

FILTERVERSE



USER MANUAL

V0.7.3 (Public Beta)

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By



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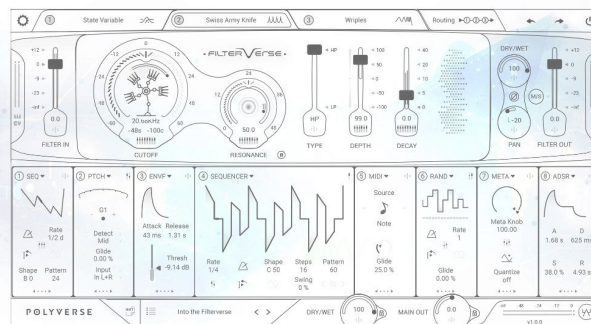
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Introduction

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Special thanks to our Beta Test Team, who were invaluable not just in tracking down bugs, but in making Filterverse a better product.



About Filterverse

An endless universe of filtering possibilities

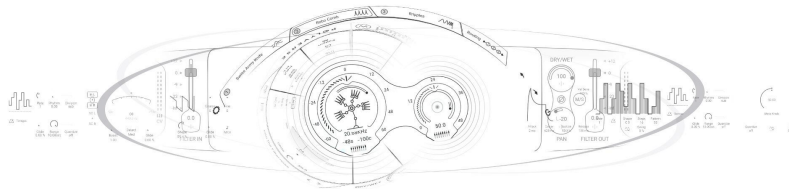
Filterverse is a new type of multi-filter plug-in that elevates the concept of the filter to a musical instrument in its own right. From traditional lowpass, bandpass, and highpass filters to experimental resonating bodies, comb designs, decimators, and much more, it's a one-stop-shop for all your filtering needs, opening up previously unimagined possibilities in sound design. You can also use Filterverse in the simplest of ways, for example, as a static lowpass synth-style filter, or one whose cutoff frequency responds rhythmically to the input signal.

Every filter within Filterverse is the product of meticulous engineering and rigorous aural testing. From the traditional filters foundational to analog synthesis to bleeding-edge sonic scalpels, every detail, resonance, and saturation pathway has been carefully conceived and engineered. The result is unparalleled audio clarity and richness.

The modulation system in Filterverse is a cosmos of possibilities. Featuring eight modulation sources including a step sequencer/LFO, audio-range oscillator, a pitch detector, randomizer, and more. It lets you craft dynamic rhythms, motions, and evolving atmospheres with precision and depth.

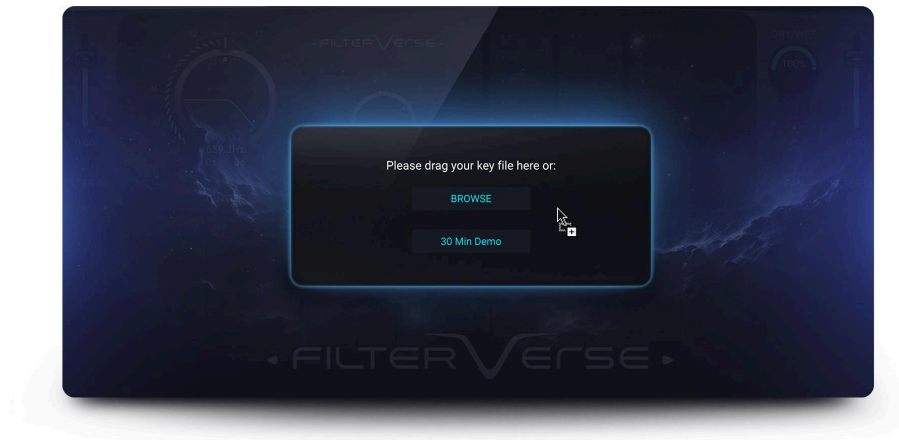
We know that the selection of plug-ins available to music creators and DAW users can be confusingly large. We believe the primary focus of any musical tool should always be creativity. That's why we intend Filterverse to provide a completely new palette of sounds and technologies formerly accessible only in elaborate modular synth setups or via bleeding-edge coding. We've done that work so you don't have to.

Step into the future of sound with Filterverse, where every detail amplifies your creativity.



Getting Started

Installation



To install Filterverse, simply run the installer and follow the instructions on your screen. For free tutorials and videos about setting up and using Filterverse with your favorite DAW (Digital Audio Workstation), please visit our website at <http://polyversemusic.com/support>

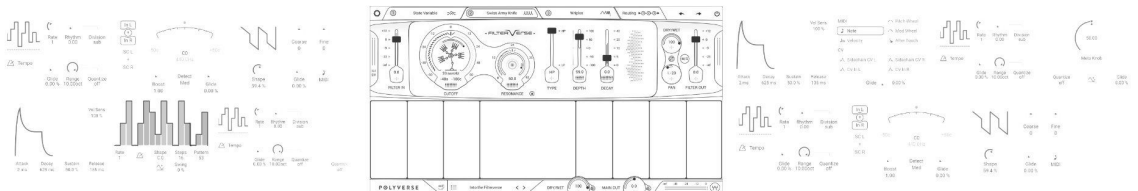
When using Filterverse for the first time, a registration dialog will appear. To register and authenticate the plug-in, first download, then drag and drop the key file you have received in your purchase confirmation email onto the dialog box.

If you've lost your confirmation email, you can re-request one to be sent automatically from our support page at <http://polyversemusic.com/support>

Compatibility:

Windows: 10 or later; 64-bit VST, VST3, AAX

OSX: 10.13 or later; 64-bit VST, VST3, AAX, AU



MIDI and CV control

Filterverse can be inserted into audio or instrument tracks just like any audio effect. However, some of its features involve responding to MIDI notes and MIDI CC (continuous controller) messages. Filterverse also responds to CV (control voltage) via sidechain input in DAWs that support it.

If you would like to use Filterverse as a standard audio effect, just load it in one of the insert slots on your DAW, and you're done. Filterverse makes its parameters available for DAW automation across AAX, AudioUnit, and VST formats.

If you want to use MIDI to control Filterverse's parameters and trigger its envelopes manually, please refer to the following pages for instructions on routing MIDI within your specific DAW.

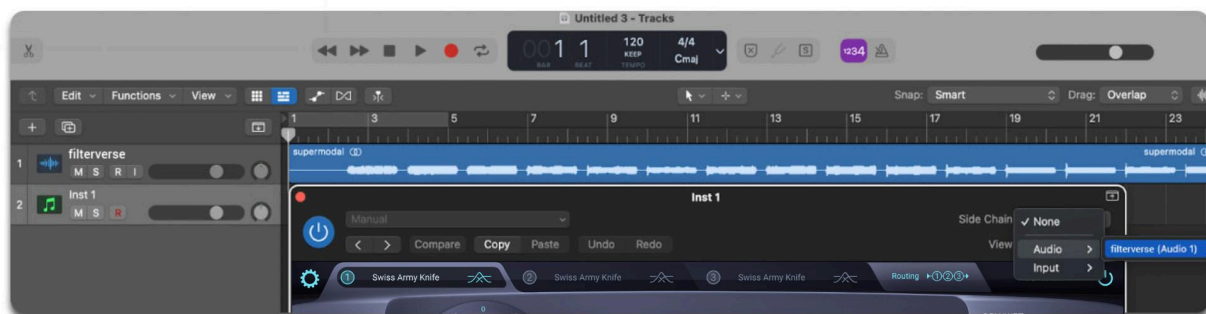
Ableton Live

- Click the "I-O" button to show the input and outputs.
- Create an audio track, and add some audio onto it.
- Add Filterverse onto the audio track.
- Create a blank MIDI track.
- Set the MIDI track's output to the audio track with Filterverse.
- While audio is flowing through Filterverse, play some MIDI notes on your virtual or external keyboard controller.



Apple Logic Pro

- Create a software instrument track.
- Click on the instrument slot to select Filterverse from the submenu “AU MIDI-controlled Effects.”
- Open a new audio track and place your audio file inside it.
- Open the Filterverse plug-in interface. At the top right, select your sidechain input to be the same as the audio track.
- Mute the audio track, as its signal is now being routed through Filterverse.
- Select/enable the plug-in’s instrument track.
- Play some MIDI notes as the audio streams through the plug-in.



Avid Pro Tools

- Create a new audio track.
- Add Filterverse as an insert effect.
- Place an audio file in the track.
- Create a new MIDI track.
- Route the MIDI track's output to the plug-in.
- Arm the MIDI track.
- Play some MIDI notes as the audio streams through the plug-in.



Image-Line FL Studio

- Drag an audio file to your playlist and route it to “Track 1” on the mixer.
- Load Filterverse as an insert on “Track 1” on your mixer.
- Open a “MIDI Out” plug-in and set the Port to 1.
- Set the Filterverse MIDI input port to 1 as well
- Select the “MIDI Out” channel.
- Play some MIDI notes as the audio streams through the plug-in.



Bitwig Studio

Bitwig Studio offers a number of ways to control a plug-in via different track types. For the sake of simplicity, we will outline a method using separate audio and instrument tracks.

- Create an audio track.
- Insert Filterverse on this audio track.
- Create an instrument track.
- Click on the output menu of the instrument track.
- In this menu, navigate to under “NOTES TO TRACKS.” Select “Tracks” then the name of the *audio* track on which Filterverse resides.

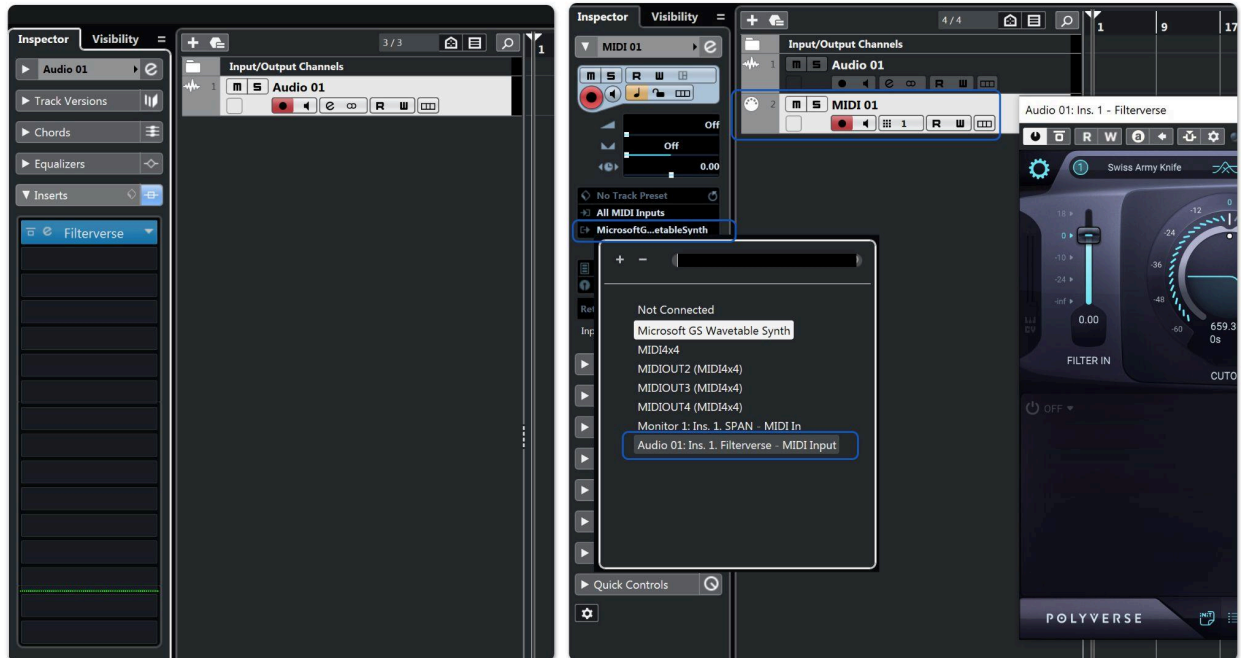


PreSonus Studio One

Coming soon ...

Steinberg Cubase

- Add an audio track.
- Add Filterverse as an insert effect.
- Place an audio file in the track.
- Add a MIDI track.
- Set the MIDI track's output to the Filterverse plug-in.
- Play some MIDI notes as the audio streams through the plug-in.



Structure of Filterverse



Filterverse is a collection of Filters, but it is so much more.

The heart of Filterverse consists of three filter sections with variable routing. You can choose from a plethora of filter and effect models for each section. The possibilities for sculpting sounds with surgical precision, broad strokes, or anything in between are virtually limitless.

Up to eight modulation sources can be active at once. Almost any parameter in Filterverse can be a modulation destination, with the modulation amount set using a convenient drop-down menu box accessed by clicking just below the desired control onscreen. Of course, one source can modulate multiple destinations and/or the same destination can be modulated by multiple sources.

Modulation sources comprise a combination sequencer/LFO, synth-style ADSR envelope generator, audio-rate oscillator, envelope follower, MIDI/CV input (which means Filterverse parameters can be modulated by our Gatekeeper plug-in), randomizer, pitch detector, and our signature Meta Knobs, which can control multiple parameters with a single control gesture. You can mix and match these modulation sources in up to eight modulation slots.

Overview

Let's begin with a brief look at the three main control areas of Filterverse, going from top to bottom of the plug-in window. Explaining what these areas do is as good a way as any to get a big picture of Filterverse's features and creative possibilities.

Filter Section

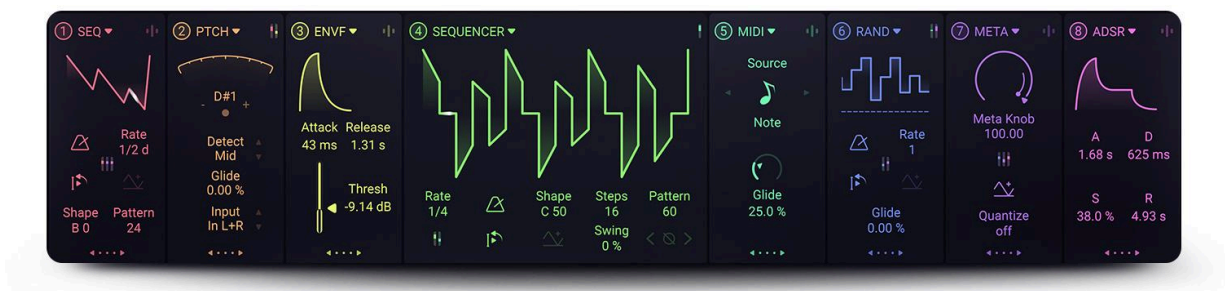


The main architecture of Filterverse consists of three filter slots, accessible via tabs across the top of the plug-in window. For each of these, you can choose from a roster of filter models; the controls for each filter model appear directly below the active tab.

