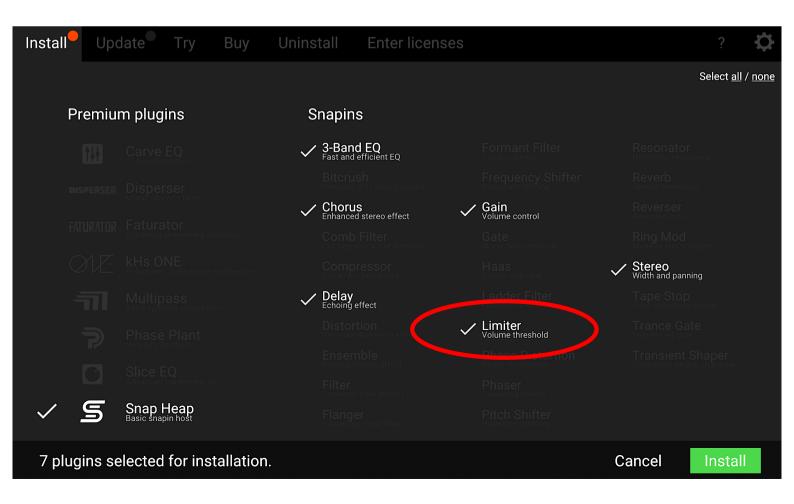


# GRATUIT

https://kilohearts.com/products/limiter



# KII OHFARTS

Kilohearts crée une nouvelle génération de plugins audio qui se concentrent sur le workflow. Nous pensons que la création de musique moderne nécessite des outils modernes qui ne sont pas limités à la réplication du matériel.

# Limiteur













Obtenez l'extension de rack

Que vous souhaitiez augmenter les dernières gouttes de gain de votre piste ou simplement contrôler quelques pics forts, un limiteur peut être l'arme de choix.

En regardant un peu vers l'avenir, un limiteur peut garantir que votre signal ne va jamais plus fort que vous le souhaitez sans déformer ou détruire les transitoires. Même avec le bouton tourné à 11.



### Dans le bouton de gain

Gain à appliquer au signal d'entrée avant la limitation.

### Bouton de gain de sortie

Gain à appliquer au signal d'entrée après limitation.

### Bouton de seuil

Le volume maximum autorisé.

### Bouton de déverrouillage

Le relâchement ajuste la vitesse à laquelle le limiteur ramène le volume à la normale après l'avoir limité en raison d'un pic du volume d'entrée.

### **VU Meter**

Affiche le niveau d'entrée actuel, le seuil sélectionné et l'atténuation actuelle du limiteur.

### Panneau Paramètres

Chaque fois que vous placez le curseur de votre souris sur un composant logiciel enfichable, il y a une petite flèche dans le coin supérieur droit (non visible sur la capture d'écran). Il ouvre un panneau de paramètres dans lequel vous gérez les préréglages.

Il dispose également d'un bouton "randomiser" qui peut être utile. Je suppose...

### Case à cocher activée

La petite case à gauche du nom du plugin est une case à cocher qui contourne l'effet lorsqu'elle est désactivée.

### Redimensionner la poignée

Le coin inférieur droit de tous les plugins Kilohearts est une poignée de redimensionnement pour mettre à l'échelle l'interface utilisateur à n'importe quelle taille. Cela vous permet d'avoir une bonne vue des commandes quelle que soit la résolution de l'écran, et est également utile si vous avez besoin de grosses commandes, par exemple lorsque vous utilisez Limiter comme effet en temps réel sur un écran tactile pendant un live set. (Ceci n'est pas disponible lorsque le composant logiciel enfichable est utilisé à l'intérieur d'un hôte de composant logiciel enfichable.)

