FILTERVERSE

USER MANUAL

V1.2.0

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By

POLYVERSE

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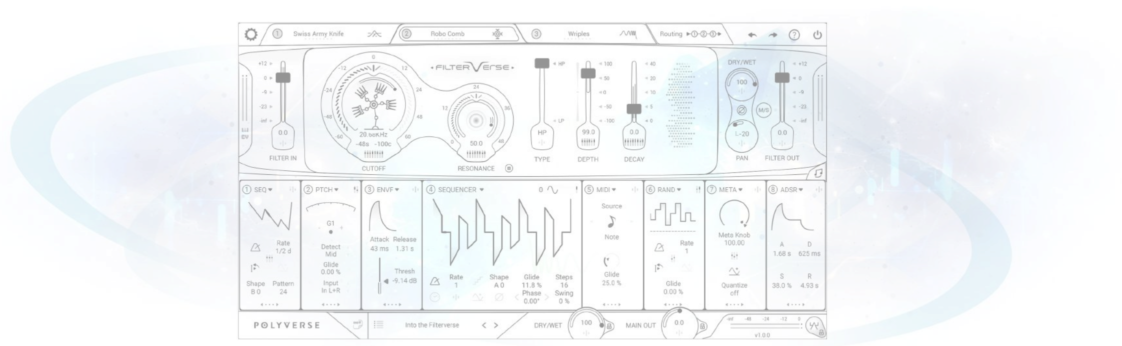


Introduction

User manual written by:
Stephen Fortner

Factory Presets by:
Assaf Dar Sagol, Erez Eisen, Navi Retlav, Pandocrator, and Databroth.

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

and also to:
Eyal Amir for your invaluable help, coding advice and friendship.
Michał Garstecki for your incredible Synthesis and DSP advice, help and friendship.
Professor Julius O. Smith for helping us figure out the final details in *fractazer*.
Geraint Luff for helping us realize our vision for the *barber pole* filter.

About Filterverse

An endless universe of filtering possibilities

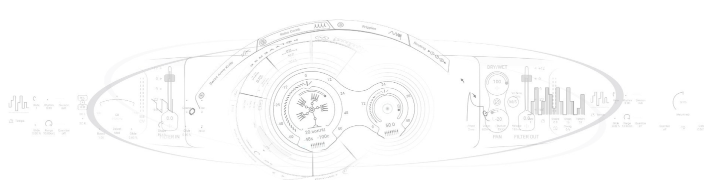
Filterverse is a new type of multi-filter plug-in that elevates the concept of the filter to a musical instrument in its own right. From traditional lowpass, bandpass, and highpass filters to experimental resonating bodies, comb designs, decimators, and much more, it’s a one-stop-shop for all your filtering 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.

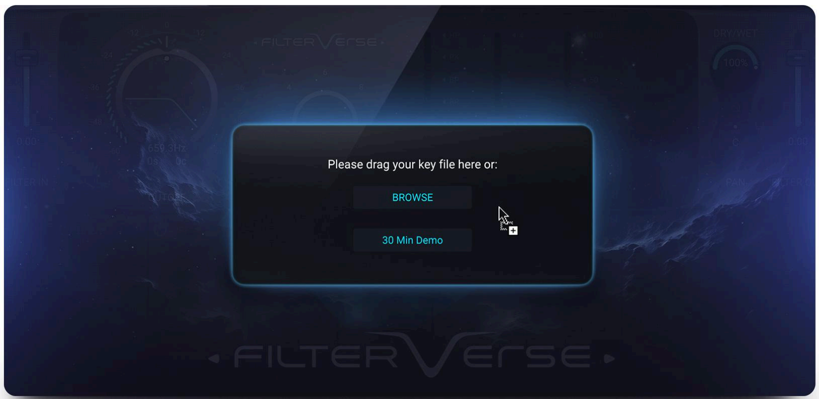
We know that the selection of plug-ins available to music creators and DAW users can be confusingly large. We believe the primary focus of any musical tool should always be creativity. That’s why we intend Filterverse to provide a completely new palette of sounds and technologies formerly accessible 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

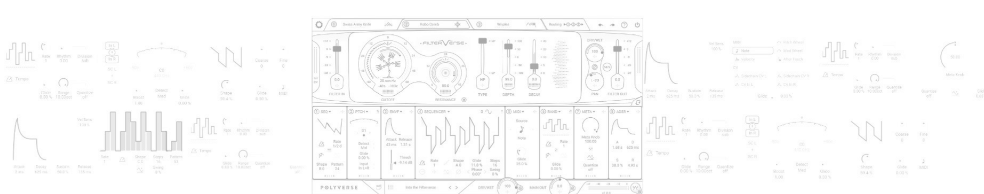


To install Filterverse, simply run the installer and follow the instructions on your screen. For free tutorials and videos about setting up and using Filterverse with your favorite DAW (Digital Audio Workstation), please visit our website at http://polyversemusic.com/support