A separate filter routing tab lets you choose from five routing schemes: serial, parallel, and three combinations of serial and parallel behavior. What's more, you can change the position of each filter slot within a routing scheme by dragging its tab to the left or right.

The top area also contains a “gear” icon to open the [Settings panel](#) as well as undo/redo arrows and the overall on/off (bypass) button for the plug-in.

Modulation



Below the filter area are the modulation sources. There are eight slots in total, each of which you can quickly and easily assign to nearly any parameter in Filterverse. Each slot can be populated by any of eight types of source: a synth-like ADSR envelope, step sequencer, oscillator, envelope follower, random value generator, pitch detector, MIDI/CV input, and Meta Knob, which simply controls multiple parameters with a single knob twist (MIDI-learning the Meta Knob to a hardware control is therefore especially powerful). Modulation sources can even cross-modulate each other.

Presets and Main Output



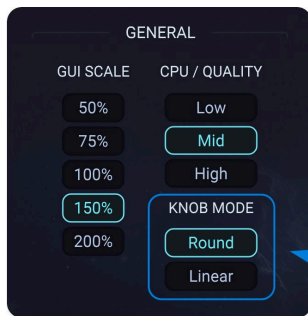
The star of the show here is the preset browser. Beyond simply stepping through presets, the browser allows you to search for presets by text (e.g. part of a name) or tagged attributes such as “dramatic,” “subtle,” and so on. The master dry/wet mix, main output level, and clipping style controls are also on hand.

We'll explore each of these sections in detail, but first, let's look at some common behaviors that apply across all the controls in Filterverse. They're designed to make your musical life easier.

Common control operations

The controls in Filterverse have a few common behaviors meant to make working with the plug-in quicker and more intuitive. Generally, you can do the following:

Knob behavior



In the [Settings panel](#), accessed by clicking the gear-shaped icon at the top left of the plug-in window, you will find a selection for **Knob Mode**. This determines how circular knobs on the screen respond to mouse movement.

- **Round:** Move the mouse around the circumference of the knob to change values, as though you were turning a physical knob.
- **Linear:** Move the mouse up or down while hovering over a knob to change values.

This setting has no effect on slider or button controls.

Fine adjustment

There are a couple of different ways to adjust parameters with fine precision in Filterverse.

- Hold the shift key while dragging on a control to move it more slowly and thus make fine adjustments. This works in both Round and Linear knob modes.
- You can also move the mouse away from the control to allow finer adjustments. The farther away the mouse cursor is, the slower the value will change in response to physical motion. Simply click-hold on any control, then move the mouse horizontally away from the control. In Linear knob mode, you would then drag up and down. In Round mode, you'd move in larger circles around the desired control. This may work more or less well depending on your "real estate" onscreen.

Double-click to default

Double-click on any graphical control to return it to its default value. This is often zero, or sometimes a median value — 50 percent for dry/wet mixes, for example. Then, double-clicking the control again resets it to the value that was set *before* it was “zeroed.” This can be an incredibly helpful tool for A-B comparisons when designing your own sounds.

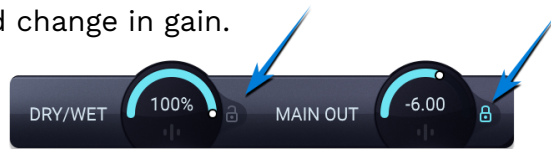


Direct entry

While you can drag up or down on any numerical value or field, double-click on the field to enter a specific value right from your computer keyboard.

Padlock icons

Settings pertaining to levels, dry/wet mixes, and panning can have a huge impact on the signal level in your track. These all have clickable padlock icons near them. When locked, the settings remain at their currently set values even when you change [Presets](#). This lets you experiment with different effects without causing a large and unexpected change in gain.

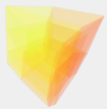


MIDI learn

You can map Filterverse parameters directly to physical controls on a MIDI controller. This includes the main controls as well as parameters within the modulation sources themselves.



1. First, make sure Filterverse is set up to receive MIDI [in your DAW](#).
2. Right-click on any parameter to display its MIDI learn pop-up.
3. Now, click the Learn button to switch the display to activate recording mode.
4. Move the control you want to use on your MIDI device, and the display should change to reflect the newly learned assignment, e.g. CC 74 assigned to filter cutoff as shown here.
5. If desired, you can click “Unmap” to unlearn the assignment and assign a different control.



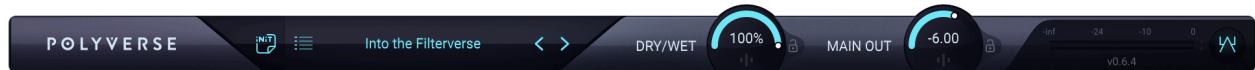
Word to the Wiz:

DAWs (e.g. Apple Logic Pro) may have their own native means of mapping physical MIDI controls to parameters in third-party plug-ins, which may conflict with or override Filterverse’s procedure. When in doubt, always consult your DAW’s documentation!

Master Output Controls

The main output controls are located in the bottom right half of the filterverse window.

Detailed descriptions of each filter’s behavior and parameters are provided in the [Filter Reference](#) section.



Master Dry/Wet Mix



The Dry/Wet mix dial at the bottom of the plug-in window controls the overall effect balance, while preserving the [dry/wet](#) mixes of the three individual filters. With the master mix here, turning the dial all the way “dry” will cause only the input signal to be heard, with no filtering. Turning it all the way “wet” will produce only the sound as processed by Filterverse.

Main Out



This is simply the master output volume. It is the last part of the signal path before audio is returned to your track, and preserves all differences and relationships in levels from “upstream” in the plug-in, such as [Filter In](#) and [Filter Out](#) settings for the individual filters.

The Main Out dial affects the overall signal regardless of the Master Dry/Wet Mix setting.

Master VU Meters



The horizontal VU meters display the overall output level (whereas the vertical VU meters in the top section display input and output levels for individual filters). Signals higher than 0dB will display red clipping indicators as shown above. Click on these to clear them.

Clipping Mode Selector

Filterverse offers a clipping and saturation algorithm that is useful for taming output signal levels that surpass 0dB, while retaining a punchy and aggressive sound. It is controlled by the small icon at the far lower right corner, to the right of the master VU meters. Click the icon to rotate through three options:



Off: Clip is off.

Clip: Filterverse will hard-clip audio at 0dB (upstream of the output knob).

Soft Clip: Clipping will have a slight curve for a softer and rounder sound.

Filter Section

Filter Tabs



The heart of Filterverse is its three filter tabs, each of which can be populated with a large variety of filters of all kinds. An important thing to remember about this section is that almost all the parameters you see here — including the Filter In and Filter Out levels, Dry/Wet Mix, Pan knob, and so on — are independent *per filter*. This makes Filterverse especially flexible. Click a tab to bring it to the front.

On/Off

Turn any of the three filters on or off by clicking its circled number icon in the tab. When the filter is active, the number and icon to the right of the name illuminate in blue. When the filter is inactive, these are grayed-out.

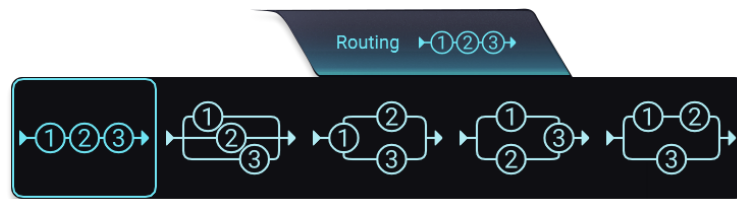
Selecting a filter



In any tab, click on the filter name to bring up a drop-down menu, from which you can then select the filter that populates that tab. You can also mouse over the graphical icon to the right of the filter's name, which will display left-right arrows that let you step through the filter list sequentially.

Detailed descriptions of each filter's behavior and parameters are provided in the [Filter Reference](#) section.

Filter routing



Click the filter routing tab to bring up the above options. From left to right, these are:

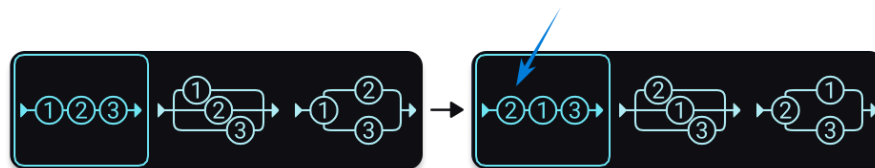
- Full serial routing
- Full parallel routing
- Filter in first tab routed serially into filters in second and third tabs, which are in parallel
- Filters in first two slots in parallel, both routed serially into filter in third tab
- Parallel routing of filters in first two tabs, which are serial relative to each other; with filter in third tab.

Why are we saying things like “filter in the first tab” and not simply “Filter 1”? There’s a good reason, which we explain in the section [How position and routing interact](#), below.

Filter routing position



You can swap the positions of any two filters by dragging one tab to the other's position, as shown above. The filter in the destination tab will then assume the position of the filter you dragged.



Here's where things get interesting. Notice anything different between the two images? That's right: Filter 2 is now in the position Filter 1 occupied previously, and vice-versa.

This is because Filterverse correlates the positions in the routing schemes to the absolute left-to-right order of the tabs. Dragging filters around is a quick way to change the overall sound, sometimes dramatically. For example, a filter driven into distortion will sound very different before or after a reverb.

Experiment with dragging some tabs around and then see how things change in the Routing tab, and this will quickly become intuitive. The positions of the numbers in the routing scheme will always correspond to those across the three filter tabs.

Filter Level and Pan Sections



Filter In

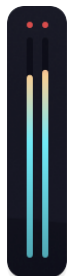
Adjusts the input volume of the filter and offers up to a 18dB boost of the input signal. Many of Filterverse's filters have internal saturation, and boosting the input signal takes advantage of this feature.

Filter Out

Adjusts the output volume of the Filter instance. This can be very useful for taming the resulting gain in your track without having to reach for the fader in your DAW's mix window, or for boosting volume to overdrive the next filter in the chain.



Level Meters



Both the Filter In and Filter Out sliders feature vertical bar-graph-style level meters. The level corresponds to the legend next to the sliders.

When a peak is reached (often indicating overdrive or saturation), two red dots will appear at the peak level. Click on the dots to clear them. Note that although the sliders are independent per filter, clicking on them in any filter will clear peak indicators for either slider in *all* filters.

MIDI and CV indicators

Below the Filter In meters are two icons: a keyboard for MIDI and the letters CV for control voltage. Either will pulse to indicate incoming MIDI or control voltage. In the case of CV, the letter C will pulse to indicate a signal coming from the left side of a stereo or two-channel input; R pulses to indicate a signal from the right.



Dry/Wet

The Dry/Wet control crossfades between the unprocessed and processed signals. At maximum (wet), the audio you hear is fully affected by the filters. At the minimum (dry), no filtering is applied to the audio.

Phase

The phase button, located in between the Dry/Wet knob and the Pan control, inverts the polarity of the filter. This does not affect the polarity of the dry signal, allowing for the creation of new filter models by mixing in the dry signal and controlling the polarity of the filter.

To use the phase button, click on it to toggle the polarity of the filter.

Pan

The pan control allows the audio signal to be positioned between the left and right channels. The pan control ranges from -100% to +100%, where -100% is full left and +100% is full right. At the center position, both channels are balanced. The modulation of the panning uses a single (mono) channel — as the effect itself controls both channels at once.



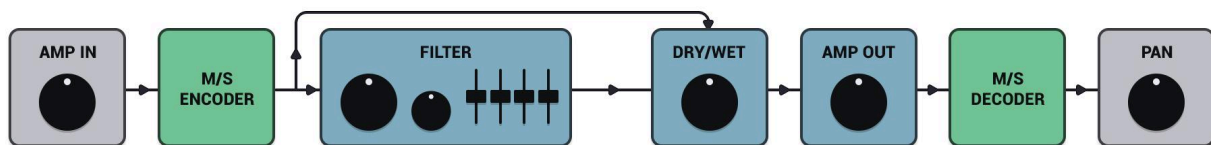
Mid-Side

The Mid-Side technique in audio plug-in processing involves separating a stereo track into *Mid* (mono content directly in front; sum of left and right) and *Side* (stereo; difference between left and right) channels. It is mainly used to process audio recorded using a miking technique of the same name, wherein the Mid content is captured via a mic directly facing the audio source and the Side is captured using a “figure-8” mic picking up sound at right angles to the source.



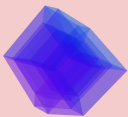
This allows independent adjustment of center and side elements, enhancing or narrowing stereo width and spatial effects in a mix. **The Side channel does not hold separate left and right information** but captures differences in the sound coming from the left and right areas of the space, enabling detailed manipulation of stereo field characteristics without focusing on microphone techniques.

Each of the three filters in Filterverse has an MS button. When activated, this button enables MS processing for that filter.



The image above illustrates the signal path when mid-side processing is used. The amplifier input and stereo panning remain outside the MS path.

For more about mid-side processing and its applications, see the [Mid-Side Appendix](#) at the end of this manual. For now, we'll just give a heads-up that Mid-Side processing in Filterverse is most audible when [stereo modulation](#) of one or more destination parameters is being applied.



WARNING:

Using M/S processing **may cause phase correlation issues** within your mixes. It is important to be mindful of the different pitfalls it may pose. Read more about it in our [M/S processing Appendix](#)

Common Filter Controls

Let's look at the control types you will always see on the interface, regardless of the type of filter selected.

Cutoff

Adjusts the cutoff-frequency of the Filter.

The filter curve corresponds to pitch. The zero in the 12 o'clock position above the knob is tuned to E5 by default, although this can be changed (see the Tuning section in Settings). Clicking on the numbers around the knob will result in jumps in octaves. This lets you tune the filter's behavior to the musical key of your source material.



Double-click on the frequency (Hertz) field to edit it.

