable filter selectable between 15 filter types. The filter types are: High-Pass 1 (6dB), High-Pass 2 (12dB), High-Pass 3 (18dB), High-Pass 4 (24dB), Hi-Band-Pass 1 (6dB), Hi-Band-Pass 2 (12dB), Band-Pass 1 (6dB), Band-Pass 2 (12dB), Low-Band-Pass 1 (6dB), Low-Band-Pass 2 (12dB), Low-Pass 1 (6dB), Low-Pass 2 (12dB), Low-Pass 3 (18dB), Low-Pass 4 (24dB) and All-Pass. This filter is capable of

self oscillation and has an extra resonance knob for extra steep filtering and ultra-thin resonance.
Poly 3

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies below or above this point will be filtered, depending on the filter type selection.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

Res(s):

These functions set the resonance amount of the filters. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance. To achieve extra steep filtering and ultra-thin resonance, use both of the resonance knobs.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Type:

This text fader allows the selection of 15 different filter types: High-Pass 1 (6dB), High-Pass 2 (12dB), High-Pass 3 (18dB), High-Pass 4 (24dB), Hi-Band-Pass 1 (6dB), Hi-Band-Pass 2 (12dB), Band-Pass 1 (6dB), Band-Pass 2 (12dB), Low-Band-Pass 1 (6dB), Low-Band-Pass 2 (12dB), Low-Pass 1 (6dB), Low-Pass 2 (12dB), Low-Pass 3 (18dB), Low-Pass 4 (24dB) and All-Pass.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Special

CO-1 Basic



CO-1 Basic is a crossover filter with High-Pass and Low-Pass outputs. It has a very moderate slope and is good for dividing signals into frequency ranges for multi-band processing. The filter is tuned so that there are no phase issues when the outputs are summed, ie., so there is no sound change.

Poly 16

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

H:

This is the audio output of this Filter, where the high-passed sound emerges.

L:

This is the audio output of this Filter, where the low-passed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through lower frequencies than the cutoff frequency at one output, and higher frequencies at the other.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

CO-2 Spice



CO-2 Spice is a crossover filter with High-Pass and Low-Pass outputs. It has a very moderate slope and is good for dividing signals into frequency ranges for multi-band processing. The filter is tuned so that there are no phase issues when the outputs are summed, ie., so there is no sound change.

Poly 16

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

H:

This is the audio output of this Filter, where the high-passed sound emerges.

L:

This is the audio output of this Filter, where the low-passed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through lower frequencies than the cutoff frequency at one output, and higher frequencies at the other.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

SH-1 Destroy



SH-1 Destroy is a sample-rate reduction filter with internal feedback. It has a very aggressive character which can destroy any audio signal passed through it. It works well with the crush shaper and with note following for super aggressive comb-like filtering on audio signals.

Poly 16

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies below this point will be filtered, passing through lower frequencies than the cutoff frequency, and higher frequencies will emerge from the nature of this filter.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

WA-3 Wackah



WA-3 Wackah is a wah-wah style filter modeled after a classic wah-wah pedal ideally suited to rock music when used on guitars. Its non linear frequency and phase response results in a very distinct wah-wah sound.

Poly 3

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a band of frequencies at the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

WA-5 Quack



WA-5 Quack is a wah-wah style filter modeled after a classic wah-wah pedal ideally suited to funk music when used on guitars. Its non linear frequency and phase response results in a very distinct wah-wah sound.

Poly 2

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a band of frequencies at the cutoff frequency.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

WA-8 Oohwah



WA-8 Oohwah is a wah-wah style filter modeled after a classic wah-wah pedal ideally suited to chill-out music when used on guitars. Its non linear frequency and phase response results in a very distinct wah-wah sound.

Poly 2

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

In Gain:

Since this filter has an internal saturation algorithm, you may use this controller to decrease the input of the filter and allow less saturation.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a band of frequencies at the cutoff frequency.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

XF-2 Ridge



Max. Polyphony: 2

This filter is a combination of multiple filtering components, routed and mixed together into the resonance circuitry. It achieves a new and unique filtered sound, producing two peaks and a notch at the processed result. This filter's harmonic character is more complex than standard filter types, making it ideal for evolving pad and lead sounds.

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

XF-3 Ridge



filtering other audio sources.

Max. Polyphony: 2

In:

This is the audio input of this Filter, where you connect the audio signal to be processed.

Out:

This is the audio output of this Filter, where the processed sound emerges.

Cutoff:

This function sets the cutoff frequency of the filter. Frequencies above and below this point will be filtered, passing through only a narrow band of frequencies at the cutoff frequency. Turning the knob to the left will allow lower frequencies to pass, while turning it to the right will allow higher frequencies through.

Cmod:

These are inputs for control signals (Envelope or LFO, for example) or audio rate signals (usually an Oscillator), which are used to modulate the cutoff frequency of the filter. The Mod knob adjusts the amount of modulation to be applied, and is bipolar.

Res:

This function sets the resonance amount of the filter. The resonance emphasizes a narrow peak at the cutoff frequency of the filter. Turning the knob to the right will result in more resonance.

Rmod:

This is an input for control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Resonance. The Rmod knob adjusts the amount of Resonance modulation derived from the control signal, and is bipolar.

key f:

This function sets the Keyboard Following amount. The input is usually modulated by the control signal from the Note output of an MVC module. The key f knob adjusts the amount of keyboard tracking to be applied to the filter cutoff relative to the incoming note value, and is bipolar. Turning the key f knob to the extreme right of centre will result in a full frequency tracking of the keyboard. Negative values (to the left of centre) will invert the tracking (lower note values = higher cutoff values).

This filter is a combination of multiple filtering components, routed and mixed together into the resonance circuitry. It achieves a new and unique filtered sound, producing three peaks and two notches at the processed result. This filter's harmonic character is more complex than standard filter types, making it ideal for evolving pad and lead sounds and also for

Envelopes

Hyper Envelopes

An envelope generates a varying control signal over time. Envelopes can be used to modulate any modulatable parameter on a modular, in an evolving, non-repetitive manner. By controlling the time and level parameters, you can determine the envelope's shape and how it will respond.

Envelopes enable us to imitate characteristics of real instruments such as amplitude, timbre, pitch, and other behaviors. Just connect the envelope's output to any modulatable parameter (cutoff, level, frequency, amount, and other mod inputs) to control it over time.

Once an ADSR envelope has received a gate-on signal, it will go through all of its stages until a gate-off message is received at which point it will jump to the release stage.

Envelopes generate unipolar signals, ranging from 0 to 1 according to the following controls:

Attack time:

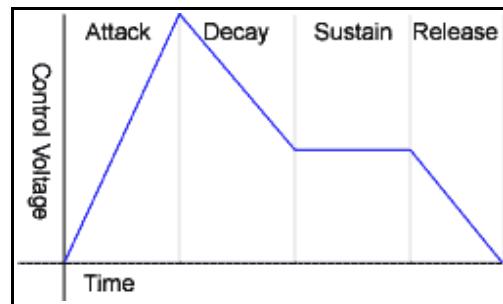
The time needed to progress from minimum (0) to maximum (1).

Decay time:

The time needed to progress from maximum to the sustain level.

Sustain Level:

The level at which the note holds after going through the attack and decay time, until a gate-off message is received.



Release Time:

The time needed to progress from the level at which the gate-off is received to minimum (0).

The above are general descriptions and can vary from envelope to envelope according to the unique features of each. Please see the Envelope reference section for detailed information on how each of FleXor's Hyper Envelopes behaves.

FleXor has two types of classic envelopes:

Hyper envelopes:

FleXor's Hyper envelopes have a unique and accurate design, making them move as fast as digitally possible. This design enables the creation of ultra fast transients, sweeps, especially useful for percussive, sequenced and bass sounds.

Vintage envelopes:

The Vintage envelopes have a unique analog like behavior, with a special curve that gives the envelopes a punchy and living character.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Hyper D



signal the envelope progresses through the decay stage. This particular envelope ignores Gate Off messages.

Poly 16

Gate:

This is an input for gate signals which are used to re-trigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Decay:

Decay time represents the amount of time it takes for its initial maximum value to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper D mod



Hyper D mod is a hyper-fast envelope with modulation inputs that is accurate down to a single sample. It is extremely punchy, and optimized for percussive sounds. The decay control changes the envelope's shape. Upon

receiving a Gate signal the envelope progresses through the decay stage. The decay can be modulated to create dynamic changes of the envelope's parameters. This particular envelope ignores Gate Off messages.

Poly 16

Gate:

This is an input for gate signals which are used to re-trigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Decay:

Decay time represents the amount of time it takes for its initial maximum value to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the decay time. The Dmod knob adjusts the amount of decay modulation derived from the control signal, and is bipolar.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Vintage AR



max) stages until a Gate Off message is received, at which point the Release stage begins.

Poly 16

Gate:

This is an input for gate signals which are used to re-trigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (sustain level). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Sustain (set on maximum).

