snapin

OPERATOR'S MANUAL

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Introduction

Snapins are isolated audio effect modules, each performing a specific task. The Kilohearts product lineup will allow you to combine these snapins in different ways in order to create interesting and unique combinations of effects. Snapins can be loaded into your DAW as a standard VST or Audio Unit plugin, as well as into other Kilohearts plugins that can act as snapin hosts, for example Multipass.

System requirements

These are the minimum recommended system requirements for running snapins.

CPU
    2 GHz or faster

Memory
    1 GB or more

Operating System
    No matter if you are a Windows or Mac user, Kilohearts recommend keeping up to date with OS versions as they become available. We are not able to support OS version that are no longer supported by their respective manufacturers. If you are using an outdated OS please try our free trials to ensure compatibility before making any purchases.

Software
    A DAW supporting VST 2, AAX, or Audio Unit plugin standards.

If you use a lot of snapins at the same time in your patch the CPU usage will increase accordingly. Thus, we cannot guarantee that the snapins will work flawlessly in all use cases even if your system does meet the minimum recommended system requirements.

The versions of the plugins installed via the Slate Digital installers, like other Slate Digital products, uses the iLok licensing scheme. If you are using these versions, an iLok (USB dongle or iLok cloud) with a valid license from Slate Digital is required to use the plugins.

Operating the controls

Most parameters of the Snapins are controlled by the knobs and the sliders seen in the UI. To move a knob or slider simply click on it, and while holding the mouse button down move the mouse up or down.

Sometimes you might want more precise control when tuning a parameter. Hold the **shift** key while moving a knob or slider to enter **fine tuning** mode, where the knob or slider will move more slowly.

You can reset a knob to its default position by double clicking it.

Finally, most controls support entering the value using your keyboard by right-clicking on them.

3-Band EQ

The EQ 3-Band is a three band EQ with adjustable split frequencies

Splits

 Adjusts cutoff frequency between low and mid, and between mid and high bands.
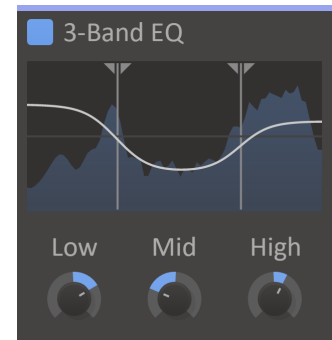
Low

 Gain for low frequency (bass) band.

Mid

 Gain for middle frequency (midrange) band.

High

 Gain for high frequency (treble) band.

Bitcrush

The Bitcrush can be used to create distorting effects that sound like that of scraping analog radio, or inherently lo-fi sound sources, like old video games. It simulates the audio being sampled and replayed using a low quality sampler with limited sample rate and bit depth.

Rate

Down sample the signal to a minimum of 200 Hz.

Bits

Quantize the amplitude of each sample of the signal.
A lower value will result in a more distorted sound.

ADC Q

Quality of the analog-to-digital conversion. A lower value will add dissonant aliasing in the low frequencies.

DAC Q

Quality of the digital-to-analog conversion. A lower value will add dissonant aliasing in the high frequencies.

Dither

Adds noise to the signal in order to reduce distortion caused by quantization.

Mix

The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Chorus

The Chorus enhances the stereo effect and presence of a sound by mixing it with delayed versions of itself.

Delay

    The average delay for the delayed voices.

Rate

    The frequency of how fast to vary the delay.

Depth

    How much to vary the delay.

Spread

    Stereo width of the effect. A lower value will go towards a mono output.

Mix

    The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Taps

    The number of chorus voices.

Comb Filter

The Comb Filter will mix the signal with a delayed version of itself, creating a filter with repeated troughs and peaks across the spectrum.

Cutoff

Cutoff setting for the filter, the distance between each peak.

Mix

The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Polarity

The polarity setting swaps troughs for peaks and vice versa, with the plus setting having a peak at 0 Hz, and the minus setting a trough at 0 Hz.

Stereo

The stereo setting flips the polarity setting for the right channel, allowing the the comb filter to be used for mono compatible stereo widening.