Entering a note value (for example: “C#4” or “Eb2”) in the frequency text field will tune it to the corresponding frequency — as interpreted by our scale tuning system.

Dragging on the Semi or Cent text sliders is another way of moving the frequency knob, which always references an increase or decrease in pitch from the zero position. The Hertz and Semi/Cent fields track each other — you never again have to wonder what frequency corresponds to a certain pitch or vice-versa!



Word to the Wiz:

You can extend the range of the Cutoff knob beyond its visible onscreen limits. Simply modulate it with the Meta Knob or another [modulation source](#).

Resonance

Resonance is a feedback loop within the filter. In classic synth filters, this is heard as a peak of frequencies around the cutoff frequency, and is responsible for a “whistling” or “rubbery” character. In other types of filters including comb and [multi-peak](#) varieties, resonance can emphasize one or more different ranges of frequencies. Resonance also adds an extra decay to the sound, and at high settings it’s possible to “ping” a filter to hear this decay.



Some of Filterverse’s filters are also capable of *self-oscillating*: generating sound on their own without any signal input. Going higher than a Resonance value of 10 can result in the filter self-oscillating. Be careful with this feature.

Buffer Clear



The small button below the Resonance knob and to the right of the label clears the audio buffers in Filterverse. Click it momentarily. This can stop runaway feedback in the event that it occurs.

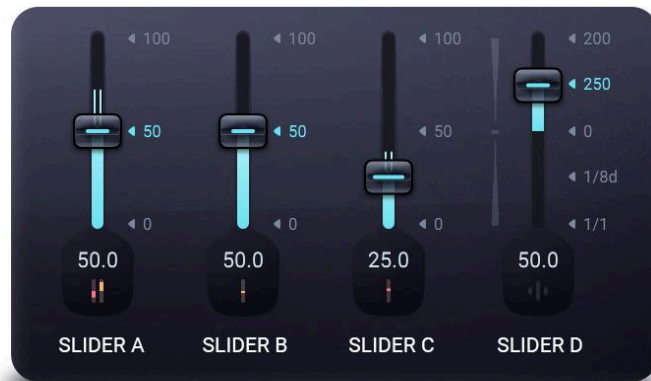
Beyond Cutoff and Resonance

In some cases, the Cutoff and Resonance knobs perform functions that play similar roles to actual cutoff frequency and resonance but are technically different. For example, many available filter models including our [Swiss Army Knife](#) and [Bread & Butter](#) filters are multi-mode. Change the [mode](#) from lowpass to bandpass and the “cutoff” frequency is now technically a *center* frequency (i.e. the center of the band being allowed to pass).

For the [multi-peak filters](#), which emphasize and notch out multiple frequency bands: the Cutoff knob determines the center around which all this action is happening.

Filter Sliders

Any filter model in Filterverse displays up to four sliders that can change functionality depending on the filter loaded. Their labels will change accordingly. Most of the time, they serve similar functions:



Slider A

This slider is typically used to select the type of filter (highpass, lowpass, or bandpass). Many of the filters are true state-variable types, meaning you can place the slider in an intermediate position to get a morph or blend of two types. Some, however, have “either-or” toggle selections on Slider A.

Slider B

This slider is typically used to select the poles of the filter, which determines how steep the filter is. It can also be used as a damp control for delays and reverbs.

Slider C

This slider is a “wild card” that can be used for various parameters, such as drive, depth, and bandwidth.

Slider D

This slider is always bipolar, meaning it can send both positive and negative values depending on whether you move it up or down from the middle position. It is used for spread, stereo width, and other effects.

Modulation

Filterverse's main parameters can all be modulated by a number of external and internal sources. You can use the various modulation sources to tune several parameters with a single knob, make the filtering responsive to MIDI or CV messages (or to the volume of the source audio using the envelope follower), modulate parameters with an LFO-like waveform or step-sequenced pattern, and more.

Modulation Slots



Eight slots are available, each of which can hold one of the eight types of modulation sources: Meta Knob, ADSR Envelope, Sequencer/LFO, Oscillator (audio-rate), Envelope Follower, MIDI/CV, Random Value Generator, and Pitch Detect.

The Modulation Source slots are color-coded to match their relevant Modulation Amount controls, on the modulation pop-up buttons and modulation depth sliders.

You can mix and match sources in the slots any way you like, including having multiple slots with the same type of source. Each slot is locked to its color regardless of the modulation source selected; slot 1 is always red, for example. So, you might have a "green sequencer", a "red sequencer", etc., and be able to tell the difference at a glance when using the [pop-up sliders](#).

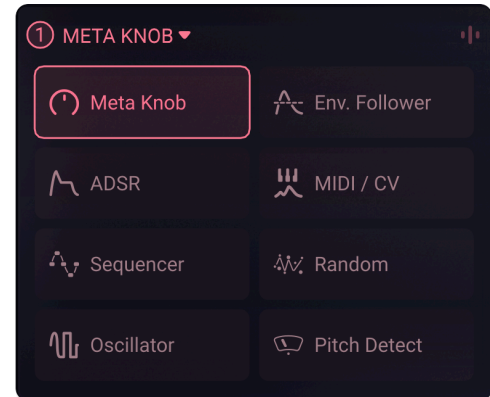
On/Off

Use this switch to enable or disable a slot. If you disable a slot, all of its settings will remain intact for the next time you enable it. A disabled slot appears to be empty and will have no effect on the sound. Once a slot is turned on, its matching Modulation Amount controls will appear in the plug-in's interface, and the modulation fun begins!

Selecting a Modulation Source

To modulate a parameter, first load a modulation source into one of the slots by clicking on its title and selecting from the choice of [available modulators](#).

The eight slots are color-coded from left to right: red, amber, yellow, green, teal, blue, purple, and magenta.

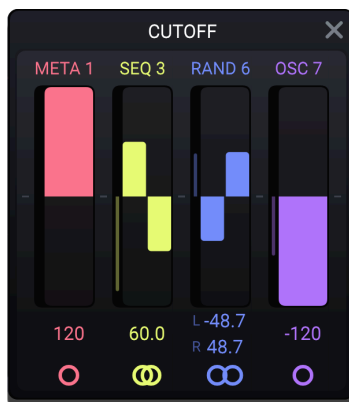


Assigning a destination

Click the Modulation Pop-up Button for the parameter you want to modulate, to open the Modulation Pop-up Sliders. This button also has a monitor allowing you to see the amount of modulation on the parameter at a quick glance.

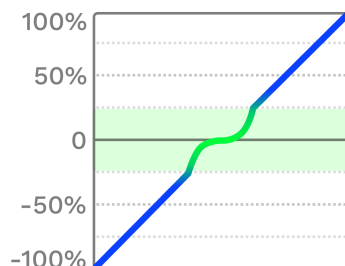


Modulation Amount



The modulation Pop-up contains modulation amount sliders. Modulation controls have the same color as the Modulation Source Slot they represent and will appear only if there is a modulation source in at least one slot.

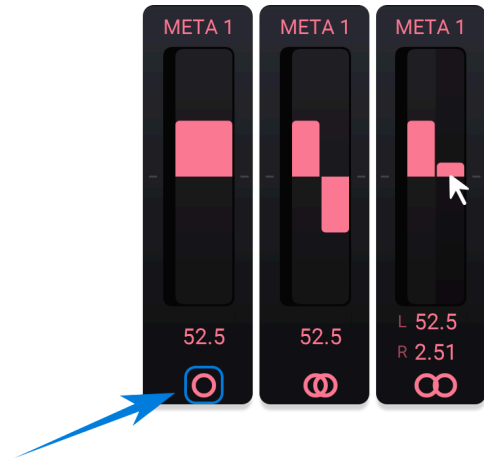
The controls are bipolar, so sliding them up to the maximum will result in positive modulation, while sliding them down to the minimum will result in negative modulation. All controls will display modulation sliders for all active sources, all the time. To ensure a given source is not modulating a parameter at all, double-click the slider to set a value of 0.



Most modulation amount sliders have a **curve** — which enables more resolution in values that are closer between zero and the 25% mark, and then linear action afterwards.

Stereo Modulation

Click the circle icon below any modulation slider to toggle modulation from that source between mono and stereo. In stereo mode, you can set different modulation depths — even one positive and the other negative — for the left and right sides of a stereo signal. This can result in some truly wild modulation effects. Want the filter cutoff to sweep up for the left side as it goes down for the right? You can do that!



Word to the Wiz:

Are the pop-up sliders in the way of other controls you want to see? You can move their window by dragging its title bar around the Filterverse interface. Click the X to close the pop-up sliders.

Modulation at a glance

Filterverse is designed to show you which sections and parameters of the plug-in are actively affected by modulation sources, via a quick look at the overall interface.

Modulation Pop-up buttons

Once a slot is on, **Modulation Pop-up buttons** will appear underneath any main control that's eligible to be a modulation source. These appear as small vertical lines that match the color of the modulation sources. Note that you will see only as many colored vertical lines as you have modulation sources active for that destination — any number from zero to eight.

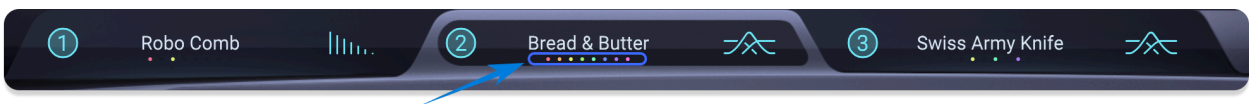


Modulation button with no mod sources active.



Modulation button with eight mod sources active.

Indicators in filter tabs



Note as well that colored dots corresponding to the modulation sources appear in the filter tabs. These let you see at a glance which modulation sources are active across all filters, even if a filter is turned off or not currently selected.

Modulation meters

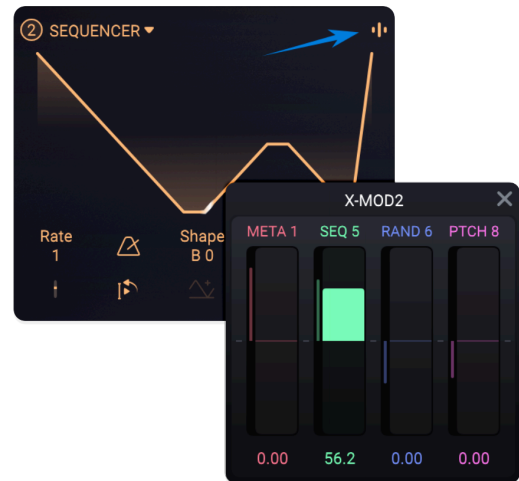
Animated blue lines (two of them for a stereo signal) in knob and slider controls will indicate the amount by which the sum total of all modulation sources is affecting that control. The modulation Indicators only work when the filter is on and active. If they appear to be stuck - it may be because the DAW has paused the audio processing.



Cross-Modulation

Can you modulate the depth (amplitude) of one modulation source with another? Of course you can! Modulation sources have their own pop-up buttons as well.

Click this button to bring up a pop-up of sliders similar to those in the main controls section. You'll see a maximum of seven sliders, as a given modulation source cannot modulate itself.

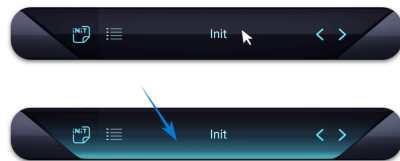


When a slot is modulated, its level will change according to the behavior of the source(s) modulating it. For example, when an envelope is being modulated by MIDI velocity, it will act like a classic velocity sensitivity slider, where more modulation means more velocity range.

Here are four more useful examples:

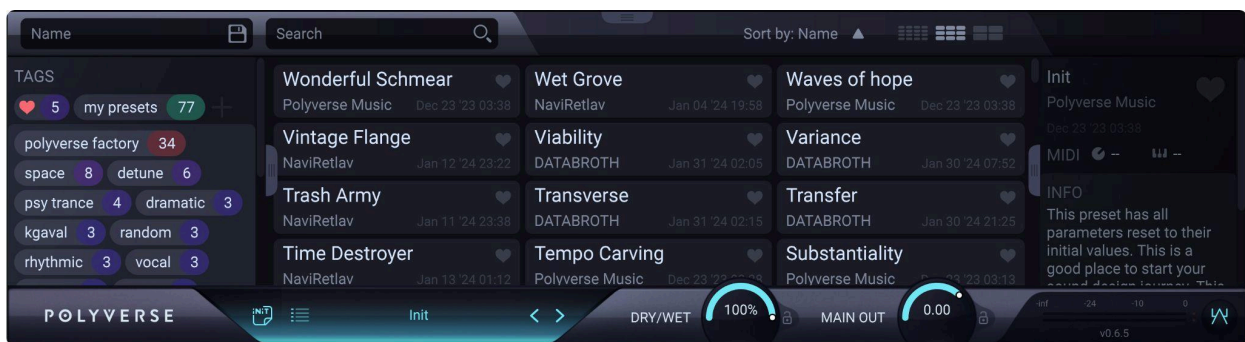
- Classic mod-wheel action: Control the depth of vibrato sourced from the LFO/sequencer using the modulation wheel (which is located in the [MIDI/CV](#) source).
- Fade in an LFO/sequencer by modulating it with an envelope.
- Control several Meta Knobs with one Meta Knob — a Meta-Meta Knob!
- A more elaborate example would be to modulate a [Classic filter](#)'s frequency parameter with an LFO and a [Multi-Peak filter](#)'s partials with an ADSR, while a single Meta Knob controls the *amount* of both modulations at the same time. The level of the Meta Knob can in turn be controlled by MIDI, aftertouch, or another performance gesture from a hardware controller.

Preset Browser



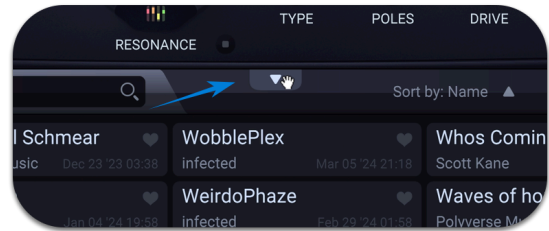
To step through Presets, click the Left/Right arrows located on either side of the Presets Bar. Clicking directly on the Preset Bar will expand the bottom half of the Filterverse interface into a full-featured Preset browser:

Here, you can browse through factory Presets, save your user Presets, or load a new preset bank.