Release:

This is the last stage of the envelope. Release time represents the amount of time it takes for the sustain value to reach minimum after a gate-off message is received. Turning the knob to the right will result in longer release times. Turning the knob to the left will make the release time shorter.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Vintage AR



Vintage AR mod is a hyper-fast envelope with modulation inputs whose curve has a vintage style and behavior. It is extremely punchy, and optimized for percussive sounds. Use the attack and release controls to

change the envelope's shape. Upon receiving a Gate signal the envelope progresses through the Attack and Sustain (at max) stages until a Gate Off message is received, at which point the Release stage begins. The attack and release can be modulated to create dynamic changes of the envelope's parameters.

Poly 16

Gate:

This is an input for gate signals which are used to re-trigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (sustain level). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Sustain (set on maximum).

Release:

This is the last stage of the envelope. Release time represents the amount of time it takes for the sustain value to reach minimum after a gate-off message is received. Turning the knob to the right will result in longer release times. Turning the knob to the left will make the release time shorter.

Amod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the attack time. The Amod knob adjusts the amount of attack modulation derived from the control signal, and is bipolar.

Rmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Release time. The Rmod knob adjusts the amount of Release modulation derived from the control signal, and is bipolar.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Vintage ADSR



Vintage ADSR is a hyper-fast ADSR envelope whose curve has a vintage style. It is extremely punchy, and optimized for percussive sounds. Use the Attack, Decay, Sustain and Release controls to change the envelope's shape. Upon receiving a Gate signal the envelope progresses through the Attack, Decay and Sustain stages until a Gate Off message is received, at which point the Release stage begins.

Poly 7

Gate:

This is an input for gate signals which are used to re-trigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using FleXor's Esync module, or an additional SonicCor envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of decay). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Decay.

Decay:

This is the second stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Attack) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Sustain:

This is the third stage of the envelope. The Sustain value controls the level at which the envelope will be held after the decay stage. Turning the knob to the right will result in higher sustain values. Turning the knob to the left will lead to lower sustain values.

Release:

This is the last stage of the envelope. Release time represents the amount of time it takes for the sustain value to reach minimum after a gate-off message is received. Turning the knob to the right will result in longer release times. Turning the knob to the left will make the release time shorter.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Vintage ADSR



Vintage ADSR mod is a hyper-fast ADSR envelope with modulation inputs whose curve has a vintage style. It is extremely punchy, and optimized for percussive sounds. Use the Attack, Decay, Sustain and Release controls to change the envelope's shape. Upon receiving a Gate signal the envelope progresses through the Attack, Decay and Sustain stages until a Gate Off message is received, at which point the Release stage begins. The controls can be

modulated to create dynamic changes of the envelope's parameters.

Poly 7

Gate:

This is an input for gate signals which are used to re-trigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using FleXor's Esync module, or an additional SonicCor envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of decay). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Decay.

Decay:

This is the second stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Attack) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Sustain:

This is the third stage of the envelope. The Sustain value controls the level at which the envelope will be held after the decay stage. Turning the knob to the right will result in higher sustain values. Turning the knob to the left will lead to lower sustain values.

Release:

This is the last stage of the envelope. Release time represents the amount of time it takes for the sustain value to reach minimum after a gate-off message is received. Turning the knob to the right will result in longer release times. Turning the knob to the left will make the release time shorter.

Amod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the attack time. The Amod knob adjusts the amount of attack modulation derived from the control signal, and is bipolar.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which

is used to modulate the decay time. The Dmod knob adjusts the amount of decay modulation derived from the control signal, and is bipolar.

Smod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Sustain level. The Smod knob adjusts the amount of Sustain modulation derived from the control signal, and is bipolar.

Rmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Release time. The Rmod knob adjusts the amount of Release modulation derived from the control signal, and is bipolar.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper AD Precision mod



decay can be modulated to create dynamic changes of the envelope's parameters.

Max. Polyphony: 12

Gate:

This is an input for gate signals, which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of decay). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the decay.

Decay:

This is the second stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post attack) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Amod:

This is an input for a control signal (usually from a sequencer module, LFO or even another envelope), which is used to modulate the attack time. The Amod knob adjusts the amount of attack modulation derived from the control signal, and is bipolar.

Dmod:

This is an input for a control signal (usually from a sequencer module or LFO), which is used to modulate the decay time. The Dmod knob adjusts the amount of decay modulation derived from the control signal, and is bipolar.

Out:

This is the control signal output of this envelope. This is where the envelope signal is generated. It may be connected to any modulation input, like filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper AD Smooth



This module is a hyper-fast envelope accurate down to individual sample level. It is extremely punchy, and optimized for percussive sounds. Use the attack and release controls to change the envelope's shape. Once a gate signal is received, the envelope will progress

through all of its stages. This particular envelope ignores gate-off messages. If a new gate signal is received while the envelope is still in progress, it will jump back to the attack stage from the same value at which it was interrupted, creating a smooth envelope transition to new notes.

Max. Polyphony: 10

Gate:

This is an input for gate signals, which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of decay). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the decay.

Decay:

This is the second stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post attack) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Out:

This is the control signal output of this envelope. This is where the envelope signal is generated. It may be connected to any modulation input, like filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper AD Smooth mod



This module is a hyper fast-envelope accurate down to one sample unit. It is extremely punchy, and optimized for percussive sounds. Use the attack and release controls to change the envelope's time. Once a gate signal is received, the envelope will progress through all of its stages. This particular envelope ignores gate-off messages. If a new gate signal is

received while the envelope is still in progress, it will jump back to the attack stage from the same value at which it was interrupted, creating a smooth envelope transition to new notes. The attack and decay can be modulated to create dynamic changes of the envelope's parameters.

Max. Polyphony: 10

Gate:

This is an input for gate signals, which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of decay). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the decay.

Decay:

This is the second stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post attack) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Amod:

This is an input for a control signal (usually from a sequencer module, LFO or even another envelope), which is used to modulate the attack time. The Amod knob adjusts the amount of attack modulation derived from the control signal, and is bipolar.

Dmod:

This is an input for a control signal (usually from a sequencer module or LFO), which is used to modulate the decay time. The Dmod knob adjusts the amount of decay modulation derived from the control signal, and is bipolar.

Out:

This is the control signal output of this envelope. This is where the envelope signal is generated. It may be connected to any modulation input, like filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper AHD Precision



This module is a hyper-fast envelope, accurate down to the individual sample level. It is extremely punchy, and optimized for percussive sounds. Use the Attack, Hold and Decay controls to change the envelope's shape. Once a gate signal is received, the envelope will progress through all of its stages. This particular envelope ignores gate-off messages.

Max. Polyphony: 12

Gate:

This is an input for gate signals which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate output of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of Hold). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Hold.

Hold:

This is the second stage of the envelope. Hold time represents the amount of time to preserve the maximum value (post-Attack). Turning the knob to the right will result in longer hold times. Turning the knob to the left will result in shorter hold times.

Decay:

This is the third stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Hold) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper AHD Precision mod



This module is a hyper-fast envelope, accurate down to the individual sample level. It is extremely punchy, and optimized for percussive sounds. Use the Attack, Hold and Decay controls to change the envelope's shape. Once a gate signal is received, the envelope will progress through all of its stages. This particular envelope ignores gate-off messages. The

Attack, Hold and Decay can be modulated to create dynamic changes of the envelope's parameters.

Max. Polyphony: 12

Gate:

This is an input for gate signals which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate output of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of Hold). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Hold.

Amod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the attack time. The Amod knob adjusts the amount of attack modulation derived from the control signal, and is bipolar.

Hold:

This is the second stage of the envelope. Hold time represents the amount of time to preserve the maximum value (post-Attack). Turning the knob to the right will result in longer hold times. Turning the knob to the left will result in shorter hold times

Hmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the hold time. The Hmod knob adjusts the amount of hold modulation derived from the control signal, and is bipolar.

Decay:

This is the third stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Hold) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the decay time. The Dmod knob adjusts the amount of decay modulation derived from the control signal, and is bipolar.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper AHD Smooth



This module is a hyper-fast envelope, accurate down to individual sample level. It is extremely punchy, and optimized for percussive sounds. Use the Attack, Hold and Decay controls to change the envelope's shape. Once a gate signal is received, the envelope will

progress through all of its stages. This particular envelope ignores gate-off messages. If a new gate signal is received while the envelope is still in progress, it will jump back to the Attack stage from the same value at which it

was interrupted, creating a smooth envelope transition to new notes.

Max. Polyphony: 10

Gate:

This is an input for gate signals which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the second stage of the envelope. Hold time represents the amount of time to preserve the maximum value (post-Attack). Turning the knob to the right will result in longer hold times. Turning the knob to the left will result in shorter hold times.

Hold:

This is the second stage of the envelope. Hold time represents the amount of time to preserve the maximum value (post-Attack). Turning the knob to the right will result in longer hold times. Turning the knob to the left will result in shorter hold times.