Compressor

The Compressor will even out the audio volume by lowering the volume when the signal is loud.

Attack

    The attack time is the time it takes to lower the volume when the input volume is over the threshold.

Release

    The release time is the time it takes to return the volume to normal when the input volume is under the threshold.

Mode

    In RMS mode the compressor will measure the volume using the root mean square method, which gives an accurate measurement of audio power. In peak mode the compressor will follow the peaks in the audio waveform, which makes it more responsive to transients.

Ratio

    The ratio decides how much the compressor will reduce the audio volume. At 1:2, for example, the volume will be lowered until it is halfway between the input volume and the threshold.

Threshold

    The threshold for when the compressor will start lowering the volume.

Makeup

    The makeup gain will increase the volume of the output signal to compensate for the loss in overall volume that the compressor causes.

VU Meter

    Displays the current input level, the selected threshold, and the compressor's current attenuation.

Sidechain

    When enabled attenuation is calculated based on a secondary input, but the effect is applied to the main input.

Convolver

Convolver uses the mathematical process of convolution to shape your input using a prerecorded sound. This allows for recording the reverb of any physical space and applying it to any sound for example.

Start

Cuts the start off the impulse response. Requires a precomputation step before the effect can be heard.

End

Cuts the end off the impulse response. Requires a precomputation step before the effect can be heard.

FadeIn

Fades in the start of the impulse response. Requires a precomputation step before the effect can be heard.

Fade Out

Fades out the end of the impulse response. Requires a precomputation step before the effect can be heard.

Stretch

Stretches the impulse response, adjusting both it's length and pitch. Requires a precomputation step before the effect can be heard.

Select Sample

Convolver comes with a large library of Impulse Responses that should cover many of your basic needs. But you can import any samples you want.

Reverse

Reverse the impulse response before processing.

Delay

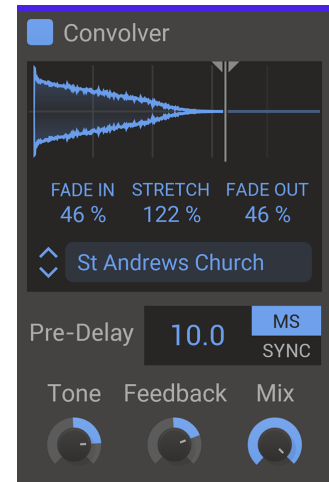Set the pre-delay time in milliseconds or as a fraction of song tempo.

Tone

Adjusts the brightness of processed output.

Feedback

Feeds the processed output back into the convolution engine.

Mix

The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Delay

The Delay will delay the input signal for an echoing effect.

Delay

The amount of time beore the delayed sound starts playing. This will be expressed in milliseconds or as parts of a beat, depending on Sync Mode.

Sync Mode

When sync is enabled the delay time will be synchronized to the song tempo.

Feedback

The feedback setting will cause the delayed sound to feed back into the delay. This will create an exponentially decaying echo.

Pan

Adjust the panning of the delayed sound.

Ping-Pong

Swaps the left and the right channel of the delayed sound when it is fed back into the delay. When combined with panning this will make the echo bounce back and forth between the speakers.

Duck

When duck is turned up, the output volume from the delay will automatically be lowered when the input volume is high.

Mix

The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.
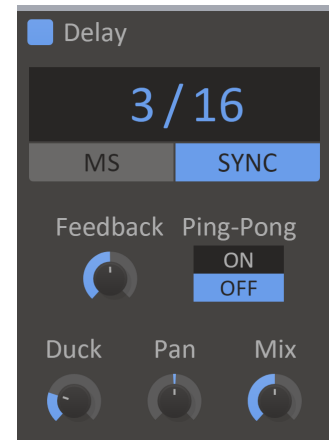
Distortion

The Distortion is a versatile distortion effect with a wide selection of algorithms.

Drive

 The drive setting will boost the input signal, causing a heavier distortion.

Bias

 The bias will add a DC offset to the signal before distorting. Adding some bias can prevent the distorted audio from sounding hollow and uninteresting.