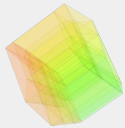
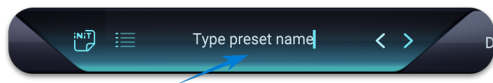


Once the button is clicked, it is possible to **browse the Presets using the arrow keys** on the keyboard.

To see more of the browser without scrolling, drag on the **light gray tab** at top center (which turns into an arrow when you hover on it) to resize the area. Click on the tab to close the browser.



To save a Preset without opening the browser, Simply right-click on the Preset name in the Preset bar. Type in the desired and press Return/Enter.



Word to the Wiz:

Saved presets can be found in the following locations:

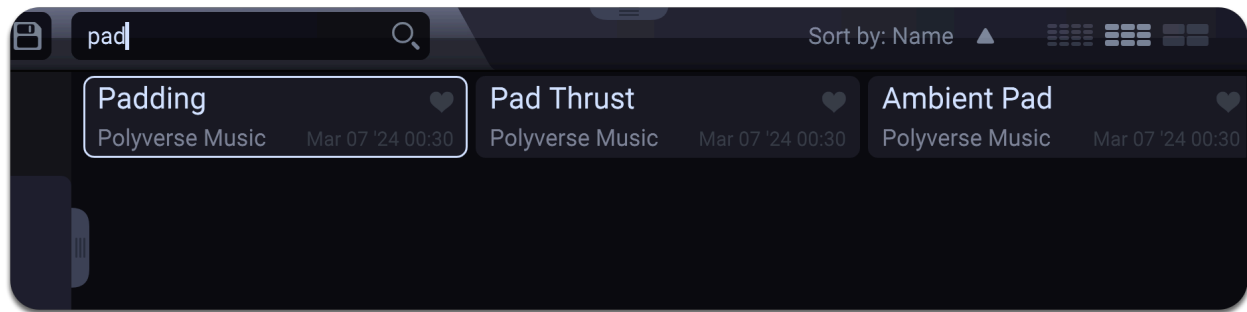
MacOS:

Users/[user-name]/Library/PolyVerse/Filterverse/Presets

Windows:

C:\Users\%USERNAME%\AppData\Roaming\Polyverse\Filterverse\Presets

Searching Presets



There are two text fields at the top of the browser: one for [saving Presets](#) and one for searching. To search for a Preset by text (in its name or author's name), type the text into the field on the right. The results will appear in the center of the browser directly below, updating in real time as you type. There is no need to press the Enter/Return key.

Selecting and Sorting Presets

You can sort presets by clicking on **Sort by**. Clicking cycles through three options: name, author, and date. Click the adjacent triangle to arrange presets in ascending or descending order.



In addition, the **panes** to the right select how densely the Presets are displayed in the results area: four across, three across, or two across.



To select and load a Preset simply click on it in the results area.

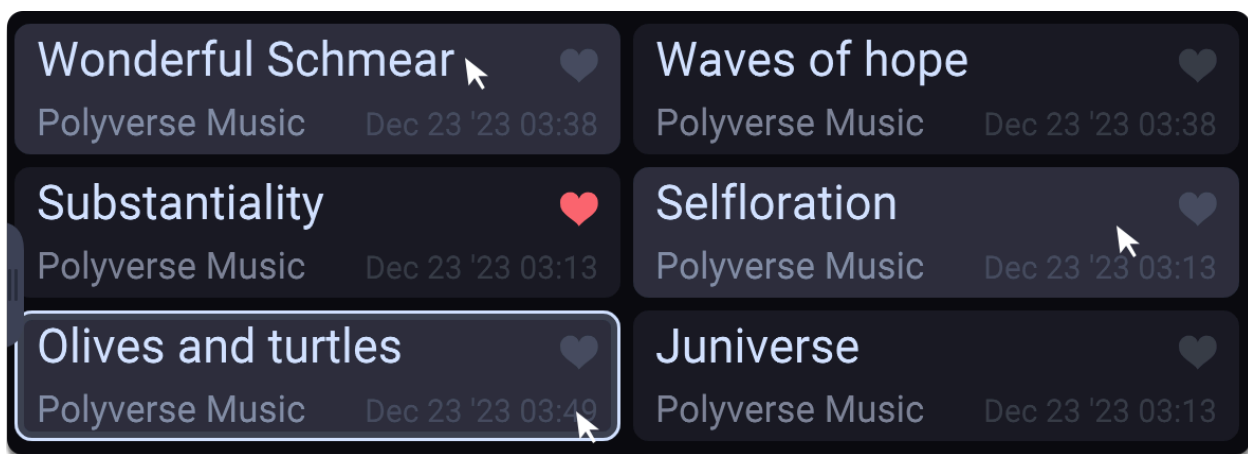
Favoriting Presets

To “like” or make a Preset a favorite, click the heart icon next to its name in the results area or the [Preset Info](#) section.

If any Presets are favorited, the heart will show up as a tag in the [Tags](#) sidebar, allowing you to display only favorite Presets with a click.

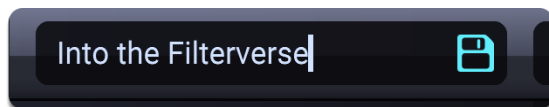


Selecting Presets



You can select multiple Presets for export or deletion. Control-click (Windows) or Command-click (macOS) to select non-adjacent presets. The background of selected Presets' turns a lighter shade of gray. To select a range of Presets (Windows and MacOS), shift-click on the first and last, and all Presets between will be selected as well. (The most recent Preset you clicked on is the one that will load and be audible.)

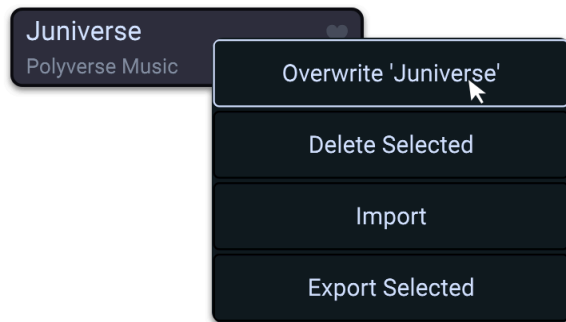
Saving a Preset



Click on the right of the two text bars (the one with the disk icon) and you will be prompted to type in a name. Press Enter/or Return to save your work as a new Preset.

This will not overwrite the Preset you have been working with; it's equivalent to a "Save As" operation.

Preset Operations Menu



Right-click on any Preset in the central results area to display a pop-up menu offering four options for Preset management:

- **Overwrite 'Preset name':** Overwrites the currently selected Preset with any changes you have made to the parameters.
- **Delete Selected:** Deletes selected Presets.
- **Import:** Opens an OS-level navigation window in which you can search for Preset files stored on your computer. Files use the extension *.preset*. You can also import a ZIP file that contains Filterverse Presets.
- **Export Selected:** Opens an OS-level navigation window where you can save selected Presets on your computer. Both individual and multiple Presets are exported as a single ZIP file.

Initialize Preset



The **Init** button on the left side of the Preset bar returns Filterverse to its default settings: three Swiss Army Knife filters routed in simple series, Cutoff at 0 semitones (12 o'clock), Resonance at 0, lowpass, 2-pole, and zero drive. Technically, it loads a Preset named "Init" that has these settings, allowing you to start from scratch.

Tags



Filterverse uses Tags to classify Presets and make them searchable. When one or more Tags are selected, the Preset results are limited to Presets that bear the Tag(s). This is in addition to how results may have been curated by a text search.

To open and close the Tabs sidebar, click the vertical gray tab on the right. Drag on it horizontally to resize.

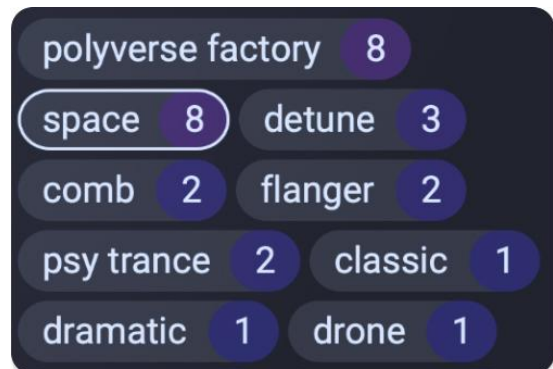
Notice the number on the right side of any tag. This refers to the number of Presets that have that Tag.

Selecting Tags

Click on any Tag to select it. Click again to deselect it.

Selecting a Tag, “space” in this example, also changes the Tag window such that the other tags are only those for the Presets also tagged “space.”

Notice that the numbers change. In the previous image, 17 Presets had the “Polyverse factory” Tag. But only eight of them also have “Space,” so that number changes to 8 here.



You can repeat this process as many times as you like to narrow the results further. For example, we now select “detune” in addition to “space” Other tags shown are only those for Presets answering to *both* of these selected Tags. Notice that the Preset counts in each Tag decrease even further.

Excluding Tags

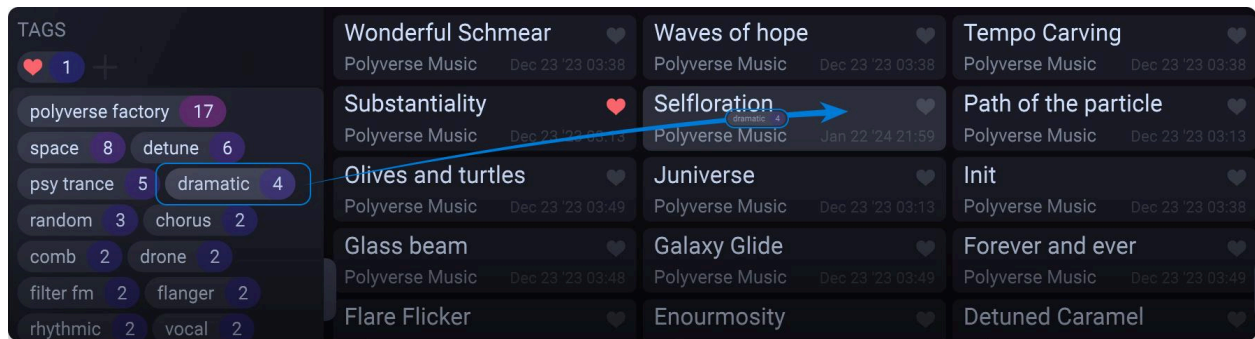
Shift-click on a Tag to *exclude* Presets bearing that Tag from your search results. The Preset count numbers in the Tags themselves will change accordingly.



Visual cues for Tags

- Tags you've actively selected are outlined in white.
- Tags that remain as a result of your selections are not outlined.
- Tags you've excluded are outlined in red.
- Tags are displayed, left to right and then top to bottom, in descending order of the number of Presets that have a given tag. When two or more tags have the same number, the order becomes alphabetical.

Adding a Tag to a Preset

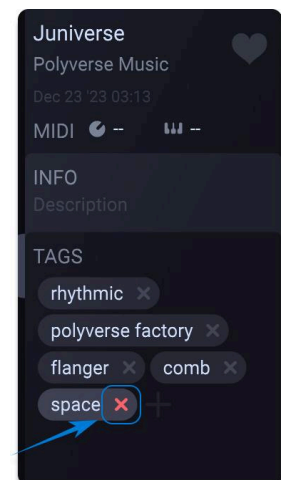


Simply drag and drop any Tag from the Tags sidebar onto a Preset in the central results area. There is another way to add tags in the [Preset Info](#) area.

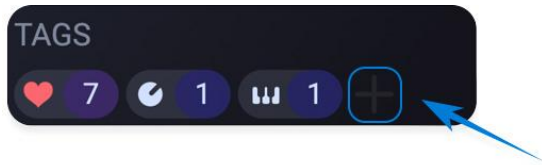
Removing a Tag from a Preset

Removing a tag is done in the [Preset Info](#) sidebar on the right side of the Preset browser.

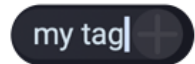
Hover over the X next to the tag name and it will turn red. Click the X to remove the Tag from the currently selected Preset.



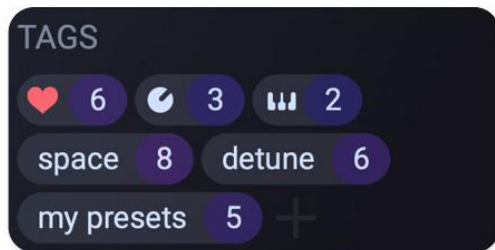
Creating a Tag



You can create your own tags. Click the + sign in the header of the Tags sidebar and it will expand into a field where you can type text. Name your tag and press Enter/Return.



My Tags



We refer to the header of the Tags sidebar as the “My Tags” area. You can click-drag tags from the lower area up here because, for example, you use them the most often.

Special Tags



Up to three special Tags may appear if one or more Presets is tagged with any of them. These always appear in the header at the top of the Tags sidebar. Otherwise, what they do works like any other tag.

- **Favorites (heart icon):** Limits search results to favorited Presets.
- **MIDI CC (knob icon):** Denotes Presets enabled for [MIDI program change](#) by sending a value on CC 119.
- **MIDI note (keyboard icon):** Denotes Presets enabled for MIDI program change when Filterverse receives a MIDI note.

We have already covered how to make a Preset a [favorite](#). The ability for a Preset to load via the MIDI CC and note tags is discussed in the [Preset Info](#) and [MIDI Program change](#) sections.

The MIDI CC and MIDI note tags are the only two that cannot be dragged and dropped onto a Preset, as they are turned on per Preset in the Preset Info section.

Tag-Based Operations

Right-click on any Tag in the left sidebar to bring up the above menu, which offers three management options at the Tag level.

- **Import:** Imports a Preset file (or ZIP file containing Presets) and applies the selected tag to all Presets imported.
- **Export:** Exports all Presets bearing the selected Tag as a ZIP file.

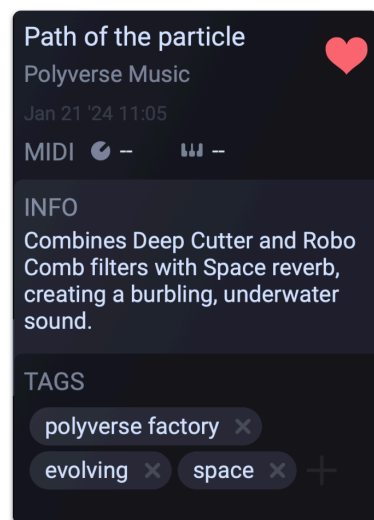


When a Preset is imported, the tag “imported” is automatically added. If the tag “my presets” was originally present, it is removed.