Decay:

This is the third stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Hold) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper AHD Smooth mod



This module is a hyper fast-envelope, accurate down to one sample unit. It is extremely punchy, and optimized for percussive sounds. Use the Attack, Hold and Decay controls to change the envelope's time. Once a gate signal is received, the envelope will progress through all of its stages. This particular envelope ignores gate-off messages. If a new gate

signal is received while the envelope is still in progress, it will jump back to the Attack stage from the same value at which it was interrupted, creating a smooth envelope transition to new notes. The Attack, Hold and Decay can be modulated to create dynamic changes of the envelope's parameters.

Max. Polyphony: 10

Gate:

This is an input for gate signals which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of Hold). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Hold.

Amod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the attack time. The Amod knob adjusts the amount of attack modulation derived from the control signal, and is bipolar.

Hold:

This is the second stage of the envelope. Hold time represents the amount of time to preserve the maximum value (post-Attack). Turning the knob to the right will result in longer hold times. Turning the knob to the left will result in shorter hold times.

Hmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the hold time. The Hmod knob adjusts the amount of hold modulation derived from the control signal, and is bipolar.

Decay:

This is the third stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Hold) to reach minimum. Turning the knob to the right will result in

longer decay times. Turning the knob to the left will result in shorter decay times.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is

used to modulate the decay time. The Dmod knob adjusts the amount of decay modulation derived from the control signal, and is bipolar.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper ADSR Precision



will begin.

Max. Polyphony: 5

Gate:

This is an input for gate signals which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of decay). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Decay.

Decay:

This is the second stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Attack) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Sustain:

This is the third stage of the envelope. The Sustain value controls the level at which the envelope will be held after the decay stage. Turning the knob to the right will result in higher sustain values. Turning the knob to the left will lead to lower sustain values.

Release:

This is the last stage of the envelope. Release time represents the amount of time it takes for the sustain value to reach minimum after a gate-off message is received. Turning the knob to the right will result in longer release times. Turning the knob to the left will make the release time shorter.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

This module is a hyper-fast ADSR envelope, accurate down to individual sample level. It is extremely punchy, and optimized for percussive sounds. Use the controls to change the envelope's shape. Once a gate signal is received, the envelope will progress through the Attack, Decay and Sustain stages, until a gate-off message is received, at which point the Release stage

Hyper ADSR Precision mod



Max. Polyphony: 5

This module is a hyper-fast ADSR envelope, accurate down to individual sample level. It is extremely punchy, and optimized for percussive sounds. Use the controls to change the envelope's shape. Once a gate signal is received, the envelope will progress through the Attack, Decay and Sustain stages, until a gate-off message is received, at which point the Release stage will begin. The controls can be modulated to create dynamic changes of the envelope's parameters.

Gate:

This is an input for gate signals which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of decay). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Decay.

Amod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the attack time. The Amod knob adjusts the amount of attack modulation derived from the control signal, and is bipolar.

Decay:

This is the second stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Attack) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the decay time. The Dmod knob adjusts the amount of decay modulation derived from the control signal, and is bipolar.

Sustain:

This is the third stage of the envelope. The Sustain value controls the level at which the envelope will be held after the decay stage. Turning the knob to the right will result in higher sustain values. Turning the knob to the left will lead to lower sustain values.

Smod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is

used to modulate the Sustain time. The Smod knob adjusts the amount of Sustain modulation derived from the control signal, and is bipolar.

Release:

This is the last stage of the envelope. Release time represents the amount of time it takes for the

sustain value to reach minimum after a gate-off message is received. Turning the knob to the right will result in longer release times. Turning the knob to the left will make the release time shorter.

Rmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is

used to modulate the Release time. The Rmod knob adjusts the amount of Release modulation derived from the control signal, and is bipolar.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be

connected to any modulation input, such as filter frequency mod controls, shaper amount controls,

VCAs and so on.

Hyper ADSR Smooth



will begin. If a new gate signal is received while the envelope is still in progress, it will jump back to the Attack stage from the same value at which it was interrupted, creating a smooth envelope transition to new notes.

Max. Polyphony: 5

Gate:

This is an input for gate signals which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the

envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack: This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of decay). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Decay.

Decay:

This is the second stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Attack) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Sustain:

This is the third stage of the envelope. The Sustain value controls the level at which the envelope will be held after the decay stage. Turning the knob to the right will result in higher sustain values. Turning the knob to the left will lead to lower sustain values.

Release:

This is the last stage of the envelope. Release time represents the amount of time it takes for the

sustain value to reach minimum after a gate-off message is received. Turning the knob to the right will result in longer release times. Turning the knob to the left will make the release time shorter.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Hyper ADSR Smooth mod



This module is a hyper-fast ADSR envelope, accurate down to individual sample level. It is extremely punchy, and optimized for percussive sounds. Use the controls to change the envelope's shape. Once a gate signal is received, the envelope will progress through the Attack, Decay and Sustain stages, until a gate-off message is received, at which point the Release stage will begin. If a new gate signal is received while the envelope is still in progress, it will jump back to the

Attack stage from the same value at which it was interrupted, creating a smooth envelope transition to new notes. The controls can be modulated to create dynamic changes of the envelope's parameters.

Max. Polyphony: 5

Gate:

This is an input for gate signals which are used to retrigger the progress of the envelope through its stages. Every time a gate-on signal (a new note, for example) is received at the gate input, the

envelope starts its progress from the attack stage. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. This particular envelope ignores gate-off messages. Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Attack:

This is the first stage from which the envelope begins its progress. Attack time represents the amount of time it takes for the initial minimum value to reach maximum (start of decay). Turning the knob to the right will result in longer attack times. When set to zero (extreme left), the envelope skips this stage and starts immediately from the next stage, which is the Decay.

Amod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the attack time. The Amod knob adjusts the amount of attack modulation derived from the control signal, and is bipolar.

Decay:

This is the second stage of the envelope. Decay time represents the amount of time it takes for its initial maximum value (post-Attack) to reach minimum. Turning the knob to the right will result in longer decay times. Turning the knob to the left will result in shorter decay times.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the decay time. The Dmod knob adjusts the amount of decay modulation derived from the control signal, and is bipolar.

Sustain:

This is the third stage of the envelope. The Sustain value controls the level at which the envelope will be held after the decay stage. Turning the knob to the right will result in higher sustain values. Turning the knob to the left will lead to lower sustain values.

Smod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Sustain level. The Smod knob adjusts the amount of Sustain modulation derived from the control signal, and is bipolar.

Release:

This is the last stage of the envelope. Release time represents the amount of time it takes for the sustain value to reach minimum after a gate-off message is received. Turning the knob to the right will result in longer release times. Turning the knob to the left will make the release time shorter.

Rmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Release time. The Rmod knob adjusts the amount of Release modulation derived from the control signal, and is bipolar.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Gate



Gate converts MVC gate signals into a classic on/off (1/0) gate signal. It is very light on DSP, and can be used together with control smoother for generating a simple envelope.

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Inv:

This is an inverse gate output. When the gate signal is on, the output generates 0 and when it is off the output generates 1(max).

Out:

This is a simple gate output. When the gate signal is on, the output generates 1(max) and when it is off the output generates 0.

Esync



Esync is a module that provides polyphony management for flexor envelopes, it processes a gate input together with an envelope input to generate an esync signal which is fed-back to the MVC. The sync signal; reports the position of each envelope to the MVC which uses this information for managing polyphony. Use this module with FleXor's amp envelope for the best results, when a SonicCore envelope is used as an amp envelope, this module is unnecessary.

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Env In:

In order to calculate your polyphony correctly

Multi Segment envelopes

These modules are not individual envelopes, but rather the building blocks for one modular envelope that can be connected in various ways to create complex envelopes. These are the separate stages of the envelope:

- **Attack:** can be used only as the first stage of an envelope. And controls the time it takes to reach from 0 (or from the release value if connected) to the next stage's level.

- **1st Step:** can be used only as the first stage of an envelope, and allows starting the envelope from a specific value each time.

- **Mid Step:** can be only used in the middle of an envelope, and can be chained endlessly together to create complex envelopes

- **Sustain:** can be only used in the middle of an envelope and sets the sustain level when connected to a previous module.

- **Loop:** This module enables a looping mechanism for the envelope and can be connected anywhere in the middle of the envelope chain.

- **Release:** can be only used at the end of an envelope. When a gate off signal is received it will sample the output of the envelope and slowly decay to minimum.

The multi segment envelopes have shortcuts in the name to distinguish them easily:

- **MSE** stands for Multi Segment Envelope – the simplest form of the envelope.
- **MSEC** stands for Multi Segment Envelope with a Curve option.
- **mod** means the envelope segment is capable of being modulated

Most of the envelope stages have the following controls:

- **Level** – The initial level at which the stage starts.
- **Time** – The time it takes to reach the next stage's initial level

Connecting the envelopes is very simple, always connect the P (previous) pad to the N (Next). It is possible to integrate other modules in the chain in many ways. Knowing what each connection does is the key to using these envelopes to the maximum:

P1:

This is an output of level information to the previous stage. This is the value which the previous stage will interpolate to. Normally it should be connected to the N1 pad of the previous module.