Spread

 The spread will add different amount of bias to the left and right channels. This can give some nice and subtle stereo widening.

Type

 The flavor of distortion. Select between overdrive, saturate, foldback, sine, hard clip and quantize.

Dynamics

 A common problem with distortion is that it can remove the dynamic content of the input signal, forcing the output to maximum volume. Turn up this knob to preserve the dynamics of the input.

Mix

 The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.
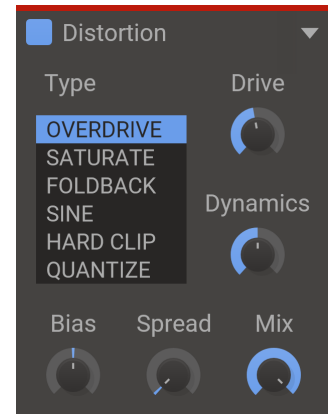
Dynamics

Dynamics processing featuring upward and downward compression as well as expansion. Thresholds and ratio are controlled by clicking and dragging on the visualization in the upper part of the plugin interface. The visualization shows how input levels are mapped to output levels by the dynamics processing. The current input and output levels are marked in the visualizer by a moving disc.

Low Threshold
> Low threshold for upwards compression and expansion.

Low Ratio
> Ratio of upwards compression or expansion below the low threshold.

High Threshold
> High threshold for downwards compression.

High Ratio
> Ratio of downwards compression above the high threshold.

Attack
> Adjusts attack time.

Release
> Adjusts release time.

Knee
> Smoothes the knee of the compression curve across the thresholds.

In Gain
> Adjusts gain for the incoming signal.

Out Gain
> Adjusts gain after compression.

Mix
> The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Ensemble

The Ensemble effect creates the illusion of many voices playing in unison. Much like a chorus it creates this effect by playing delayed copies of the incomping sound. On top of this it also modulates the phases of each voice to create a silky smooth result without any metallic flanging. The delay times for each voice is also modulated in order to detune each voice slighty.

Voices

> Number of voices to play simultaneously.

Detune

> How quickly to modulate the delay for each voice, affecting how detuned the voices will be.
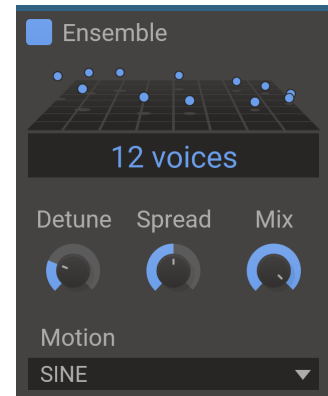
Spread

> Pans voices left or right for a stereo effect.

Mix

> The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Motion

> Selects between different patterns for the modulations of the voices.

Filter

The Filter snapin provides a selection of common filters.

Type

The type of filter. Select between low pass, band pass, high pass, notch, low shelf, peak and high shelf filters.

Cutoff

The operating frequency of the filter. In a low-pass filter this is the frequency where the signal is reduced by 3dB.

Q

The filter Q setting. High values for Q will make the filter resonate at the cutoff frequency.

Gain

The gain value for the low shelf, peak and high shelf filter types.

Filter slope

The slope of the filter cutoff, with 1x representing a classic 2-pole filter.

Flanger

Creates a flanging effect by mixing the audio with a slightly delayed version of itself. The length of the delay can be adjusted manually and modulated. Optionally, this effect can also add a phase shift between the dry and wet signals to create an infinte barberpole-style flanging effect upwards or downwards.

Delay

> Adjusts the minimum delay.

Depth

> Depth of delay modulation. Added on top of the minimum delay set by the delay knob.

Rate

> Rate of delay modulation.

Scroll

> Enables the phase offset and motion functions of the effect.

Offset

> Phase offset between dry and wet signals.

Motion

> Rate of modulation for the phase offset.

Spread

> Stereo spread between left and right channels. Affects delay modulation and phase offset.

Feedback

> Feedback of the wet signal back into the delay line.