# Explorez l'écosystème Snapin

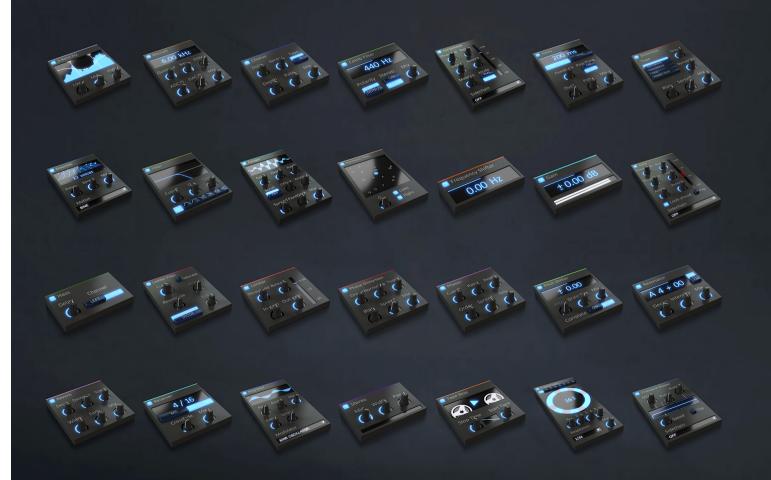
Les snapins sont des plugins d'effets réguliers qui fonctionnent également comme des modules dans nos hôtes de snapin. Ce sont des plugins légers, mais puissants, qui vous présentent les paramètres importants de manière simple. Chaque réglage et composant est clairement documenté, vous savez donc exactement ce qui se passe lorsque vous tournez ce bouton.

Les composants logiciels enfichables sont achetés individuellement ou en lots, alors bénéficiez d'une remise importante sur les lots ou choisissez vos favoris à la cerise. Le nombre de snapins disponibles est en constante augmentation, et Kilohearts vise à faire de Kilohearts Toolbox le bundle de conception sonore le plus complet du marché.

Commencez avec Kilohearts Toolbox FREE qui dispose de 6 snapins et de l'hôte snapin Snap Heap. Il offre de nombreuses options de modulation pour créer des presets impressionnants et des tranches de console personnalisées très rapidement.



Multipass est un hôte de composant logiciel enfichable à fractionnement de bande, où vous pouvez diviser votre son entrant en 5 bandes personnalisables au maximum, puis appliquer les effets que vous aimez à chacun.



# <u>snapin</u>

# **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**

Windows (7 or newer) or Mac OS X (10.7 or newer)

### **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

# **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.



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.



# **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



## **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 and hard clip.

### **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

# **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.

# Ensemble 12 voices Detune Spread Mix Motion SINE

### **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.



# **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



# **Formant Filter**

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

### **Vowel Selector**

Selects two frequencies to boost.

Q

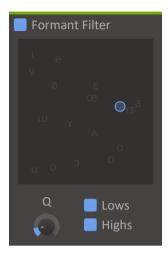
Adjust how powerful and narrow the frequency boost is.

### 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

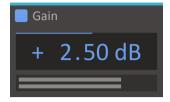
How much to shift all frequencies by.

# 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

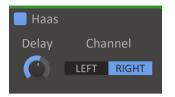
The HAAS effect will widen the stereo of the audio by delaying the left or the right channel slightly.

### 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.

# Cutoff Saturate Criminal Drive Resonance Bias TRANSISTOR DIODE

### **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.



# **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.

# Phase Distortion Drive Normalize Tone Bias Spread Mix

### **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

# **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

# **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

# 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



# 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

# 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

# **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.

# Stereo Mid Width Pan (1)

### Width

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

### Pan

Adjusts the panning.

### **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.



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**

This development of these products was helped by the following pieces of excellent open source software:

**Boost C++ Libraries** 

Skia Graphics Library Copyright © 2011, Google Inc.

Symbiosis AU/VST Copyright © 2010-2013, NuEdge Development / Magnus Lidström

LodePNG Copyright © 2005-2015, Lode Vandevenne

C++ optimized SHA1 algorithm Copyright © 2011, Micael Hildenborg

miniz By Rich Geldreich

FastDelegate

By Don Clugston

FFTReal By Laurent de Soras