P2:

This is an input for sync signal – this is the way the stage knows when to kick in. It can be operated with a ramp or pulse signal. Normally it should be connected to the N2 pad of the previous module.

P3:

This is the envelope signal input – this is where the envelope signal is constructed. It is possible to feed the envelope any audio source you want to be played until the stage overrides it. Normally it should be connected to the N3 pad of the previous module.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

Important Note - For all FleXor envelopes connect the MVC's Gate output to its Esync input in order to cross-link them together, otherwise the MVC will not be activated. This process is unnecessary if you are using an additional Creamware envelope with Esync in your patch.

Here is an example patch on how to connect the MSE envelopes.



MSE 1st step



MSE 1st step is used for the first step of a Multi Segment Envelope. It can only be the first stage of a Multi Segment Envelope, and it can be swapped with the MSE Attack. Upon receiving a Gate signal this envelope stage progresses from the initial value set by the Level control to

the next stage's level (derived from input N1 which should be connected to the module that forms the next envelope stage). The speed of the progress is controlled by the Time value.

Poly 16

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Level:

This parameter sets the initial level at which the stage starts.

Time:

This parameter sets the time it takes to reach the next stage's initial level.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

MSE Attack



MSE Attack forms the Attack stage of a Multi Segment Envelope. It can only be the first stage of a Multi Segment Envelope, and it can be swapped with the MSE 1st step. Upon receiving a Gate signal this envelope stage

progresses from the release or minimum level to the next stage's level (derived from input N1 which should be connected to the module that forms the next envelope stage). The speed of the progress is controlled by the Time value.

Poly 15

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Release:

Connect the release output here for a “rising envelope” effect – the attack will start from the release position.

Time:

This parameter sets the time it takes to reach the next stage's initial level from the starting point of 0.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

MSE Loop



MSE Loop adds a looping function to the Multi Segment Envelope. Connect it before (P2) and after (N2)

the stages where you want the loop to start, and to the stage you want the loop to end (N2).
Poly 16

Last2:

Connect the N2 output of the last module you wish to be inside the loop.

P2:

This is an input for sync signal – this is the way the stage knows when to kick in. It can be operated with a ramp or pulse signal. Normally it should be connected to the N2 pad of the previous module (the module before the loop starts).

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module(first module in the loop).

MSE mid step



MSE mid step forms the middle step of a Multi Segment Envelope. It can never be the first stage of a Multi Segment Envelope. When the previous stage has finished, this stage progresses from the value set by the Level

control to the next stage's level (derived from input N1 which should be connected to the module that forms the next envelope stage). The speed of the progress is controlled by the Time value.
Poly 16

Level:

This parameter sets the initial level at which the stage starts.

Time:

This parameter sets the time it takes to reach the next stage's initial level.

P1:

This is an output of level information to the previous stage. This is the value which the previous stage will interpolate to. Normally it should be connected to the N1 pad of the previous module.

P2:

This is an input for sync signal – this is the way the stage knows when to kick in. It can be operated with a ramp or pulse signal. Normally it should be connected to the N2 pad of the previous module.

P3:

This is the envelope signal input – this is where the envelopes signal is constructed. It is possible to feed the envelope any audio source you want to be played until the stage overrides it. Normally it should be connected to the N3 pad of the previous module.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

MSE Release



controlled by the Time value.

Poly 15

MSE Release forms the Release stage of a Multi Segment Envelope. It can only be the last stage of a Multi Segment Envelope. Upon receiving a Gate Off signal this envelope stage progresses from the level of the entire envelope to minimum level. The speed of the progress is

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Time:

This parameter sets the time it takes to reach 0.

P3:

This is the envelope signal input – this is where the envelopes signal is constructed. It is possible to feed the envelope any audio source you want to be played until the stage overrides it. Normally it should be connected to the N3 pad of the previous module.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

MSE Sustain



Gate Off message is received.

Poly 16

MSE Sustain forms the sustain step of a Multi Segment Envelope. It can never be the first stage of a Multi Segment Envelope. When the previous stage has finished, this stage will remain on the Level control's value until a

Level:

This parameter sets the initial level at which the stage starts.

P1:

This is an output of level information to the previous stage. This is the value which the previous stage will interpolate to. Normally it should be connected to the N1 pad of the previous module.

MSEC 1st step



MSEC 1st step forms the first step of a Multi Segment Envelope and has a curve control. It can only be the first stage of a Multi Segment Envelope, and it can be swapped with the MSE Attack. Upon receiving a Gate signal this envelope stage progresses from the initial value set by the

Level control to the next stage's level (derived from input N1 which should be connected to the module that forms the next envelope stage). The speed of the progress is controlled by the Time value. This module also has a curve control (exp/log) for the shape of the stage.

Poly 14

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Curve:

This parameter controls the amount of curve applied to the envelope. Positive values will give a logarithmic curve while negative values will give an exponential curve. To change the value click and drag the mouse up/down.

Level:

This parameter sets the initial level at which the stage starts.

Time:

This parameter sets the time it takes to reach the next stage's initial level.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

MSEC Attack



MSEC Attack forms the Attack stage of a Multi Segment Envelope and has a curve control. It can only be the first stage of a Multi Segment Envelope, and it can be swapped with the MSE 1st step. Upon receiving a Gate signal this envelope stage progresses from the release or minimum level to the next stage's level (derived from input N1 which should be connected to the module that forms the next envelope stage). The speed of the progress is controlled by the Time value. This module also has a curve control (exp/log) for the shape of the stage.

Poly 11

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Release:

Connect the release output here for a “rising envelope” effect – the attack will start from the release position.

Curve:

This parameter controls the amount of curve applied to the envelope. Positive values will give a logarithmic curve while negative values will give an exponential curve. To change the value click and drag the mouse up/down.

Time:

This parameter sets the time it takes to reach the next stage's initial level from the starting point of 0.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

MSEC mid step



MSEC mid step forms the mid step of a Multi Segment Envelope and has a curve control. It can never be the first stage of a Multi Segment Envelope. When the previous stage has finished its progress, this stage progresses from

the Level control value to the next stage's level (derived from input N1 which should be connected to the module that forms the next envelope stage). The speed of the progress is controlled by the Time value. This module also has a curve control (exp/log) for the shape of the stage.

Poly 11

Curve:

This parameter controls the amount of curve applied to the envelope. Positive values will give a logarithmic curve while negative values will give an exponential curve. To change the value click and drag the mouse up/down.

Level:

This parameter sets the initial level at which the stage starts.

Time:

This parameter sets the time it takes to reach the next stage's initial level.

P1:

This is an output of level information to the previous stage. This is the value which the previous stage will interpolate to. Normally it should be connected to the N1 pad of the previous module.

P2:

This is an input for sync signal – this is the way the stage knows when to kick in. It can be operated with a ramp or pulse signal. Normally it should be connected to the N2 pad of the previous module.

P3:

This is the envelope signal input – this is where the envelopes signal is constructed. It is possible to feed the envelope any audio source you want to be played until the stage overrides it. Normally it should be connected to the N3 pad of the previous module.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

MSEC Release



MSEC Release forms the Release stage of a Multi Segment Envelope and has a curve control. It can only be the last stage of a Multi Segment Envelope. Upon receiving a Gate Off signal this envelope stage progresses from the level of the entire envelope to minimum level. The speed of the progress is controlled by the Time value. This module also has a curve control (exp/log) for the shape of the stage.

Poly 9

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Curve:

This parameter controls the amount of curve applied to the envelope. Positive values will give a logarithmic curve while negative values will give an exponential curve. To change the value click and drag the mouse up/down.

Time:

This parameter sets the time it takes to reach 0.

P3:

This is the envelope signal input – this is where the envelopes signal is constructed. It is possible to feed the envelope any audio source you want to be played until the stage overrides it. Normally it should be connected to the N3 pad of the previous module.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

MSE Sustain mod



MSE Sustain mod forms the sustain step of a Multi Segment Envelope and has modulation inputs. It can never be the first stage of a Multi Segment Envelope. When the previous stage has finished its progress, this stage will remain on the value set by the Level control until a Gate Off message is received. The Sustain level can be modulated.

Poly 16

Level:

This parameter sets the initial level at which the stage starts.

Lmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Level of the stage. The Lmod knob adjusts the amount of Level modulation derived from the control signal, and is bipolar.

P1:

This is an output of level information to the previous stage. This is the value which the previous stage will interpolate to. Normally it should be connected to the N1 pad of the previous module.

MSEC 1st step mod



MSEC 1st step mod forms the first step of a Multi Segment Envelope, and has a curve control plus modulation inputs. It can only be the first stage of a Multi Segment Envelope, and it can be swapped with the MSE Attack. Upon receiving a Gate signal this envelope stage progresses from the initial value set by the Level control to the next stage's level (derived from input N1 which should

be connected to the module that forms the next envelope stage). The speed of the progress is controlled by the Time value. This module also has a curve control (exp/log) for the shape of the stage. The Level and Time parameters can be modulated.