Mix

> The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Formant Filter

The Formant Filter will boost two frequencies to mimic the sounds of different vowels.

Vowel Selector
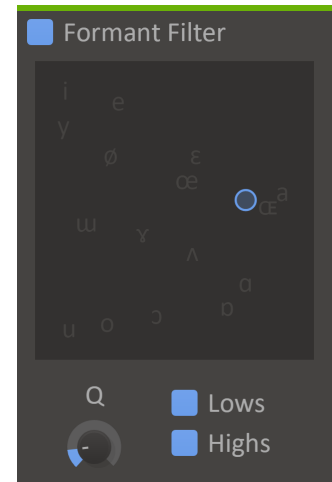> Selects two frequencies to boost.

Q
> Adjust how powerful and narrow the frequency boost is.

Lows
> Allow low frequencies through the filter.
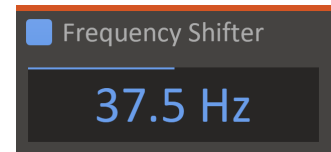
Highs
> Allow high frequencies through the filter.

Frequency Shifter

Frequency shifter will shift all the frequencies in the input signal up or down by a certain amount. This kind of shifting will ruin the harmonic content of the input signal, making it sound dissonant.

Shift
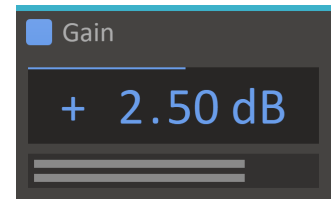
How much to shift all frequencies by.

Gain

The gain snapin will increase or decrease the volume of a signal.

Gain

Set how much to increase or decrease the volume, in decibels.

VU Meter

Displays the current output level on the left and right channels.

Gate

The Gate will only let audio through when the input level is above a set threshold.

Attack

    The attack time is the time it takes to for the gate to fully open when the input volume is over the threshold.

Hold

    The hold time is the minimum time the gate will stay open.

Release

    The release time is the time it takes to for the gate to fully close when the input volume is under the threshold.

Threshold

    The volume threshold for when the gate will open.

Tolerance

    A hysteresis range requiring the volume to drop a set amount of dB under the threshold before closing.

Range

    The amount to attenuate the signal when the gate is closed.

Look-ahead

    When enabled, a 5ms look-ahead will be used, allowing transients through at the cost of latency.

Flip

    When flipped, the gate will act in reverse attenuating the signal when the gate is open.

VU Meter

    Displays the current input level, the selected threshold & tolerance, and the gate's current state.

Sidechain

    When enabled transients are detected based on a secondary input, but the effect is applied to the main input.
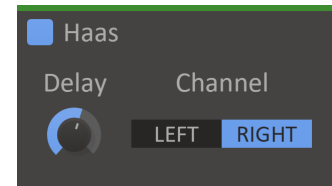
Haas

The HAAS effect will widen the stereo of the audio by delaying the left or the right channel slightly.

Channel
> Which channel to delay.

Delay
> The delay time.

Ladder Filter

The Ladder Filter simulates low pass filters found in classic hardware synths.

Cutoff
> The filter cutoff frequency.

Resonance
> The filter resonance setting. High values will make the filter resonate at the cutoff frequency.

Topology
> Selects between transistor ladder and diode ladder topology. The diode ladder have a slightly more gentle rolloff after the cutoff frequency. The two topologies also behave differently when saturation is enabled.

Saturate
> Simulates saturation of electronic components in the filter.

Drive
> Simulates overdrive of the components in saturate mode.

Bias
> Simulates bias voltage over the components in saturate mode.

Limiter

The Limiter will prevent the audio volume from going over a certain threshold.

In gain
  Gain to apply to the input signal before limiting.

Out gain
  Gain to apply to the input signal after limiting.

Threshold
  The maximum allowed volume.

Release
  The release adjust how quickly the limiter returns the volume back to normal after limiting it due to a peak in the input volume.

VU Meter
  Displays the current input level, the selected threshold, and the limiter's current attenuation.