Preset Info

The sidebar on the right of the Preset browser displays and edits information about the currently selected Preset. As with the Tags sidebar and Preset browser overall, it has a central gray handle for opening/closing and resizing.

Note: Attributes in the Preset Info sidebar are all “hot editable.” That means that if you change something in one of the areas described below, your change will be retained even if you switch Presets then switch back. In other words, there is no need to perform a Save or Overwrite operation once you’ve edited something here. With that in mind, here are the functions/info areas from top to bottom:



Preset Name

Click on the top line of text to edit the Preset name. Press enter/return to save your changes.

Preset Author/Designer

The line of text just below the Preset name is for an author or designer, though you could conceivably use it for other information. Click on it to edit and press enter/return to save.

Heart Icon

To the right of the Preset name and author is a heart icon for favoriting that Preset. This is a mirror of those in the Preset list in the search results area.

MIDI Program Change

You can send program changes to Filterverse on MIDI CC 119, allowing you to change Presets using a hardware controller. You can also send changes using a MIDI note, letting you set up a zone of a MIDI keyboard for “key-switching” Presets. To use this functionality, make sure Filterverse is set up to receive MIDI [in your DAW](#).

Filterverse enables either method Preset by Preset. Here’s how to set it up.

Program Change via CC

Program change messages can be sent via CC 119. To use this functionality, set the desired knob, slider, or button on your hardware controller to transmit on CC 119.



- Click the knob-shaped icon in the [Preset Info](#) area to automatically assign the lowest available value for CC 119 as the program change trigger.
- To assign a custom value, simply type it directly into the adjacent number field.

Remember, the number you're typing in is *not* a CC — it's the trigger **value** for a program change. Now, whenever Filterverse receives that value from your controller, it will load the Preset.

Program Change via Note



- Click the keyboard-shaped icon in the [Preset Info](#) area so that it illuminates.
- Type in the value of the MIDI note you would like to trigger the program change. You can either type in a “piano style” reference (e.g. C-1) or the MIDI value (0-127). The possible range is from C-1 to G9 — a total of 127 possible values. Press enter/return.

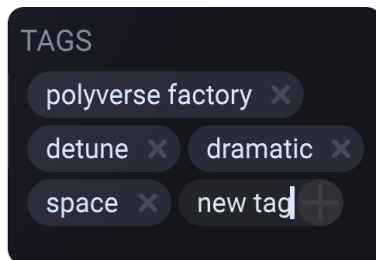
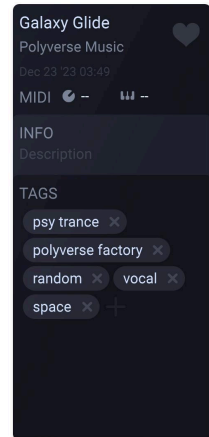
Now, striking the corresponding key will load the selected Preset. Note that in Step 4, the initial note displayed will be the lowest available, i.e. that has not already been assigned to trigger a different Preset.

Info

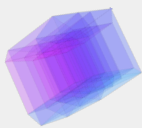
Here, you can enter descriptive notes about the current Preset. Just type in the desired text and hit enter/return to save.

Tags in Preset Info

The bottom section displays all Tags applied to the currently selected Preset. You may remove a tag from the Preset by clicking the red X next to its name.



Click the + sign that appears at the end of the Tag list to add/create a Tag for the current Preset.



Word to the Wiz:

Everything in the Filterverse Preset browser is editable and overwritable — even factory Presets and Tags. By default, factory presets have the tag “Polyverse factory.” If you overwrite or delete any factory Presets accidentally, don’t worry. You can always download the factory Presets from the Filterverse support page at polyversemusic.com/support.

Settings



Click the gear-shaped icon to open the Settings panel, where three useful categories of settings reside: General, Tuning and MIDI.



These are global settings, meaning they affect all Presets equally and do not change at the Preset level.

General Settings

These govern the size of the Filterverse plug-in window and the behavior of its controls.

GUI Scale

You can resize the Filterverse window from 50% to 200% in five steps. Smaller sizes are good for situations where you may want to cram a lot of plug-ins on screen. Larger ones are ideal for multi-display setups or working with Filterverse as a sound design tool all by itself.



You may also change the scale using the following key combinations:

- increase size: “ctrl =” (Windows) or “command =” on (macOS)
- reduce size: “ctrl -” (Windows) or “command -” on (macOS)

CPU/Quality

This controls the CPU intensity of the plug-in, with low, medium and high settings, high being the most detailed. Note that this is not a global setting; it is saved *per instance* (not per Preset) within a project. So, you could have different tracks using separate instances of Filterverse, running at different CPU use settings.

Knob Mode

This governs the response of the main Cutoff knob as well as any other knob-shaped control in Filterverse’s interface (such as those in the modulation sources).

Round: Change the frequency by dragging around the circumference of the knob, almost like you were turning a physical knob.

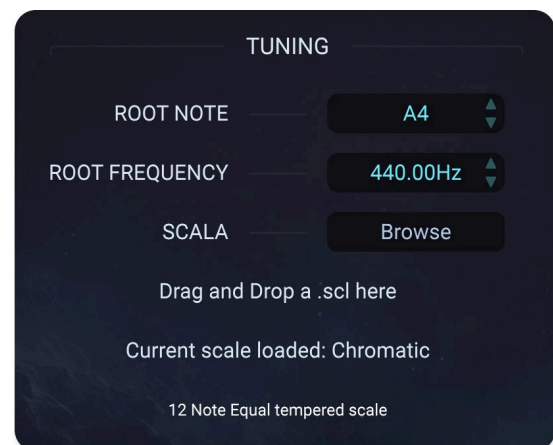
Linear: Change the frequency by dragging up and down on the knob.

Tuning

As we have seen, Filterverse is a very musical filter plug-in that “thinks” in terms of musical notes. Here, you can adjust the basis of that thinking, so to speak.

Root Note

In terms of pitch, this changes the musical note that corresponds to the zero (12 o’clock) position of the Classic Filter’s main [Frequency](#) knob, as well as the root note for the Scale parameter of the [Random Generator](#), which is in turn meant to affect the Frequency.



Root Frequency

You can also adjust the root pitch in terms of frequency in Hertz.

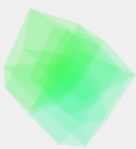
Root note and frequency are two values that tie between pitch and frequency. Western music uses a chromatic scale (12 semitones are 100 cents apart) with A4 corresponding to 440Hz as a standard. This however can change according to different musical styles and cultures. Authentic baroque musicians often tune their instruments to A4=415Hz. Some people believe in the healing powers of A4=432Hz. Scientific pitch is set by C4=256Hz, and ethnic music often is measured by the root note of flutes and hard-to-tune instruments.

The center of the frequency knob is always set to C5. The frequency of that note is determined by the root note, the root frequency, and the scale of the tuning system.

Scala

Filterverse supports Scala tuning files. These are files that determine the scale based on the selected root note, and have the extension .scl. Loading a Scala file that matches your source material can extend the sound-sculpting power of Filterverse, which uses the western equal-tempered chromatic scale by default.

Click “Browse” to open an OS-level dialogue and navigate to any Scala files on your computer. Or, simply drag and drop one from your desktop to the designated area in the Tuning settings.



Word to the Wiz:

To read more about scala:

<https://www.huygens-fokker.org/scala>

Download over 4000 different scales:

<https://www.huygens-fokker.org/docs/scales.zip>

MIDI

Voicing

Mono: Filterverse's envelopes retrigger on every new note. Glide will occur between all notes.

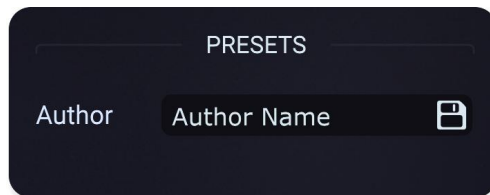
Legato: Envelopes do not retrigger. Glide will occur only when overlapping notes.



Pitch Wheel

Tune the range of the pitch wheel (when using MIDI notes as a modulator) in cents. You can set upward and downward wheel movement separately.

Presets



Here, you can set the default author name that will be recorded in the [Preset Info](#) when you save a Preset.

Appendix 1: Filter Reference

Filterverse offers a wide variety of filter types for each of its three filter tabs. They're divided into categories such as Classic, Multi-Peak, and "Non-Filters" (other effects which nonetheless have sound-sculpting abilities), and more.

Word to the Wiz:

You'll often hear filters described as having a certain number of **poles** or **dB per octave**. In theory, the more poles a filter has, the steeper the slope with which it reduces the volume of frequencies outside the cutoff. In practice, and especially in Filterverse, the reality can diverge from this, especially with multi-peak filters and other compound or sophisticated types. That said, that slope is expressed as a number of decibels (dB) for each octave, and these are the traditional correspondences:



Poles	Gain reduction in dB per octave
1	6dB
2	12dB
3	18dB
4	36dB
8	48dB
12	72dB
16	96dB

Classic Filters



The classic filter collection contains filter models that employ the modes most commonly associated with synthesizers: low-pass, highpass, Band-pass, Band-reject (notch), and Peak (all-pass). Within this collection there are many different topologies and flavors. Some filters are steep, some are gentle. Some are linear, and some are not.

While Filterverse has familiar filter topologies such as the ladder filter or the state-variable filter, our implementations do not attempt to recreate, clone, or resurrect any hardware versions of these filters. Instead, we reimagined such filters, optimizing them to sound as good as possible while emphasizing different characteristics of each for a maximally wide palette of timbres.

Most of the filters in this collection are *non-linear*. This means they have a set headroom and can therefore saturate if fed with high gain. Each filter model sounds different according to the level of audio pushed into it. In each filter topology this nonlinearity will manifest in a way that's unique to that filter, giving it a distinct sound. Try driving filters with and without resonance to hear the difference. Since saturation involves an inherent degree of compression, we invite you to use that extra 18db boost to drive filters — on purpose!

Two of our classic filters (Brick Wall and Clear Glass) are *linear* and do not saturate. Instead, they convey every dB of gain to the output with no inherent compression. This requires careful consideration when changing filters while signal is playing through them, as levels can spike dramatically if you change from a filter that saturates to one that does not.

All of the filters in this section are optimized for the audio-rate modulation that can be provided by the [Oscillator](#) modulation source.

Swiss Army Knife

Self-oscillation: Yes

Saturation: Yes

Types: Lowpass, band-reject (notch), bandpass, peak (Allpass), highpass

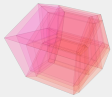
Poles: 2, 4

Drive: Screaming overdrive

A state-variable filter with a juicy sound and internal saturation. It has five types of filters in 2-pole (12dB per octave) and 4-pole (24dB per octave) slopes.

Adding more resonance will not reduce the amplitude of the bass frequencies. Pushing gain against the internal saturation shifts the position of the resonance in relation to the audio and provides a generally fatter sound.

The resonance peak on this filter is rather wide.



Word to the Wiz:

A “state-variable” filter is one in which the topology (lowpass, bandpass, highpass, etc.) can change in real time and even take on intermediate states between two topologies. It was popularized by Oberheim’s SEM and Matrix-12 synthesizers.

Type

Slider A smoothly crossfades between five states. From slider low to high: lowpass, band-reject, bandpass, peak (allpass), and highpass.

Poles

Slider B smoothly crossfades between 2-pole (12dB per octave) and 4-pole (24dB per octave) filter slopes.

Drive

Slider C pushes gain into the output saturation module, with extra pre/post filtering to ensure a screaming overdrive sound.

Bread & Butter

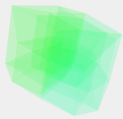
Self-oscillation: Yes

Saturation: Yes

Types: Lowpass, bandpass, highpass

Poles: 1, 2, 3, 4

Bread & Butter is a sweet sounding ladder filter with internal saturation. Adding resonance will slightly reduce the amplitude of the low end - leaving more headroom for the resonance peak. This is why pushing gain against the internal saturation will not displace the resonance, but it will result in a satisfying purr. The resonance peak on this filter is rather narrow.



Word to the Wiz:

The ladder filter was invented and popularized by Robert Moog with the Minimoog synthesizer and the 904-A filter module. This topology is known for its exceptionally musical sound and is one of the most popular filter designs today. Our Bread & Butter filter can emulate but is certainly not limited to this character.

Type

Slider A smoothly crossfades between three states: lowpass, bandpass, and highpass.

Poles

Slider B smoothly crossfades between 1-, 2-, 3-, and 4-pole slopes.

Drive

Slider C pushes extra gain into the input, the output, and an internal saturator — with no pre/post filtering for a creamy saturated sound.

Deep Cutter

Self-oscillation: Yes

Saturation: Yes

Types: Lowpass, bandpass, highpass

Poles: 2, 4, 6, 8, 12, 16

This is a Butterworth filter (optimized for a smooth slope) with internal saturation. Adding resonance will not reduce the amplitude of the low end.

Pushing gain against the internal saturation will not displace the resonance. Instead, a “warble” effect is achieved. The resonance peak on this filter is of medium bandwidth.

This filter starts with two poles (12db per octave) and goes all the way up to 16 poles (96db)! Using higher numbers of poles introduces a substantial amount of phase shift, causing a pitch-shifting effect when modulated.

Type

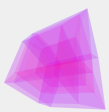
Slider A smoothly crossfades between three states: lowpass, bandpass, and highpass.

Poles

Slider B switches the number of poles between 2 , 4, 6, 8, 12, and 16.

Drive

Slider C pushes gain into the output saturation module - as well as the different filter stages for a fuzzy sound.



Word to the Wiz:

Filterverse’s Butterworth filter is a filter designed to have as flat a frequency response as possible in the band that is allowed to pass. This results in a very smooth slope.

Brick Wall

Self-oscillation: No

Saturation: No

Types: Lowpass, highpass

Poles: 6, 12

A super-steep Chebyshev type filter. This filter has a steeper curve than the Butterworth filters, making its 12-pole setting much steeper than the 16-pole Butterworth setting in [Deep Cutter](#).

This is the steepest filter offered in Filterverse.