Poly 14

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Curve:

This parameter controls the amount of curve applied to the envelope. Positive values will give a logarithmic curve while negative values will give an exponential curve. To change the value click and drag the mouse up/down.

Level:

This parameter sets the initial level at which the stage starts.

Time:

This parameter sets the time it takes to reach the next stage's initial level.

Lmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Level of the stage. The Lmod knob adjusts the amount of Level modulation derived from the control signal, and is bipolar.

Tmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Time of the stage. The Tmod knob adjusts the amount of Time modulation derived from the control signal, and is bipolar.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

MSEC Attack mod



MSEC Attack mod forms the Attack stage of a Multi Segment Envelope, and has a curve control plus modulation inputs. It can only be the first stage of a Multi Segment Envelope, and it can be swapped with the MSE

1st step. Upon receiving a Gate signal this envelope stage progresses from the release or minimum level to the next stage's level (derived from input N1 which should be connected to the module that forms the next envelope stage). The speed of the progress is controlled by the Time value. This module also has a curve control (exp/log) for the shape of the stage. The Time parameter can be modulated.

Poly 11

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Release:

Connect the release output here for a “rising envelope” effect – the attack will start from the release position.

Curve:

This parameter controls the amount of curve applied to the envelope. Positive values will give a logarithmic curve while negative values will give an exponential curve. To change the value click and drag the mouse up/down.

Time:

This parameter sets the time it takes to reach the next stage's initial level from the starting point of 0.

Tmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Time of the stage. The Tmod knob adjusts the amount of Time modulation derived from the control signal, and is bipolar.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

MSEC mid step mod



MSEC mid step mod forms the mid step of a Multi Segment Envelope, and has a curve control plus modulation inputs. It can never be the first stage of a Multi Segment Envelope. When the previous stage has finished its progress, this current stage progresses from the value set by the Level control to the next stage's level (derived from input N1 which should be connected to the module

that forms the next envelope stage). The speed of the progress is controlled by the Time value. This module also has a curve control (exp/log) for the shape of the stage. The Level and Time parameters can be modulated.

Poly 11

Curve:

This parameter controls the amount of curve applied to the envelope. Positive values will give a logarithmic curve while negative values will give an exponential curve. To change the value click and drag the mouse up/down.

Level:

This parameter sets the initial level at which the stage starts.

Time:

This parameter sets the time it takes to reach the next stage's initial level.

Lmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Level of the stage. The Lmod knob adjusts the amount of Level modulation derived from the control signal, and is bipolar.

Tmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Time of the stage. The Tmod knob adjusts the amount of Time modulation derived from the control signal, and is bipolar.

P1:

This is an output of level information to the previous stage. This is the value which the previous stage will interpolate to. Normally it should be connected to the N1 pad of the previous module.

P2:

This is an input for sync signal – this is the way the stage knows when to kick in. It can be operated with a ramp or pulse signal. Normally it should be connected to the N2 pad of the previous module.

P3:

This is the envelope signal input – this is where the envelopes signal is constructed. It is possible to feed the envelope any audio source you want to be played until the stage overrides it. Normally it should be connected to the N3 pad of the previous module.

N1:

This is an input for the next stage level. This is the value which the current stage will interpolate to. It can be a sustain module, another envelope, an LFO or any other modulation source. Normally it should be connected to the P1 pad of the next module.

N2:

This is an output of sync signal – the sync signal is a progressing ramp with the duration of the stage. This output can be used as a modulator as well. Normally it should be connected to the P2 pad of the next module.

N3:

This is the output of the envelope itself – this is where the envelopes signal is constructed. It is possible to use this as an envelope output. Normally it would connect to the P3 pad of the next module.

MSEC Release mod



MSEC Release mod forms the Release stage of a Multi Segment Envelope, and has a curve control plus modulation inputs. It can only be the last stage of a Multi Segment Envelope. Upon receiving a Gate Off message

this envelope stage progresses from the level of the entire envelope to minimum level. The speed of the progress is controlled by the Time value. This module also has a curve control (exp/log) for the shape of the stage. The Time parameter can be modulated.

Poly 9

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module.

Curve:

This parameter controls the amount of curve applied to the envelope. Positive values will give a logarithmic curve while negative values will give an exponential curve. To change the value click and drag the mouse up/down.

Time:

This parameter sets the time it takes to reach 0.

Tmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Time of the stage. The Tmod knob adjusts the amount of Time modulation derived from the control signal, and is bipolar.

P3:

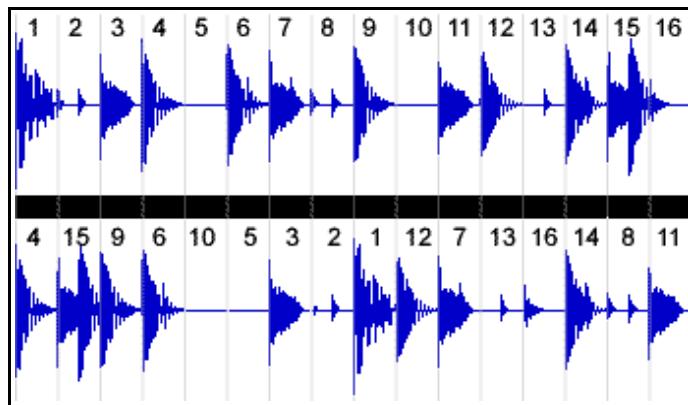
This is the envelope signal input – this is where the envelopes signal is constructed. It is possible to feed the envelope any audio source you want to be played until the stage overrides it. Normally it should be connected to the N3 pad of the previous module.

Out:

This is the control signal output of the envelope, where the envelope signal is generated. It may be connected to any modulation input, such as filter frequency mod controls, shaper amount controls, VCAs and so on.

Granular

Granular processing is a simple and versatile way of synthesizing and even composing. The term “granular” refers to ‘Grains’: small particles of a whole sound, which can be reorganized and processed individually to create totally new sounds.



This can result in extreme special effects such as pitch-shifting, time-stretching, reversing, freezing, scrambling, formant shifting, re-phrasing and utter deconstruction of the sound.

FleXor's granular section contains simple and basic granular modules.

Displacer and ReGroover:

These processors will cut your audio into 16 steps, allowing you to reorder those steps in real-time for dynamic time-stretching and scrambling effects.

Granular sequencer and Inter Sequencer:

These processors allow you to construct a single signal with grains coming from various sources. They are able to cut up to 16 different inputs into grains and sequence them into one sound.

Pitcher:

This processor will cut your audio into grains of a definable size, and pitch-shift each of them individually, without damaging the placement of each grain. This allows to maintain the same length while pitch shifting.

Granular Delay:

This processor is a granular delay time. It is modulated using a granular method. It allows extreme time stretching, as well as audio scrambling and exotic delay effects.

Override Delay:

This processor is a delay with feedback and tempo control, and gate, allowing to instantly loop any incoming audio. It can be used to create granular beat slicing and rearranging of grooves.

With these modules, you are able to construct a vast array of complex granular processing devices for use on samples, synths, drums, vocals, and even entire mixes.

Displacer



The Displacer is a granular processing module. It slices the incoming audio signal into 16 steps, and plays them at any desirable order. Consecutive playing order is achieved with a FleXor Ramp module connected to the Ramp input. Different playing orders can be achieved with other control signals, such as LFOs and step sequencers. Each step has its own fader to adjust its relative playback start position. The Amount control adjusts the relative playback start position of all steps together and the Fade control adjusts the amount of volume cross-fading between the steps.

Monophonic.

In:

This is the audio input of this module, where you connect the audio signal to be processed.

Faders 1-16:

These faders set the delay time for each step. Delay range is from 0 to 5461 milliseconds (when Amount is set to max). The delay time value is displayed on the screen window below the faders.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the Displacer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO waveshape cycle.

Amount:

This function sets delay time globally for all steps, while preserving their relative differences, as indicated by their fader position. It effectively sets the range of the delay time possible with each fader.

Fade:

This function sets the amount of cross-fading between the steps. With no cross-fade (extreme left), there is an immediate “jump” between the steps during progress of playback. Adding cross-fade helps to smooth the transitions between the steps.

Amod:

This is an input for control signal (usually a constant value or sample-and-hold), which is used to modulate the Amount function. The Amod knob adjusts the amount of Amount modulation, and is bipolar. The Amod control is async, meaning it will crackle if modulated with changing dc. Using sample and hold or a bit-crusher, can help reducing those crackles.

Fmod:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the Fade function. The Fmod knob adjusts the amount of Fade modulation, and is bipolar.

Manage Patterns:

This switch manages patterns by opening the preset window, allowing saving and loading of pattern presets. The preset bank should be indexed as bank 0 with the presets inside auto-indexed (right-click on the preset numbers) in order for this function to work.

Pattern Select:

On the Pattern Select display, click and drag up/down to scroll thru the preset patterns. The preset bank should be indexed as bank 0 with the presets inside auto-indexed (right-click on the preset numbers) in order for this function to work.