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

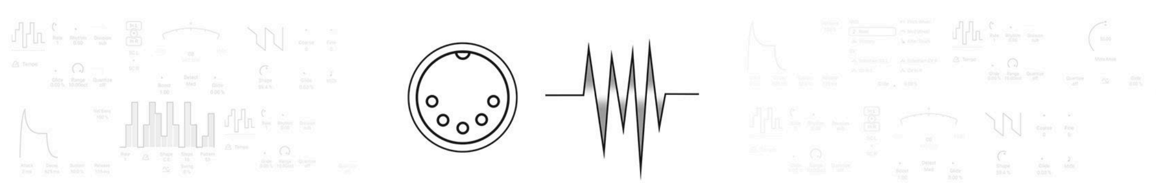


MIDI and CV control

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, this next section offers a quick overview for popular DAWs. Consult your own DAW's documentation for further details.

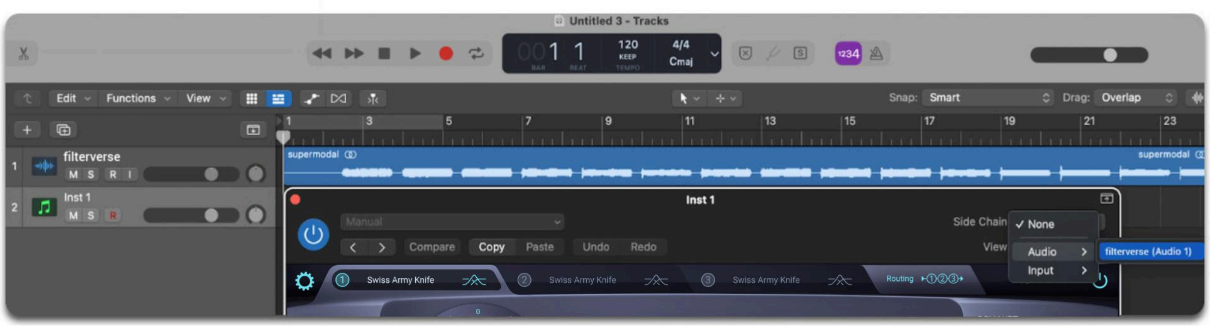


Ableton Live

- 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.

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.

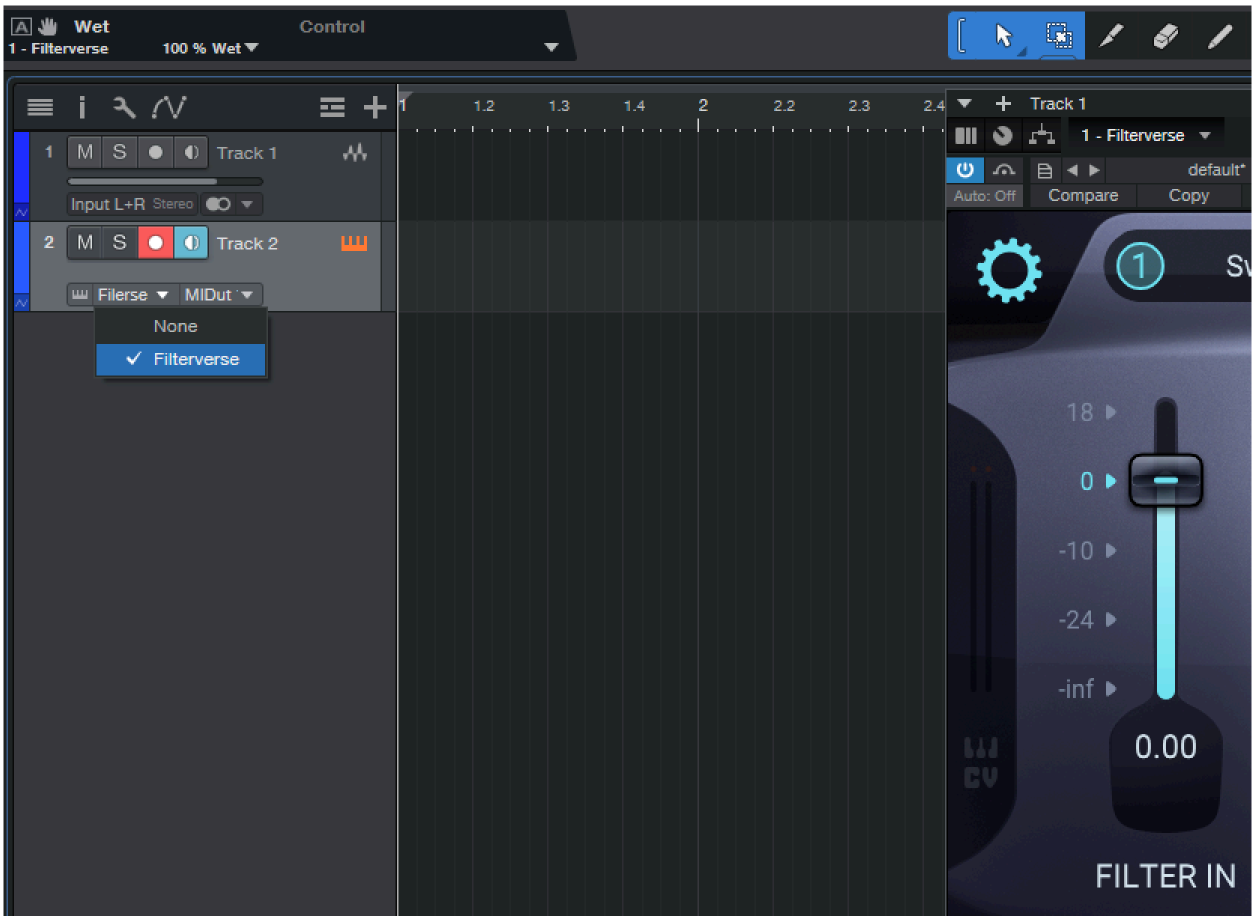
- Create an audio track.
- Insert Filterverse on this audio track.
- Create an instrument track.
- Click on the output menu of the instrument track.
- In this menu, navigate to under “NOTES TO TRACKS.” Select “Tracks” then the name of the *audio* track on which Filterverse resides.