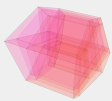
We chose to make this filter linear and non-saturating so that it can be used as a precision sound-sculpting tool. This means that pushing gain into the input will do nothing more than result in more gain at the output. The gain will not affect the timbre or character of the sound.

Type

Slider A switches between two states: lowpass and highpass.

Poles

Slider B switches the number of poles between 6 and 12. Brick Wall is also an exception to the usual roles relating poles to slopes: Its 12-pole filter is much steeper than the 16-pole Butterworth option in *Deep Cutter*.



Word to the Wiz:

A Chebyshev filter exhibits a steeper rolloff than a Butterworth filter. Also note that the Brick Wall filter does not saturate but has unlimited headroom. So, be careful with the gain — for example, when switching from a saturation-capable filter in which you've been pushing the gain to add character.

Clear Glass

Self-oscillation: No

Saturation: No

Types: Lowpass, peak, highpass

The Clear Glass filter features a linear amplitude response. Its primary strength lies in its transparency. It has a slope of 12dB per octave.

When the filter is in an open state, it bypasses the signal entirely, ensuring no coloration of the sound. This makes it especially suited for master mixes, due to the smooth transition between the bypassed and filtered states.

We chose to make this filter linear and non-saturating for reasons of precision. This means that pushing gain into the input will do nothing more than result in more gain at the output. More or less gain will not affect the frequency response and character of the sound.

When the Clear Glass filter is completely open in lowpass or highpass states, the filter curve is completely flat. Likewise for the peak state when no resonance is applied.

Furthermore, Clear Glass handles audio-rate modulation exceptionally well.

Type

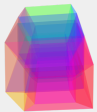
Slider A smoothly crossfades between three states: lowpass, peak, and highpass.

Multi-Peak Filters

The Multi-Peak Filters category is home to filters that manipulate audio by introducing multiple resonant peaks and notches in the frequency spectrum. This creates complex tonal and textural changes. The unifying characteristic of this category is the ability of each filter to affect multiple frequency bands simultaneously, either boosting or attenuating them. This results in rich harmonic content and dynamic spectral shaping.

Unlike the Classic category of filters, the Multi-Peak filters are exotic and diverse. They are more likely to produce wild and varied sonic landscapes, from enhancing natural resonances in the source material to simulating vocal-like qualities.

This category will help you discover sounds from uncharted galaxies of the Filterverse. Be bold, explore, and you'll be rewarded with the rarest of sonic gems!



Word to the Wiz:

One of the most common multi-peak filter types is the comb filter, so named because its tightly-spaced peaks and valleys in the frequency response resemble the teeth of a comb.

Dueling Bands

Self-oscillation: Yes

Saturation: Yes

Poles: 1, 2, 3, 4

Dueling Bands blends two bandpass filters, one of which is phase-inverted. When their cutoff frequencies come close to each other, the filters cancel each other out, leaving a *very* narrow bandpass.

This filter features both internal saturation and post-filter drive. While internal saturation affects each band separately, the post drive applies to both bands simultaneously, generating complex and engaging intermodulation sidebands for a richer sonic character.

Poles

Slider A smoothly crossfades between 1, 2, 3, and 4-pole slopes.

Spread

Slider B moves the dual cutoff frequencies of the filters apart for a gap of up to ten octaves.

Drive

Slider C pushes extra gain into the output post-filtering. As opposed to saturating the filters using the input gain, the Drive parameter in Dueling Bands adds intermodulation grunge to the character of the output.

Ripples

Self-oscillation: No

Saturation: Yes

Types: Lowpass, highpass

Peaks: 2, 3, 4, 5, 6

Ripples is a unique Chebyshev-type filter that combines several filters at different cutoff points. This design achieves an exceptionally steep filter curve. Adding resonance to Ripples creates a wave-like effect across the frequency spectrum. The “ripples” themselves are narrow peaks and valleys in the frequency response leading up (or down) to the cutoff frequency. These resonance points become closer together in the frequency spectrum as they approach the cutoff, creating dynamic and rich sonic textures.

Use the Resonance knob to adjust the intensity (amplitude) of the peaks and valleys.

The internal saturation of this filter is soft and warm. Pushing extra gain into the filter will slightly shift the resonance positions, helping to generate a warm and balanced tone — even when pushed to its limits.

Type

Slider A control toggles between highpass and lowpass filter types.

Peaks

Slider B toggle-selects between 2, 3, 4, 5 and 6 peaks and valleys.

Drive

Slider C pushes extra gain into the different stages of the filter for a creamy saturated sound.

Double-Slit

Self-oscillation: No

Saturation: Yes

Types: Lowpass, band-reject (notch), bandpass, peak, highpass

Peaks: 2, 3, 4

The Double-Slit filter, inspired by the legendary double-slit experiment in physics, combines two mirroring Ripple filters to create a complex matrix of interference patterns. This setup mimics the wave-like interference seen in the physics experiment, resulting in complex phase interactions within the audio. These interactions produce nodes and antinodes, akin to the bright and dark fringes in the experiment, crafting unique and intricate sonic textures.

The **Spread** parameter functions similarly to adjusting the distance between slits in a double-slit experiment. Altering this spread parameter effectively changes the distances between the two Ripple filters, which in turn modifies the diffraction patterns produced. This adjustment leads to varied phase interactions and “interference patterns” in the audio signal.

Type

Slider A smoothly crossfades between five states: lowpass, band-reject, bandpass, peak (allpass), and highpass.

Peaks

Slider B selects between 2, 3, or 4 peaks and valleys *per filter*. When both filters are combined, intricate interference patterns appear.

Drive

Slider C pushes extra gain into the output stage of the filter for a screaming sound reminiscent of a human vocal tract.

Spread

The spread parameter adjusts the distance between the cutoff frequencies of the filters — akin to setting the distance of the slits in the double-slit experiment — creating an interplay of two resonance patterns.

All-Pass Filters / Phasers



Coming Soon...

This category includes various types of phasers and all-pass filters. It features standard phasers, nested and contrary phasers, and specialized filters like the Schmeear, which employs a long chain of all-pass filters for pronounced subgroup delays. Each filter alters phase relationships in the audio signal, offering different effects from subtle phase shifts to more complex transformations.

Comb Filters / Flangers



Comb filters are a type of filter used in signal processing that create a series of peaks and troughs in the frequency response, resembling the teeth of a comb. This effect is achieved by adding a delayed version of a signal to itself, causing constructive and destructive interference. The result is a distinctive pattern of sound reinforcement and cancellation at regular intervals across the frequency spectrum. Comb filters are versatile in their application, capable of producing a wide range of audio effects from subtle phase shifts and flanging to more pronounced resonant and rhythmic patterns.

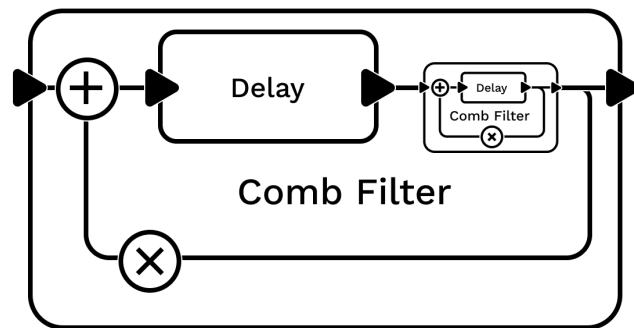
Robo Comb

Self-oscillation: No

Saturation: Yes

Types: 0:1, 0:2, 0:3, 0:4, 0:5, 1:1, 1:2, 1:3 1:4, 1:5, 2:2, 2:3, 2:4, 2:5, 3:3, 3:4, 3:5, 4:4, 4:5, 5:5

Robo Comb uses comb filters nested in each others' signal paths, then combines the delayed and original signals, to create a highly complex comb filter effect. It features two nesting layers — external and internal — allowing up to six comb filters in each layer for a maximum of 36 filters.



This setup offers 20 different filter combinations, providing flexibility in configuring both the external and internal nests. Think of the *external* nest as the “main” delay line. *Each* delay in that line has a feedback path (denoted by the X in the image above), and you can insert up to six *more* delays in each path. That’s the *internal* nest.

Robo Comb excels at producing pronounced vocal-like effects, special flanger effects, and fractal-like comb filter patterns. Users can adjust the symmetry between peaks, enabling a wide range of sonic textures from simple comb filtering to more complex, modally rich sounds.

Type

Slider A switches between 20 states to select the number of delays. The first number in each pair refers to the external nest; the second refers to the internal nest.

1:2, 1:3, 1:4, 1:5, 1:6,
2:2, 2:3, 2:4 2:5, 2:6,
3:3, 3:4, 3:5, 3:6,
4:4, 4:5, 4:6,
5:5, 5:6,
6:6

Damp

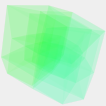
Slider B controls the amount of high frequency damping in the comb filters. This is achieved by placing a low pass filter in the feedback loops of the comb filters. This rounds the sound a bit reducing the edge of the high frequency resonances

Symmetry

Slider C controls adjusts the balance between the delays of the external and internal nests, and results in the frequency response peaks clustering towards one side of the frequency spectrum or the other.

Stretch

Slider D stretches the distance between the nested peaks across the spectrum, or pinches them together. Its interaction with the Symmetry control can create a vast array of filtering characteristics.



Word to the Wiz:

We love to tell you what's going on "under the hood" in Filterverse, but if the advice "play with the controls until you hear something you like" is ever appropriate, it applies to the Robo Comb filter!

Cliff Hanger

Coming soon ...

Flanger

Coming soon ...

Non-Filters



The “Non-Filters” category encompasses a variety of audio processing tools that, while not conventional filters, offer filter-like modulation and sound-shaping capabilities. This category is a collection of unique sound processors including variations of reverbs, pitch shifters, ring shifters, decimators, and granular effects. These tools are designed for creative sound manipulation, blurring the lines between traditional filter effects and other forms of audio processing. The category provides a playground for experimental and innovative sound design, allowing users to explore and manipulate audio in unconventional ways.

Space



Self-oscillation: Yes

Saturation: Yes

Space is a unique adaptation of our renowned Comet Reverb. However, it functions more like a filter than a traditional reverb effect.

In this design, the parameter typically known as *size* in reverbs is reimagined as the *cutoff frequency*, with an inverse relationship: lower frequencies correspond to a larger size and higher frequencies to a smaller size.

The Resonance control adjusts the reverb decay, offering a wide range of sonic possibilities.

This innovative approach allows Space to produce a broad spectrum of effects, from flanger-like modulations to elastic, liquid-like sounds. Of course, it can produce more familiar — and very lush-sounding — reverbs as well!

Damp

Slider A dampens the high frequencies and rolls them off in the feedback path (the portion of Space’s output sent back into its input). The higher the Damp setting, the darker the reverb will sound over time. High decay values intensify the damping effect.

Defuse

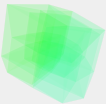
Slider B sets the rate of which the density of reflections build up in the reverb. This makes the reverb more dense and lush when increased or more airy and sparse when decreased.

Detune

Slider C detunes the tail of the reverb increasingly as time passes. An internal modulation mechanism (not unlike a synthesizer's detune effect) is calibrated to create especially smooth clusters of pitch. A little bit of detune goes a long way.

Pre Delay

Slider D delays sets the amount of time delay before Space begins processing the sound. As with most reverb plug-ins, this is useful if you want notes, drum hits, words, or whatever is on your source track to have a crisp, articulate attack phase. This is a bipolar slider. Move it down from 0, and the pre-delay is expressed as a division of a musical bar synced to your project tempo. Move it up, and pre-delay is expressed in milliseconds.



Word to the Wiz:

It is possible to play pseudo-melodies with the Space reverb tail. You can do this by modulating the Cutoff with either MIDI notes (using the MIDI/CV modulation source) or the Pitch Detect modulation source.

String

Coming soon ...

The String processor applies the properties of a plucked-string physical model to the source signal, using a waveguide algorithm.

Springs

Coming soon ...

This non-filter type applies the frequency response characteristics of a spring reverb — like those found in some classic guitar amps — to your sound.

Tension

Slider A adjusts the tension of the virtual springs, with higher tension resulting in a “tighter” sound with a quicker decay.

Damp

Slider B attenuates the high-frequency response of the virtual springs.

Stereo Image

Slider C adjusts the stereo image from mono output (into whatever stereo processing Filterverse may be applying downstream) to a full stereo image as though the signal were passing through two spring “tanks” with subtle variations between them.

Barber Poles

coming soon..

Barber Poles is a frequency shifter. It is also capable of producing “Shepard’s Tone” effects, a psychoacoustic illusion in which a pitch seems to be infinitely rising or falling even though its fundamental frequency stays unchanged.

Tempolay

Tempolay is a tempo delay that smoothly crossfades between tempo divisions. This means that modulating the delay time is possible, and it will always stay on time. The feedback path of this delay is linear until -3dB at which point it starts to saturate. To achieve a clean delay, feed it with audio that reaches up to -3dB in volume.

Time

In Tempolay, the Cutoff knob is repurposed as the delay time Parameter. A musical note appears in the center of the knob to illustrate the rhythmic subdivision closest to the set time. This can change if Time is being modulated.

Feedback

The Resonance knob is likewise repurposed as a Feedback control, determining the amount of processed signal sent back into the input.

Low Damp

This is a highpass filter in the feedback path. It reduces the low frequency content in every repetition of the delay.

High Damp

A lowpass in the feedback path - reduces high frequency content in every repetition of the delay.

Cross Feedback

Crossfades between self feedback per side - and cross feedback (left feeding right and vise-versa). This control is only relevant when the delay times between the sides are different.

Div

Slider D selects between “straight,” dotted, and triplet rhythmic subdivisions.

Bypass

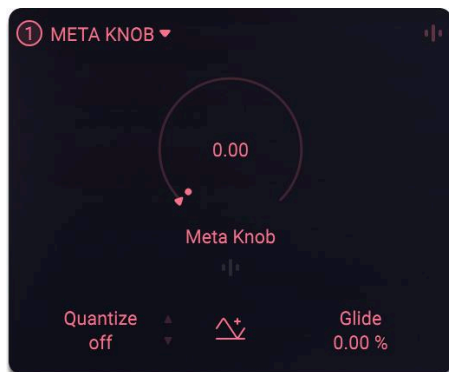
In the Bypass filter type, the Cutoff and Resonance knobs do not affect the sound, and there are no sliders. Why is it here at all? So you can use the panning, phase and mid-side features without any filtering, or mix in some dry signal to a more complicated routing scheme.