Ext. (pattern select):

The External Pattern Select is a control signal input, used for scrolling through the preset patterns in the bank selected with the Manage Patterns function. A ramp generator such as saw-up LFO (the FleXor Ramp module is the most recommended) will make the Displacer scroll through its preset patterns one-after-another as numbered. Other LFO shapes will scroll through the preset patterns in various orders, according to the amplitude changes in the LFO waveshape cycle.

No. of patterns:

On the Number of Pattern display, click and drag up/down to set the number of patterns to be scrolled either by control signal modulating the Ext input, or by clicking and dragging at the Pattern Select text-fader.

Draw:

Enabling the Draw switch will activate the Virtual pencil. This tool enables to freely draw the position of faders in one mouse movement. Click the pencil and drag it to draw your new displacer sequence.

Out:

This is the audio output of the module, where the processed sound emerges.

ReGroover

The ReGroover is a granular processing module. It slices the incoming audio signal into 16 steps, and plays them at any desirable order. Consecutive playing order is achieved with a FleXor Ramp module connected to the Ramp input. Different playing orders can be achieved with other control signals, such as LFOs and step sequencers. The Amount control adjusts the relative playback start position of all steps together and the Fade control adjusts the amount of volume cross-fading between the steps.

Monophonic.**In:**

This is the audio input of this module, where you connect the audio signal to be processed.

Ramp:

This is an input for control signal, which dictates the playing order of 16 steps. A ramp generator or a saw-up LFO will make the ReGroover play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps at various orders, depending on the amplitude changes in the LFO waveshape cycle.

Amount:

This function sets delay time globally for all steps, while preserving their relative differences. Turning to the right will create a longer delay between each step.

Fade:

This function sets the amount of cross-fading between the steps. With no cross-fade (extreme left), there is an immediate “jump” between the steps during progress of playback. Adding cross-fade helps to smooth the transitions between the steps.

Amod:

This is an input for a control signal (usually a constant value or sample-and-hold), which is used

to modulate the Amount function. The Amod knob adjusts the degree to which the incoming control signal modulates the Amount function. The Amod control is async, meaning it will crackle if modulated with changing DC. Using a sample-and-hold or bit-crusher can help to reduce these crackles.

Fmod:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the Fade function. The Fmod knob adjusts the amount of Fade modulation, and is bipolar.

Out:

This is the audio output of this module, where the processed sound emerges.

Granular Sequencer



control adjusts the amount of cross-fading between the steps.

Monophonic.

This module plays 16 different audio inputs in a sequence, at any desirable order. Up to 16 different sound sources can be connected to this module. Consecutive playing order is achieved with a FleXor Ramp module connected to the Ramp input. Different playing orders can be achieved with other control signals, such as LFOs and step sequencers. The Fade

Inputs 1-16:

These are the inputs for the audio signals to be processed by the Granular Sequencer.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps at various orders, according to the amplitude changes in the LFO waveshape cycle.

Fade:

This function sets the amount of cross-fading between the steps. With no cross-fade (extreme left), there is an immediate “jump” between the steps during progress of playback. Adding cross-fade helps to smooth the transitions between the steps.

mod:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the Fade function. The Fmod knob adjusts the amount of Fade modulation, and is bipolar.

Out:

This is the audio output of this module, where the processed sound emerges.

Pitcher Control



This module generates control signals, optimized especially for the FleXor Pitcher Engine modules. The "C" outputs of the module need to be connected to the corresponding "C" inputs on the Pitcher Engine modules, in order for it to work. The Speed controller adjusts grain size, while Amount will adjust the amount of pitch-shifting. Use smaller grains for more extreme

degrees of pitch-shifting.

Monophonic.

Speed:

This function sets the grain size of the pitch shifter, and the speed of transition between grains; as the grains become smaller (turn the knob towards the right) the amount of pitch-shifting possible increases.

Amount:

This function sets the amount of pitch shifting applied. Its range depends on the speed parameter. Turning the knob to the right of centre will shift the pitch up, while turning it left will shift it down.

Smod:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the Speed function on the incoming signal. The Smod knob adjusts the amount of Speed modulation, and is bipolar.

Amod:

This is an input for a control signal (usually an Envelope or LFO), which is used to modulate the Amount function on the incoming signal. The Amod knob adjusts the degree of modulation of the Amount function, and is bipolar.

C 1-4 outputs:

Connect these outputs to the Pitcher Engine C inputs. These signals link the Pitcher Control to the Pitcher Engine, allowing control of many engines with one Pitcher Control.

Ext. Freq :

This is an input for an external frequency generator such as an MVC module, Pitch2Cv (for real-time formant shifting!) or Hyper Note Sequencer, with which you can externally control the speed value. When this input is connected, the Pitcher Control Speed knob is disabled.

Pitcher Engine



pitch shifter makes it ideal for special effects, including turntable-esque scratch effects.

Monophonic.

This module, combined with the Pitcher Control module, forms an extreme granular pitch shifting effect. In order for it to work, the "C" inputs of the module need to be connected to the corresponding "C" outputs of a Pitcher Control module. The extreme range of this

In:

This is the audio input of this module, where you connect the audio signal to be processed.

C 1-4 inputs:

Connect these inputs to the corresponding Pitcher Control C outputs. These signals link the Pitcher Control to the Pitcher Engine, allowing the control of many engines with one Pitcher Control.

Out:

This is the audio output of this module, where the processed sound emerges.

Pitcher Engine FB



Monophonic.

This module, combined with the Pitcher Control module, forms an extreme granular pitch shifting effect. In order for it to work, the “C” inputs of the module need to be connected to the corresponding “C” outputs of a Pitcher Control module. The feedback delay in this module enables the creation of various sci-fi style effects.

In:

This is the audio input of this module, where you connect the audio signal to be processed.

Feedback:

This function sends an adjustable amount of the signal from the module's output back into its input. The feedback chain will cause the audio signal to be transposed over and over again, producing the classic sci-fi pitch shifting effect. Turning the knob to the right will result in positive feedback, while turning it to the left leads to negative feedback.

mods:

These are inputs for control signals (usually an Envelope or LFO), which are used to modulate the feedback function on the incoming signal. The Mod knobs adjust the amount of Feedback modulation, and are bipolar.

Delay:

This function sets the delay time for the feedback audio return.

C 1-4 inputs:

Connect these inputs to the corresponding Pitcher Control C outputs. These signals link the Pitcher Control to the Pitcher Engine, allowing the control of many engines with one Pitcher Control.

Out:

This is the audio output of this module, where the processed sound emerges.

Interseq 16



Interseq 16 cross-fades between 16 different audio inputs in a sequence, in any desired order. Up to 16 different sound sources can be connected to this module. Consecutive playing order is achieved with a FleXor Ramp

module connected to the Ramp input. Different playing orders can be achieved with other control signals, such as LFOs and step sequencers. The cross-fades between the steps are interpolated and are calculated from the position of the ramp or control signal.

Ins:

These are the audio inputs of the module. Each input will be active according to the Ramp state.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO wave-shape cycle.

Out:

This is the output of the module, where the processed signal emerges.

InterSeq 8



InterSeq 8 cross-fades between 8 different audio inputs in a sequence, in any desired order. Up to 8 different sound sources can be connected to this module. Consecutive playing order is achieved with a FleXor Ramp

module connected to the Ramp input. Different playing orders can be achieved with other control signals, such as LFOs and step sequencers. The cross-fades between the steps are interpolated and are calculated from the position of the ramp or control signal.

Ins:

These are the audio inputs of the module. Each input will be active according to the Ramp state.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO wave-shape cycle.

Out:

This is the output of the module, where the processed signal emerges.

InterSeq 4



InterSeq 4 cross-fades between 4 different audio inputs in a sequence, in any desired order. Up to 4 different sound sources can be connected to this module.

Consecutive playing order is achieved with a FleXor Ramp module connected to the Ramp input. Different playing orders can be achieved with other control signals, such as LFOs and step sequencers. The cross-fades between the steps are interpolated and are calculated from the position of the ramp or control signal.

Ins:

These are the audio inputs of the module. Each input will be active according to the Ramp state.

Ramp:

This is an input for a control signal, which dictates the playing order of 16 steps. A ramp generator such as the Ramp module or a saw-up LFO will make the sequencer play its steps one-after-another as numbered (16 steps per ramp cycle). Other LFO shapes will play the steps in various orders, according to the amplitude changes in the LFO wave-shape cycle.

Out:

This is the output of the module, where the processed signal emerges.

Granular Delay



Granular Delay is a delay module that has granular properties when modulated. It delays each grain separately and allows modulation of delay time with granular artifacts. It is ideal for slowing audio or speeding it up, and with a buffer length of 5 seconds it is capable of creating extreme granular chaos.

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

Delay:

This parameter sets the delay time of the module. Turning it to the right will result in longer delay times.

Grain Size:

This parameter sets the length of the individual sound grains. Turning it to the right will result in larger grain size.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar. Use this input to achieve the desired granular effect.

Gmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time of the stage. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar.

Out:

This is the output of the module, where the processed signal emerges.

Override Delay



Override Delay is a delay module that overrides the input with a looped version of itself when gated. It is possible to control the feedback amount with the Sustain control (left-0 middle-1 right-2), and the length of the cross-fade at the looping point with the Fade control. The catch option selects if the override kicks in when a Gate On message is received (catch on), or waits one loop cycle

after the Gate On message (catch off).

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

Gate:

This is an input for gate signals. Usually, gate signals will come from the gate out of an MVC module, or from a sequencer module. When a gate On message is received the delay will start looping the incoming audio.

Catch:

The catch option selects if the override kicks in when a Gate On message is received (catch off), or waits one loop cycle after the Gate On message (catch on).

BPM:

This parameter sets the Time of the LFO in BPM.

Div:

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds:

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Fade:

This parameter controls the time it takes to fade into the looping audio. Turn it to the right for more Fade time.

Sustain:

This parameter controls the loop's sustain. At its center point the loop will repeat itself endlessly. Turn it to the left to decay the loop, and to the right to gradually amplify it.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Fade parameter. The Fmod knob adjusts the amount of Fade modulation derived from the control signal, and is bipolar.

Smod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Sustain parameter. The Smod knob adjusts the amount of Sustain modulation derived from the control signal, and is bipolar.

Out:

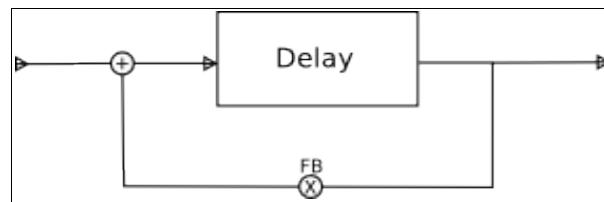
This is the output of the module, where the processed signal emerges.

Delays

The simple delay is a processor that stores an input signal to memory, and then plays it back after a period of time. Most delays have a feedback loop, which creates a repeating effect, by feeding the output of the delay back into its input. Delays are very useful for creating spatial effects and are very common in studio use.

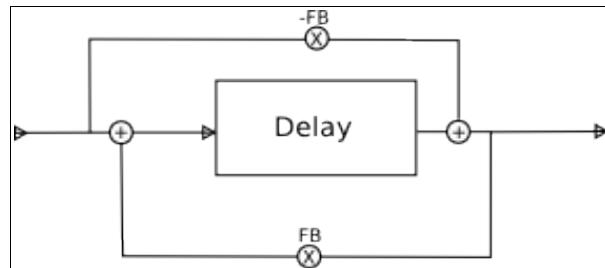
Delays are the basic building block of reverbs, and in Flexor they come in two main topologies:

Comb Delay:



This is the classic form of a delay where the signal is fed back from the output and mixed with the incoming signal at the input.

All-pass Delay:



This is a common building block from reverbs. The original signal is mixed with the delayed signal and only then it is fed back mixed again with the incoming signal at the input. This creates a more diffused character to the delay, making it great for adding space to a track.

The delays come in two categories:

Time based: where it is possible to tune the time of the delays in milliseconds.

Tempo Based: where it is possible to tune the time of the delays using BPM and division controls.

Delay



maximum delay time is 5 seconds.

Flexor's **Delay** module is a delay with time control and feedback. The time of the delay is set by the Delay control, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). It is ideal for use as an effect and for adding spatial properties to any sound source. The

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

Delay:

This parameter sets the delay time of the module. Turning it to the right will result in longer delay times.

Feedback:

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

ULLI:

In order for this module to work, please tune this controller to your hardware's ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Out:

This is the output of the module, where the processed signal emerges.

Delay F



Delay F is a delay with time control and filtered feedback. The time of the delay is set by the Delay control, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). This module has a high-pass filter and a low-pass filter in the feedback path for gradual damping.

damping. It is ideal for use as an effect and for adding spatial properties to any sound source. The maximum delay time is 5 seconds.

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

Delay:

This parameter sets the delay time of the module. Turning it to the right will result in longer delay times.

Feedback:

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

HP:

Adjusts a high-pass filter cutoff frequency. The high-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the low frequencies of the signal.

LP:

Adjusts a low-pass filter cutoff frequency. The low-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the high frequencies of the signal.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Out:

This is the output of the module, where the processed signal emerges.

Delay S



Delay S is a stereo delay with time control and feedback. The time of the delay is set by the Delay control, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). Each channel has two feedback controls: one for self feedback and one for cross feedback. It is ideal for use as an effect and for adding spatial properties to any sound source. The maximum delay time is 5 seconds.

Ins:

These are the audio inputs of this delay module, where you connect the audio signal to be processed.

Delay (L/R):

These parameters set the delay time for each side (left and right). Turning it to the right will result in longer delay times.

Feedback (L/R):

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate. Each channel sends its own signal both to itself and to the other channel. FB L knobs set the feedback to the left channel, and FB R knobs set the feedback to the right channel.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

ULLI:

In order for this module to work, please tune this controller to your hardware's ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Outs:

These are the outputs of the module, where the processed signal emerges.

Delay SF



source. The maximum delay time is 5 seconds.

Delay SF is a stereo delay with time control and feedback. The time of the delay is set by the Delay control, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). Each channel has two feedback controls: one for self feedback and one for cross feedback. This module has a high-pass filter and a low-pass filter in the feedback paths for gradual damping. It is ideal for use as an effect and for adding spatial properties to any sound

Ins:

These are the audio inputs of this delay module, where you connect the audio signal to be processed.

Delay (L/R):

These parameters set the delay time for each side (left and right). Turning it to the right will result in longer delay times.

Feedback (L/R):

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate. Each channel sends its own signal both to itself and to the other channel. FB L knobs set the feedback to the left channel, and FB R knobs set the feedback to the right channel.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

HP:

Adjusts a high-pass filter cutoff frequency. The high-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the low frequencies of the signal.

LP:

Adjusts a low-pass filter cutoff frequency. The low-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the high frequencies of the signal.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

ULLI:

In order for this module to work, please tune this controller to your hardware's ULLI state (as

listed under 44.1khz on the ULLI menu in scope).

Outs:

These are the outputs of the module, where the processed signal emerges.

Tempo Delay



Tempo Delay is a delay with tempo control and feedback. The time of the delay is set by the BPM and division controls and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). It is ideal for use as an effect and for adding spatial properties to any sound source. The maximum delay time is 5 seconds.

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

BPM:

This parameter sets the Time of the LFO in BPM.

Div:

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds:

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Feedback:

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect) . At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Out:

This is the output of the module, where the processed signal emerges.

Tempo Delay F



Tempo Delay F is a delay with tempo control and feedback. The time of the delay is set by the BPM and division controls, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). This module has a high-pass filter and a low-pass filter in the feedback path for gradual damping. It is ideal for use as an effect and for adding spatial properties to any sound source. The maximum delay time is 5 seconds.

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

BPM:

This parameter sets the Time of the LFO in BPM.

Div:

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds:

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardware's ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Feedback:

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

HP:

Adjusts a high-pass filter cutoff frequency. The high-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the low frequencies of the signal.

LP:

Adjusts a low-pass filter cutoff frequency. The low-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the high frequencies of the signal.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Out:

This is the output of the module, where the processed signal emerges.

Tempo Delay S



Tempo Delay S is a stereo delay with tempo control and feedback. The time of the delay is set by the BPM and division controls, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). Each channel has two feedback controls: one for self feedback and one for cross feedback. It is ideal for use as an effect and for adding spatial properties to any sound source. The maximum delay time is 5 seconds.

Ins:

These are the audio inputs of this delay module, where you connect the audio signal to be processed.

BPM:

This parameter sets the Time of the LFO in BPM.

Div (L/R):

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds(L/R):

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardware's ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Feedback (L/R):

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate. Each channel sends its own signal both to itself and to the other channel. FB L knobs set the feedback to the left channel, and FB R knobs set the feedback to the right channel.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Outs:

These are the outputs of the module, where the processed signal emerges.

Tempo Delay SF



Tempo Delay SF is a stereo delay with tempo control and filtered feedback. The time of the delay is set by the BPM and division controls, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). Each channel has two feedback controls: one for self feedback and one for cross feedback. This module has a high-pass filter and a low-pass filter in the feedback paths for gradual damping. It is ideal for use as an effect and for adding spatial

properties to any sound source. The maximum delay time is 5 seconds.

Ins:

These are the audio inputs of this delay module, where you connect the audio signal to be processed.

BPM:

This parameter sets the Time of the LFO in BPM.

Div (L/R):

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds(L/R):

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Feedback (L/R):

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate. Each channel sends its own signal both to itself and to the other channel. FB L knobs set the feedback to the left channel, and FB R knobs set the feedback to the right channel.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

HP:

Adjusts a high-pass filter cutoff frequency. The high-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the low frequencies of the signal.