Nonlinear Filter

Filter that colors and distorts the signal due to internal nonlinearities.

Type

    The type of filter. Select between low pass, band pass, high pass and notch filters.

Cutoff

    The operating frequency of the filter.

Q

    The filter Q setting. High values for Q will make the filter resonate at the cutoff frequency.

Drive

    Overdrives the filter, which makes the effect of the nonlinear behaviour more prominent.

Mode

    Selects what flavor of nonlinearity you want. The clean mode has no nonlinearies, and does not color the signal. All other modes distort and color the signal in different ways to some extent.

Phase Distortion

The phase distortion distorts the signal by offseting the phases of the individual harmonics of the input signal. The amount of phase offset is controlled by the signal itself, much like FM feedback.

Drive
>Controls the amount of distortion.

Normalize
>Normalizes the signal, making the effect insensitive to input gain.

Tone
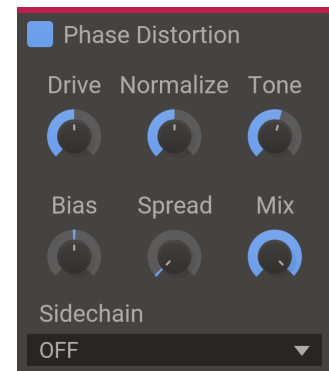>Filters the modulation to reduce high frequency noise.

Bias
>Adds a constant phase offset to all harmonics.

Spread
>Spreads phase offset for left and right channels for a stereo effect.

Mix
>The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Phaser

The Phaser will filter the input signal, creating a series of moving peaks and troughs in the audio spectrum.

Order

A higher order setting will increase the order of the filters used by the phaser, creating a more pronounced effect with more peaks and troughs.

Cutoff

Sets the cutoff of the filters in the phaser, moving the peaks and troughs in the audio spectrum.

Depth

Adjusts the depth of modulation of the cutoff.

Rate

Adjusts the rate of modulation of the cutoff.

Spread

Adds a phase offset for the cutoff modulation between the left and right channels, for a stereo widening effect.

Mix

The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Pitch Shifter

The Pitch Shifter will adjust the pitch of the input signal up or down.

Pitch

> How much to adjust the pitch, in semitones.

Jitter

> How much randomness to add to the pitch. A high jitter setting can give a unison-like effect.

Grain Size

> During processing the pitch shifter chops up the audio into small snippets called grains. This setting adjusts the length of the grains, which can influence the sound.

Mix

> The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Resonator

The Resonator snapin adds harmonic resonance to the input signal.

Pitch

    The frequency at which to resonate.

Decay

    Sets how long it takes for the resonance to ring out after the input goes silent.

Intensity

    Adjusts how much the resonance amplifies the input signal.

Timbre

    Switches between two different harmonic series for the resonance. Choose between all harmonics (saw tooth wave) or odd harmonics (square wave).

Mix

    The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Reverb

The Reverb adds the sense of space to any sound by emulating the sound bouncing off the walls in a physical room.

Decay

The reverberation time, i.e. the time it takes for the reverb to go silent after sound has passed through it.

Dampen

Adds damping to high frequencies so that they decay faster than low frequencies.

Size

Adjust the size of the virtual room that reverb simulates. Ranges from closet to church.

Width

Adjusts the stereo width of the reverb. At 100% the left and right channels are completely uncorrelated in the wet sound.

Early

Adjusts the balance between early and late reflections. A higher value will give a brighter and more responsive reverb.

Mix

The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.
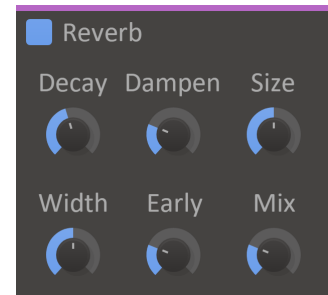
Reverser

The Reverser plays back delayed reversed sections of the input, mixed with the original dry sound.