Appendix 2: Modulation Sources

Now, let's have a look at each type of modulation source in detail.



Meta Knob



The Meta Knob allows you to modulate several parameters with a single knob.

Polarity

The buttons across the bottom select the knob's type. Knobs can be *positive-unipolar* (ranging from zero to 100% and marked "+"), *negative-unipolar* (ranging -100% to zero and marked "-"), or *bipolar* (ranging -100% to 100% and marked "B").

Name

You can give a Meta Knob a meaningful name to clarify its action in a preset. To rename a knob simply double-click on its title (which is right below the actual knob) and type in a new name.



Value Monitor

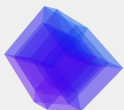
The current output of a knob is displayed as a percentage value at the center of the knob. Changing the knob's output is as easy as double-clicking the number and entering a new value.

Quantize

The Quantize field causes the Meta Knob to modulate the Cutoff and/or Resonance in quantized amounts or according to a selection of various [musical scales](#) and intervals. Drag up and down on the field value to select a scale. At “Off,” the Cutoff and Resonance move completely smoothly. At other values, either will jump the interval of the next note in the selected scale. “Next note” can mean up or down, depending on which direction you move the Meta Knob as well as the Polarity setting. **Important:** In order to have the Cutoff track the scales correctly the modulation amount should be set to 120 (100%).

Glide

Glide changes the time it takes for the values of destination parameters to change in response to you moving the Meta Knob. In terms of what you hear, this smooths the motion. At 0%, there is no lag. This is sometimes referred to as “slew.”



Word to the Wiz:

Want another modulation source to be quantized and/or use Glide? Set up the desired Quantize and Glide values in the Meta Knob, then [modulate the Meta Knob](#) with the other source. That source will now modulate the final destination parameter by way of the Meta Knob, thus applying its Quantize and Glide settings!

ADSR

The ADSR is a classic four-stage envelope generator, **triggered by incoming MIDI notes**. To learn more about connecting MIDI to the plug-in please refer to the [MIDI control](#) section of the manual

Attack

Sets the amount of time it will take for the envelope signal to climb from zero to 100% once a MIDI note is received.

Decay

Adjusts the amount of time it takes for the envelope signal to drop from 100% to the sustain level. The Decay stage is initiated immediately after the Attack stage is completed.

Sustain

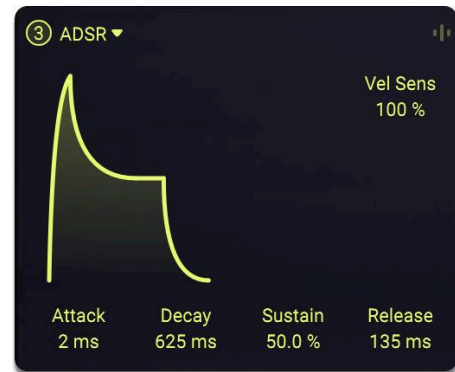
Sets the level at which the envelope signal will be held after the decay stage is completed, for as long as the incoming MIDI note is held.

Release

Determines the amount of time for the envelope amplitude to drop back to zero after the MIDI note is released.

Velocity Sensitivity

Controls how much the incoming velocity of MIDI notes will affect the overall output level of the envelope. High values will be more dynamic (i.e. respond more to velocity), while 0 will always play the envelope at full scale.



MIDI / CV

This very useful modulation source converts MIDI or CV (Control Voltage) into a modulation signal. Simply highlight the type of modulation to use:

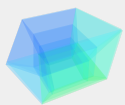


MIDI

- **Note:** A bipolar value based on the pitch of incoming MIDI notes, E4 (note number 64) being the center pitch (modulation amount = zero). Notes above E4 result in positive modulation amounts; notes below E4 send negative amounts. The note signal also incorporates a ranged pitch wheel offset that can be set in the [MIDI section of the settings menu](#). **Important:** To have the cutoff control track the pitch - its modulation amount must be set to 12 (semitones).
- **Velocity:** A unipolar value mapping modulation depth to playing velocity (0-127) on a keyboard or pad controller.
- **Pitch Wheel:** A bipolar function that sends positive modulation amounts based on upward bends and negative amounts based on downward ones. This produces a full scale control.
- **Modulation wheel:** Unipolar and positive; maps the modulation amount to the mod wheel position or value of MIDI CC 1.
- **After Touch:** Unipolar and positive; maps the modulation amount.

CV

- **Sidechain CV L/R:** This treats a signal routed into the plug-in's sidechain input as if it were control voltage, with greater amplitude amounting to greater modulation depth. CV can be bipolar or unipolar. (Note: Sidechain routing varies greatly between DAWs, check your DAW's documentation for details.)
- **CV In L/R:** This treats the signal on the track in which Filterverse is inserted as if it were control voltage. Greater amplitude produces greater modulation depth. In effect, this lets the track self-modulate any destinations assigned to the MIDI/CV source.



Word to the Wiz:

Different DAWs have different ways of routing MIDI notes and control

messages to an effect plug-in residing on an audio track. Consult the section “[With or Without MIDI](#)” for details on how to do this.

DAW Limitations on Sidechain for CV:

- **Apple Logic Pro** requires you to route through a bus channel for stereo CV applications.
- **Avid Pro Tools** does not support stereo sidechaining. Right and Stereo options will default back to the Left channel as the sidechain source.

Glide

Similarly to the Glide setting in the [Meta Knob](#) source, this slows and smooths out transitions between incoming MIDI or CV and their effect on assigned destination parameters.

Gatekeeper with Filterverse

We highly recommend using our **Gatekeeper** plug-in to generate CV signals for Filterverse.

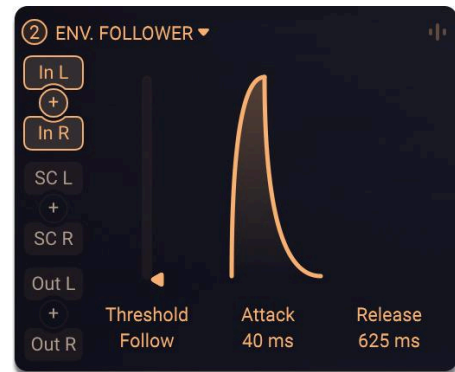
- Load Gatekeeper on a new audio track and *mute the track*.
- Right-click inside Gatekeeper’s Editor and select: “CV Output”
- Route Gatekeeper into the sidechain input of Filterverse (Please refer to your DAW’s manual for details).
- Add the MIDI/CV source to one of Filterverse’s Modulation slots.
- Under CV, select “Sidechain CV L” or “Sidechain CV R” as the input source.
- Assign the MIDI/CV source to any control in Filterverse.
- For regular audio, we recommend using the Envelope Follower mod source.



Envelope Follower

An envelope follower listens to the amplitude of an incoming audio signal and converts it into a modulation signal — often a filter cutoff in the case of envelope-following effects pedals. In Filterverse, it can modulate any destination you want.

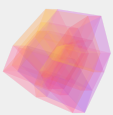
Our envelope follower has a split personality. It can either function as a full-time follower or an attack-release envelope triggered by the audio reaching a certain threshold.



Inputs

Selects the input of the envelope follower.

- **In L:** Left side of the main input.
- **In R:** Right side of the main input.
(Click + icon to use both).
- **SC L:** Left side of the sidechain input.
- **SC R:** Right side of the sidechain input
(Click + icon to use both).
- **Out L:** Left side of the main output.
- **Out R:** Right side of the main output
(Click + icon to use both).



Word to the Wiz:

A *sidechain* is a signal *other* than the current track's audio. It controls an effect *applied* to the current track's audio. In a DAW, the source of the sidechain signal could be another track, a bus, etc. A common application is to route a kick drum into the sidechain input of a compressor inserted on one or more other tracks, creating the “pumping” effect popular in electronic dance music. Consult your DAW's instructions for how to route a sidechain input into a plug-in.

Threshold

When Threshold is set to minimum, the modulation source functions as a full-time envelope follower. Set to any other value, it determines the threshold that the incoming signal has to pass in order to trigger the attack-release Envelope.

Attack

When the Threshold is set to minimum, Attack sets the amount of time it will take for the envelope signal to climb from zero to the input signal level. When the Threshold is set higher, it sets the amount of time it will take for the envelope signal to climb from zero to 100% once the Threshold is reached.

Release

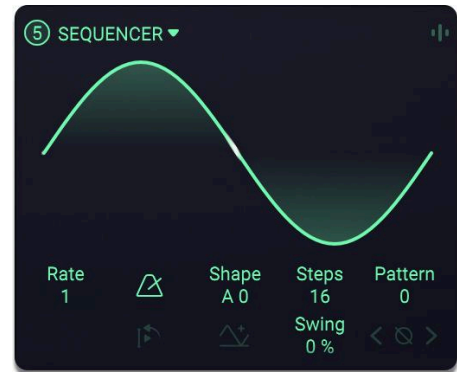
When the Threshold is set to minimum, Release sets the amount of time it will take for the envelope signal to drop to a lower level of the input signal. When the Threshold is set higher, it sets the amount of time it will take for the envelope signal to drop to zero once the incoming signal drops back below the Threshold.

Sequencer

We call it a sequencer, but this source can also work like an LFO, and either run continuously or be set to retrigger from the beginning each time a MIDI note is received.

Pattern

Loads one of 100 preset patterns (numbered from 0 to 99) into the Sequencer. Once a pattern has been loaded, you can leave its shape as-is or edit its 16 steps manually and thereby change its shape in the [Sequence Editor](#) (see below).



Switching to another pattern will discard any edits. Notice that some of the patterns correspond to simple waveforms, like a sine in the case of pattern 0. Using these patterns makes the sequencer equivalent to an LFO.

Random Pattern: Tuning the Pattern selector to the maximum (until it reads “random”) will create a new, random pattern every time that value is reached.

Shape

The Shape parameter varies the shapes of the transitions *between* individual steps of the sequencer. Therefore, you can create a wide variety of sub-patterns given a single setting of the Pattern parameter. Use this knob to gradually morph between five shaping methods:

- A.** Rounded transitions (great for creating curvy LFOs).
- B.** Linear interpolation between the steps.
- C.** Sample-and-hold / Steps.
- D.** Per-step drops (signal will decay back to zero after each step; good for grooves).
- E.** (D 100) Curved per-step drops (same as previous method but more punchy).

There are 100 degrees of variation within each method (labeled A-D), so the variation you can create here is really quite precise.

Steps

Drag on this field to set the number of steps in the sequence, from 2 to 16.

Swing

This introduces a characteristic “behind the beat” feel. Musically speaking, what’s going on is that certain rhythmic subdivisions (typically eighth- and sixteenth-notes) have alternating longer and shorter durations even though they wouldn’t be written as such in conventional music notation. Think of it as the longer durations “stealing” a little time from their shorter counterparts.

Unipolar

When The unipolar button is on, the sequencer will be scaled to operate on the positive polarity only (0 to 1). When it is off, it will be bipolar, taking both the negative and the positive polarities of the signal (-1 to 1).

Rate

Drag on this field to set the rate of the Sequencer. When Rate Quantize is enabled, Rate will be set in multiples or divisions of the beat, including dotted and triplet feels (suffixed with d or t). When disabled, Rate will be set in Hertz.

Tempo

Click the metronome icon to quantize the Rate to tempo divisions of your project’s tempo.

Retrigger

This decides whether the sequencer runs continuously or if it resets to the beginning of the waveform/pattern every time a MIDI note is received. If Retrigger is off, the Sequencer will be synchronized to the song position. If Retrigger is active, the LFO will sync to the beginning of each incoming MIDI note.

Rate Modulation

When retrigger is engaged, you will see the familiar pop-up sliders button appear beneath the Rate control. This means the Sequencer/LFO Rate can now be modulated by any of the active sources, including the Sequencer itself. Click on the icon to open the [pop-up sliders](#) for the Rate.

Sequencer Phase

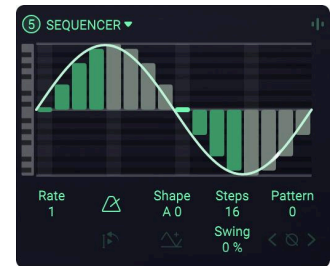
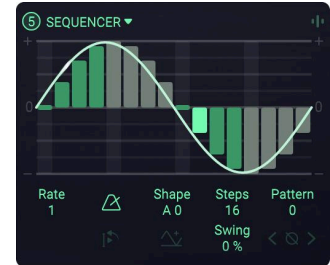
These controls can reverse and shift the phase (the point at which the pattern begins in time) of the Sequencer. Click the center button to flip the phase 180 degrees. Use the arrows to nudge the phase forward or backward by one sequencer step at a time. Click the right arrow once, for example, and step 1 becomes step 2, step 2 becomes step 3, and the final step becomes the new step 1.

An LFO-like shape may appear to be shifting smoothly. What's really going on is that the Shape setting is causing the *transitions* between steps to be smooth; clicking the phase forward or backward always shifts the shape by an entire step at a time. Try a choppy pattern set to, say, eight steps, and you'll get some clear visual feedback about how this works.

Sequence Editor

Filterverse lets you either free-draw or carefully design your own sequences and LFOs. Hover with the mouse over the pattern/waveform display to show the editor. To edit the steps, use these techniques:

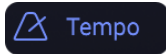
- **Click-drag** in the editor to draw a new **freehand** pattern.
- **Right-click individual steps** to have **fine** control over their value.
- **Shift-drag** to **quantize** the values of steps to quarters (shown in upper image).
- **Double-click** a step to zero it out.
- **Command-drag** (Mac) or **Control-drag** (PC) to **zero** several steps at once.
- **Left or Right keys** shift the **phase** of the steps.
- **Up or Down keys** invert the **polarity** of the steps.
- Use the '**A, B, C, D, E**' keys on your computer keyboard to set the shape.
- **Alt-drag** (Option-drag on Mac) to quantize the values of steps to 12ths. This editing mode can facilitate a melodic sequence of **notes** (shown in lower image). To do this, also set the modulation amount for the Cutoff to +/-12 semitones.



Random Generator

As its name implies, the Random pattern generator is capable of randomizing tempo divisions and note quantization, then converting the resulting output to a modulation source.