LP:

Adjusts a low-pass filter cutoff frequency. The low-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the high frequencies of the signal.

Outs:

These are the outputs of the module, where the processed signal emerges.

All-Pass

All-Pass



maximum delay time is 5 seconds.

All-Pass is an all-pass delay with time control and feedback. All-pass delays transfer all frequencies equally, which makes them ideal for reverb construction. The time of the delay is set by the Delay control, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). The

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

Delay:

This parameter sets the delay time of the module. Turning it to the right will result in longer delay times.

Feedback:

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Out:

This is the output of the module, where the processed signal emerges.

All-Pass F



All-Pass F is an all-pass delay with time control and filtered feedback. All-pass delays transfer all frequencies equally, which makes them ideal for reverb construction. The time of the delay is set by the Delay control, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). This module has a high-pass filter and a low-pass filter in the feedback path for gradual damping. The maximum delay time is 5 seconds.

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

Delay:

This parameter sets the delay time of the module. Turning it to the right will result in longer delay times.

Feedback:

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time . The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

HP:

Adjusts a high-pass filter cutoff frequency. The high-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the low frequencies of the signal.

LP:

Adjusts a low-pass filter cutoff frequency. The low-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the high frequencies of the signal.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect) . At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Out:

This is the output of the module, where the processed signal emerges.

All-Pass S



is 5 seconds.

All-Pass S is a stereo all-pass delay with time control and feedback. All-pass delays transfer all frequencies equally, which makes them ideal for reverb construction. The time of the delay is set by the Delay controls, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). Each channel has its own feedback control, plus there is a dedicated control for the amount of cross feedback between the channels. The maximum delay time

Ins:

These are the audio inputs of this delay module, where you connect the audio signal to be processed.

Delay (L/R):

These parameter set the delay time for each side (left and right). Turning it to the right will result in longer delay times.

Feedback (L / R / Cross):

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate. Each channel sends its own signal both to itself and to the other channel. FB L knobs set the feedback to the left channel, and FB R knobs set the feedback to the right channel. Cross FB sends each of the channel to its opposite channel.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect) . At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Outs:

These are the outputs of the module, where the processed signal emerges.

All-Pass SF



module also has a high-pass filter and a low-pass filter in the feedback paths for gradual damping. The maximum delay time is 5 seconds.

Ins:

These are the audio inputs of this delay module, where you connect the audio signal to be processed.

Delay (L/R):

These parameter set the delay time for each side (left and right). Turning it to the right will result in longer delay times.

Feedback (L / R / Cross):

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate. Each channel sends its own signal both to itself and to the other channel. FB L knobs set the feedback to the left channel, and FB R knobs set the feedback to the right channel. Cross FB sends each of the channel to its opposite channel.

Dmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Delay Time. The Dmod knob adjusts the amount of Delay modulation derived from the control signal, and is bipolar.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

HP:

Adjusts a high-pass filter cutoff frequency. The high-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the low frequencies of the signal.

LP:

Adjusts a low-pass filter cutoff frequency. The low-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the high frequencies of the signal.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as

listed under 44.1khz on the ULLI menu in scope).

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Outs:

These are the outputs of the module, where the processed signal emerges.

Tempo All-Pass



Tempo All-Pass is an all-pass delay with tempo control and feedback. All-pass delays transfer all frequencies equally, which makes them ideal for reverb construction. The time of the delay is set by the BPM and division controls, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). The maximum delay time is 5 seconds.

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

BPM:

This parameter sets the Time of the LFO in BPM.

Div:

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds:

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardware's ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Feedback:

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Out:

This is the output of the module, where the processed signal emerges.

Tempo All-Pass F



Tempo All-Pass F is an all-pass delay with tempo control and filtered feedback. All-pass delays transfer all frequencies equally, which makes them ideal for reverb construction. The time of the delay is set by the BPM and division controls, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). This module has a high-pass filter and a low-pass filter in the feedback path for gradual damping. The maximum delay time is 5 seconds.

In:

This is the audio input of this delay module, where you connect the audio signal to be processed.

BPM:

This parameter sets the Time of the LFO in BPM.

Div:

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds:

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Feedback:

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate.

Fmod:

This is an input for a control signal (usually from a sequencer module, envelope or LFO), which is used to modulate the Feedback of the delay. The Fmod knob adjusts the amount of Feedback modulation derived from the control signal, and is bipolar.

HP:

Adjusts a high-pass filter cutoff frequency. The high-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the low frequencies of the signal.

LP:

Adjusts a low-pass filter cutoff frequency. The low-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the high frequencies of the signal.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Out:

This is the output of the module, where the processed signal emerges.

Tempo All-Pass S



Tempo All-Pass S is an all-pass stereo delay with tempo control and feedback. All-pass delays transfer all frequencies equally, which makes them ideal for reverb construction. The time of the delay is set by the BPM and division controls, and the feedback can be set from negative feedback (extreme left) through no feedback (middle) to positive feedback (extreme right). Each channel has its own feedback control, plus there is a dedicated control for the amount of cross feedback between the channels. The maximum delay time is 5 seconds.

Ins:

These are the audio inputs of this delay module, where you connect the audio signal to be processed.

BPM:

This parameter sets the Time of the LFO in BPM.

Div (L/R):

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds(L/R):

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardware's ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Feedback (L / R / Cross):

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate. Each channel sends its own signal both to itself and to the other channel. FB L knobs set the feedback to the left channel, and FB R knobs set the feedback to the right channel. Cross FB sends each of the channel to its opposite channel.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Outs:

These are the outputs of the module, where the processed signal emerges.

Tempo All-Pass SF



channels. This module has a high-pass filter and a low-pass filter in the feedback paths for gradual damping. The maximum delay time is 5 seconds.

Ins:

These are the audio inputs of this delay module, where you connect the audio signal to be processed.

BPM:

This parameter sets the Time of the LFO in BPM.

Div (L/R):

Clicking and dragging the Div control will set the clock division of the BPM, dividing it to beat values such as 4/4, 1/2, 1/4, 5/8, 3/16 and so on.

Milliseconds(L/R):

Shows the time of the LFO in milliseconds.

ULLI:

In order for this module to work, please tune this controller to your hardwares ULLI state (as listed under 44.1khz on the ULLI menu in scope).

Feedback (L / R / Cross):

This parameter sets the amount of internal feedback in the delay, and is bipolar. Turning this knob to the left will result in more negative feedback, while turning it to the right will result in positive feedback. Setting the knob to the middle will result in no feedback. The more feedback the module has the more repetitions the delay will generate. Each channel sends its own signal both to itself and to the other channel. FB L knobs set the feedback to the left channel, and FB R knobs set the feedback to the right channel. Cross FB sends each of the channel to its opposite channel.

Dry/Wet:

This parameter sets the ratio between the dry signal (original) and the wet signal (the processed effect). At the minimum value only the dry signal is heard, while at the maximum only the wet signal is heard.

HP:

Adjusts a high-pass filter cutoff frequency. The high-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the low frequencies of the signal.

LP:

Adjusts a low-pass filter cutoff frequency. The low-pass filter is inside the feedback loop and thus its effect is recursive. It can be used for damping the high frequencies of the signal.

Inv:

When on, the wet signal is inverted when mixed with the dry signal.

Outs:

These are the outputs of the module, where the processed signal emerges.

Extra

Sample Delay



Monophonic.

This is a delay module, adjustable at sample level for high-precision time control. The delay ranges from 0 to 2000 samples. It is especially useful in making fixed-phase alignments and pseudo-stereo effects.

In:

This is the audio input of the module, where you insert any signal you want to delay.

Out:

This is the audio output of this module, where the delayed signal emerges.

Phase Modulation



This module allows FM-like (Phase Modulation) processing on a live input. It modulates the incoming signal's pitch according to the modulation input's ballistic behavior (speed and direction of movement). It is especially good for creating FM-like effects on sample oscillators or live inputs, modulated with oscillators or even other audio material.

Max. Polyphony: 16

In:

This is the module's audio input, where you connect audio signals to be processed.

mods:

These are inputs for control signals (usually an envelope, LFO or oscillator) which are used to modulate the incoming signal's phase. The Mod knobs adjust the amount of phase modulation derived from the control signal, and are bipolar.

Out:

This is the module's audio output, where the processed signal emerges.

Double Sided Phase Modulation



like effects on sample oscillators or live inputs, modulated with oscillators or even other audio material.

Max. Polyphony: 9

This module allows rich FM-like (Double Side Phase Modulation) processing on a live input. It modulates the incoming signal's pitch according to the modulation input's ballistic behavior (speed and direction of movement). It is especially good for creating dense FM-

In:

This is the module's audio input, where you connect audio signals to be processed.

mods:

These are inputs for control signals (usually an envelope, LFO or oscillator) which are used to modulate the incoming signal's phase. The Mod knobs adjust the amount of phase modulation derived from the control signal, and are bipolar.

Out:

This is the module's audio output, where the processed signal emerges.