Delay time

How long sections to delay and reverse. For example, setting this to 1/4th means every beat will be played back in reverse 1/4th of a bar later.
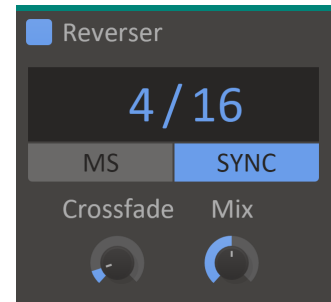
Sync

When sync is enabled the delay time will be synchronized to the song tempo.

Crossfade

Time to ramp in/out the reversed audio to avoid pops, in percent of the reversed section length.

Mix

The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Ring Mod

Ring Mod modulates the input with either an internal signal generator or a secondary input signal.

Bias
> Amount of positive bias to add to the secondary input.

Rectify
> Amount of positive or negative rectification to apply to the of the secondary input.

Mix
> The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Frequency
> The base frequency of the internal oscillator or filter cutoff for the internal noise generator.

Spread
> Shifts the frequency of the internal generator slightly for left and right channels to achieve a stereo effect.

Stereo

The Stereo snapin can adjust the stereo width and panning. It also displays the current balance and channel correlation visually.
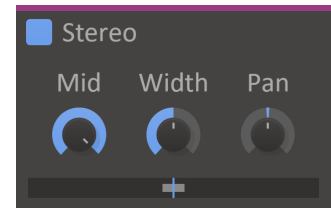
Mid

Adjusts the volume of the mono part of the signal.

Width

Adjusts the volume of the stereo part of the signal. The input audio must have at least a little stereo information for this knob to do anything.

Pan

Pans the signal to the left or right.

Stereo Meter

Displays the current balance and channel correlation. When the meter moves into the red area the correlation is less than zero, which can cause problems with mono compatibility.

Tape Stop

Tape Stop simulates the sound of slowly stopping and starting a playing tape.

Play
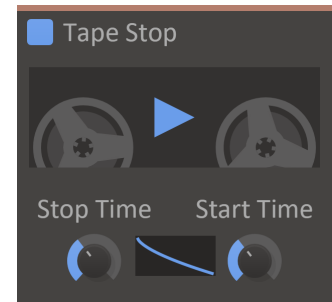> The current state of the tape motor.

Stop Time
> Time until the tape motor reaches full stop when stopping.

Start Time
> Time until the tape motor reaches full speed when starting.

Curve
> The speed curve of the tape motor starting/stopping.

Trance Gate

The Trance Gate will modulate the volume of your audio based on a programmable rhythmic sequence.

Pattern Select

Switches between the eight different pattern slots.

Pattern Editor

Edits the current pattern. Click to toggle steps on or off, click and drag to tie steps together.

Length

The length of the current pattern.

Attack

Attack time for the amplitude ADSR envelope.

Decay

Decay time for the amplitude ADSR envelope.

Sustain

Sustain level for the amplitude ADSR envelope.

Release

Release time for the amplitude ADSR envelope.

Mix

The dry/wet mix of this effect. A lower value will let some of the unmodified signal through.

Resolution

Length of one step in the sequencer.

Transient Shaper

Transient Shaper adjusts the dynamics of a sound, with a focus on the initial hit; the transient.

Attack
> The amount of amplification or attenuation of the transient.

Pump
> The amount of attenuation directly after the transient, emphasizing the transient without increasing the level.

Sustain
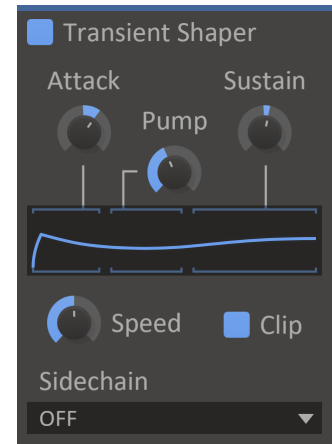> The amount of amplification or attenuation of the sustained sound.

Speed
> Higher values results in snappier transient modification, and lower values result in smoother curves.

Clip
> When enabled, the output signal is clipped to 0dB.

Sidechain
> When enabled transients are detected based on a secondary input, but the effect is applied to the main input.

Acknowledgements