Rate



Drag on this field to set the rate of the Random generator. When Tempo is enabled, Rate will be set in multiples or divisions of the beat, including dotted and triplet feels (suffixed with d or t). When Tempo is disabled, Rate will be set in Hertz. Rate can be a modulation destination.



Rate Modulation

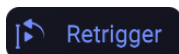


When retrigger is engaged, you will see the familiar pop-up sliders button appear beneath the Rate control. This means the Sequencer/LFO Rate can now be modulated by any of the active sources, including the Sequencer itself. Click on the icon to open the [pop-up sliders](#) for the Rate.

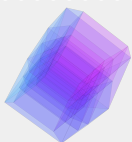
Tempo

Click the Metronome icon to quantize the Random Generator's Rate to tempo divisions of your project's BPM.

Retrigger



When active, this generates a new random value every time a MIDI note is received and resets the Random Generator's clock. If Retrigger is off, the Generator will be synced to the song position. If Retrigger is active, the LFO will sync to each incoming MIDI note.



Word to the Wiz:

Dragging the graph left and right will adjust rate while dragging it up and down will adjust the smooth value

Unipolar

When The unipolar button is on, the Random Generator will be scaled to operate on the positive polarity only (0 to 1). When it is off, it will be bipolar - taking both the negative and the positive polarities of the signal (-1 to 1).

Glide

This determines the amount of smoothing applied to the modulation signal. Higher values result in slower transition times when a value in the random pattern changes.

Range

The range in octaves of the random signal. This control only has a significant effect when a modulation amount slider for a destination parameter is set at 100%.

Rhythm

This parameter sets the probability that a beat will be divided and the resolution at which that will happen. The higher the parameter's value, the more frequent and finer-grained the divisions will become.

Division

This selects a set of tempo divisions according to which the Random Generator performs its work.

Value	Result	Divisions
Sub	Divisions of 2	1, 2, 4, 8, 16, 32, 64, 128
Tuplet	Divisions of 3	1, 3, 6, 9, 12, 18, 24, 36
Sub+Tuplet	Divisions of 2 and 3	1, 2, 3, 4, 6, 8, 12, 16
Complex	Complex divisions of 5 and 7	1, 2, 3, 4, 5, 6, 7, 8
Prime	Divisions corresponding to prime numbers	1, 2, 3, 5, 7, 11, 13, 17

Scale Quantize

This parameter will quantize the random modulation signal to musical notes. Turning this knob will select different [scales](#) for quantization. This is meant to be used with the Random Generator applied to the Classic Filter's frequency knob, with the knob's modulation depth slider set at 100%. (It won't work correctly and really doesn't make sense with other destinations.)

Oscillator

This is not an LFO, but an audio-rate oscillator that can produce subtle, dramatic, and even rude effects when applied to the Cutoff, Resonance, or even Type settings of any of the filters. The amplitude, pitch, and shape of the oscillator waveform all contribute to the character of the modulation.



Coarse

The Coarse control sets the pitch of the Oscillator in semitones. Note the pop-up sliders button under the control; you can modulate the coarse pitch

Fine

This fine-tunes the pitch of the Oscillator in cents, with a range of +/- 50 cents.

Shape

Continuously crossfades the Oscillator waveform from sine (0%) through triangle, saw, square, and a thin pulse wave (100%).

Glide

Introduces a lag time between changes in the Oscillator amplitude and corresponding changes in the destination parameter(s) it's modulating.

MIDI

When the MIDI icon is inactive, the absolute pitch of the Oscillator is solely determined by the Coarse and Fine controls. When the MIDI icon is active, the base Oscillator pitch is determined by incoming MIDI notes, with any additional transposition or pitch-shifting imposed by the Coarse and Fine controls. To learn more about connecting MIDI to the plug-in please refer to the [MIDI control](#) section of the manual.

Pitch Detect

The Pitch Detect modulation source turns the pitch of an incoming audio signal into modulation information. You can load one instance of Pitch Detect in a Preset; if you load multiple instances, their settings will mirror the first one you loaded.

With a positive depth set for a given destination in the [pop-up sliders](#), a rise or fall in pitch translates into a rise or fall in the destination's value. With a negative depth set, that relationship is reversed.

To have the Cutoff-frequency track the pitch detected, tune the Cutoff-modulation amount to +12 semitones.



Pitch Meter

The needle will only activate when a note is detected, and shows how closely tuned the note is (-+50 cents). The values just below the needle display the pitch of the incoming signal in two ways: musical note and Hertz.

The dot at the bottom of the meter shows the level of the signal fed into the pitch detector.

Input Select

The icons on the left select the input source whose pitch will be detected.

- **In L:** Left channel of the track on which Filterverse is inserted.
- **In R:** Right channel of the track on which Filterverse is inserted.
- **SC L:** Left sidechain input the track on which Filterverse is inserted.
- **SC R:** Right sidechain input the track on which Filterverse is inserted.

To use both sides of either the main or sidechain input, click the + sign between them.

Detect

This selects the range of the pitch detection, and has three values:

- **Low:** Use for very low-pitched (bass) audio sources.
- **Mid:** Works well for most melodic material.
- **High:** Use for high-pitched (soprano) audio sources.

A good general strategy is to start with medium, then try either low or high (depending on the character of the source) if you're not hearing detection as precise as you would like.

Glide

Similarly to other modulation sources we've covered so far, Glide smooths out the audible transitions between a given pitch being detected and its effect on the modulation output value. At higher settings, the modulation output might be almost static. This is because if your source track has a lot of pitch changes, these can "outrun" pitch detection before it has time to catch up.

Appendix 3: List of Scales

This chart lists the musical scales available to both the [Meta Knob](#)'s quantize function and the [Random Generator](#).

Value	Description	Notes (assuming C root)
Off	No quantization applied	None
Chromatic	12-tone chromatic scale	C, C#, D, D#, E, F, F#, G, G#, A, A#, B
Octatonic a	8-tone symmetrical scale starting with a minor second	C, C#, D#, E, F#, G, A, A#
Octatonic b	An 8-tone symmetric scale starting with a major second	C, D, D#, F, F#, G#, A, B
Major	Major diatonic scale	C, D, E, F, G, A, B
Minor	Minor diatonic scale	C, D, Eb, F, G, Ab, Bb
Melodic	A melodic minor scale	C, D, Eb, F, G, A, B
Maj pent	Major pentatonic	C, D, E, G, A
Raag pent	Raag Vrindavani Sarang	C, D, F, G, B
Sus pent	Suspended pentatonic	C, D, F, G, Bb
Man pent	Blues minor based on Chinese scale	C, Eb, F, Ab, Bb
Yo pent	Yo Japanese scale	C, D, F, G, A
Min pent	Minor pentatonic	C, Eb, F, G, Bb
In pent	In Japanese scale	C, Db, F, G, Ab
Kokin pent	Kokin-Joshi Japanese scale	C, Db, F, G, Bb
Han pent	Han-Kumoi Japanese scale	C, D, F, G, Ab
Whole tone	Each interval is a whole-step only	C, D, E, F#, G#, A#
Diminished	Notes of diminished triad	C, Eb, Gb, A
Augmented	Notes of augmented triad	C, E, G#
Maj triad	Notes of major triad	C, E, G
Min triad	Notes of minor triad	C, Eb, G
Sus2	Notes of suspended-2 triad	C, D, G
Sus4	Notes of suspended-4 triad	C, F, G
Fifths	Perfect fifth interval	C, G
Octaves	Octave intervals only	C3, C4, etc.

Appendix 4: Mid-Side Processing

Mid-Side processing is a technique that involves **two microphone signals**: the **'Mid'**, which captures audio from the front (usually with a cardioid microphone), and the **'Side'**, which captures audio from the sides (typically with a figure-eight microphone). In mixing, the Mid represents the center of the stereo image, while the Side contributes to the stereo width.

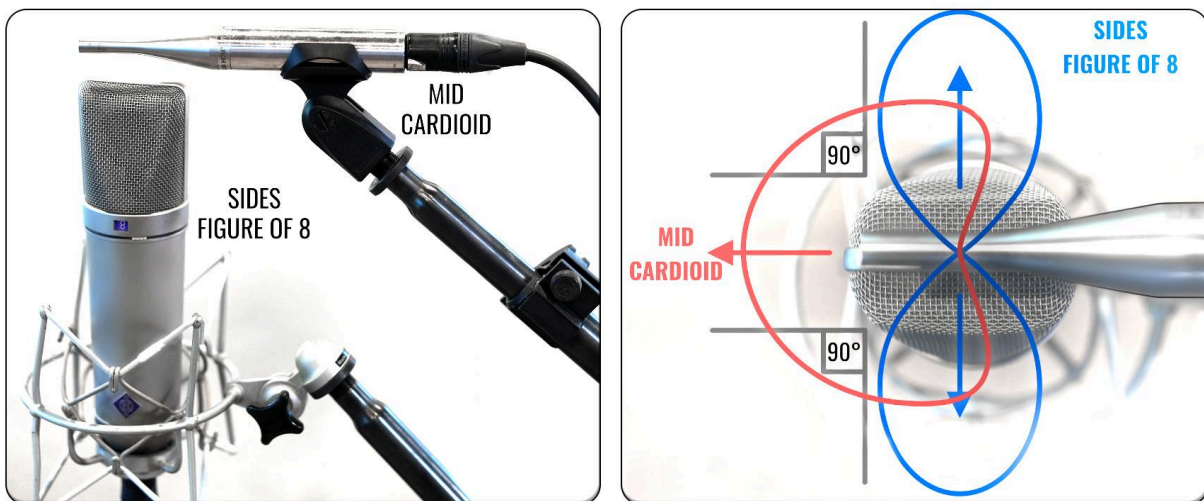


The **'Mid'** and **'Side'** channels are then decoded to Stereo - **'Left'** and **'Right'** using the following formula:

Left = Mid + Sides

Right = Mid - Sides

MS processing is versatile and can enhance various recording scenarios. For instance, it can add depth to drum overheads by emphasizing room ambience or provide a wider stereo image for acoustic guitar tracks. Understanding the context and desired outcome is key to effective MS processing.



Unlike M/S microphone recording, M/S plug-in processing typically starts with a standard stereo track, which is first encoded into Mid and Side components:

Mid = (Left + Right) / 2.0

Sides = (Left - Right) / 2.0

After encoding, processing is applied separately to the Mid and Side channels. Finally, the processed Mid and Side components are decoded back into a stereo signal.

Common pitfalls and misconceptions

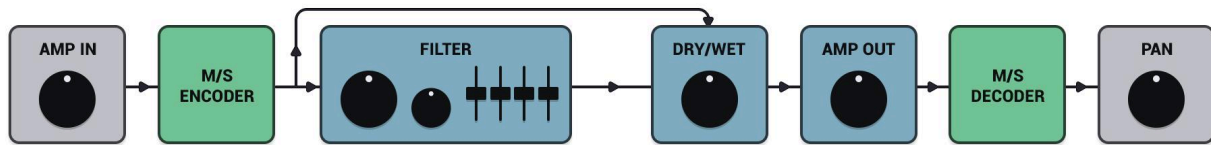
- **There is no distinct “center channel” that can be completely isolated:** “Mid” in Mid-Side does not equate to isolating audio solely in the center, nor does “Sides” represent separating everything that’s not in the center. The term Mid-Side, derived from a microphone technique, is about the relationship between the center image and stereo width. It is not possible to fully separate the center image from the sides
- **Phase relationship:** The efficacy of MS processing hinges on the **90-degree** phase relationship between the channels. This relationship is **crucial** for a natural and coherent stereo image. Failing to recognize the critical 90-degree phase relationship between Mid and Side can lead to phase issues and a compromised stereo image.
- **Incorrect source material:** Using MS processing on stereo tracks where the left and right channels are unrelated can result in unnatural or distorted stereo image. MS processing is not a tool to create a stereo image from two unrelated mono tracks. Similarly, applying it to a mono file is ineffective because there’s no “sides” information to manipulate.
- **Overuse of sides information:** Excessively boosting the side signal can make the mix sound unbalanced and can cause phase compatibility issues, especially in mono playback. removing the “Mid” component will not leave the left information on the left and the right information on the right. Rather it will result in an out of phase mix of the left and right on both sides.
- **Neglecting mono compatibility:** Failing to check for mono compatibility can lead to elements of the mix disappearing or significantly changing when summed to mono.



Understanding these pitfalls helps in effectively leveraging MS processing in Filterverse for a balanced and phase-coherent stereo mix.

Mid/Side in Filterverse

Each of the three filters in Filterverse has an MS button. When activated, this button enables MS processing for that filter. This means the filter's operations, including its effects and controls, are applied in the MS domain.



Before the signal reaches the filter, it is **encoded from stereo to MS**.

Once in MS mode, the stereo modulation controls, previously labeled for left and right, are relabeled to Mid and Sides. This allows for targeted processing of the mid content and the sides separately. Only the Input level modulation and panning still remain in the stereo domain (L/R) for an additional layer of flexibility.

After processing through the filter stages, the signal is decoded back to stereo.

Pan

Unlike some plug-ins where the pan control is repurposed for MS balance, the Pan knob for Filterverse maintains its functionality even when M/S mode is engaged.

M/S balance

The MS balance can be tuned using the “Filter Out” stereo modulation controls. This design choice reduces the likelihood of accidental misuse and allows for a more flexible control over the stereo image.

Dry/Wet

If you want the filter effect to apply solely to the Mid (center) content, adjust the Dry/Wet parameter to be fully wet on the left side. Conversely, if your focus is on processing only the Side components of the stereo field, set it to be fully wet on the right side.

Self-Resonance

Some filters in Filterverse are capable of self-resonance, meaning they can self-oscillate and generate a tone at high resonance settings. While this feature can be creatively useful, it requires careful handling in MS processing, particularly with mono signals.

When a mono signal is processed in MS mode, and one of the filters is set to self-resonate, it can lead to unpredictable results. Since the Mid and Side components are processed separately, the self-resonating filter might continue to oscillate even without receiving audio input. This self-oscillation can cause the two filters (Mid and Side) to enter different phase relationships. For example, if one filter (e.g., the Sides) continues to resonate without audio input, while the other (the Mid) resonates but also receives audio, this can create a phase imbalance.