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.

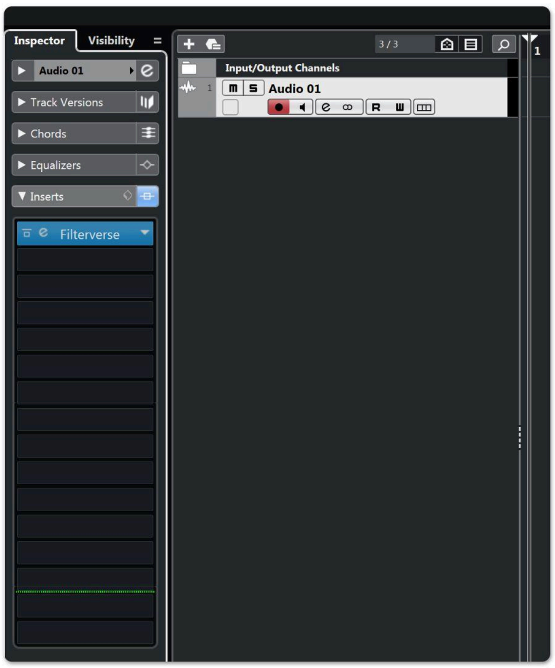
PreSonus Studio One

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



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 pop-up 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

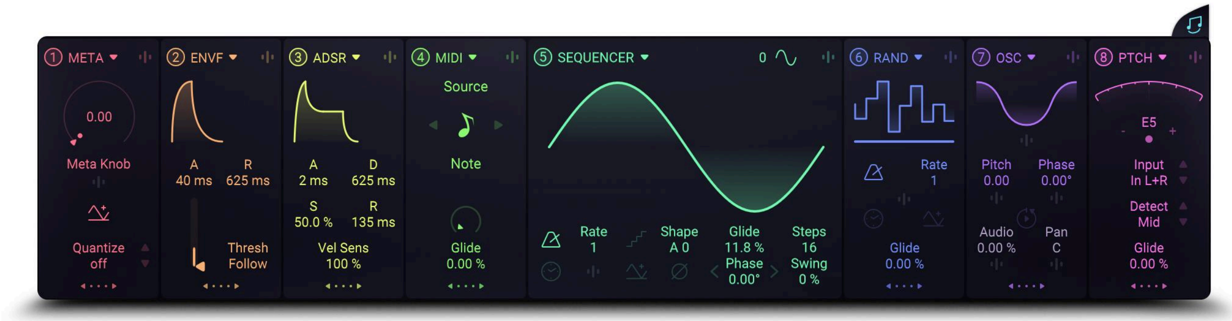
The main architecture of Filterverse consists of three filter slots, accessible via tabs across the top of the plug-in window. For each of these, you can choose from a roster of filter models; the controls for each filter model appear directly below the active tab.

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.

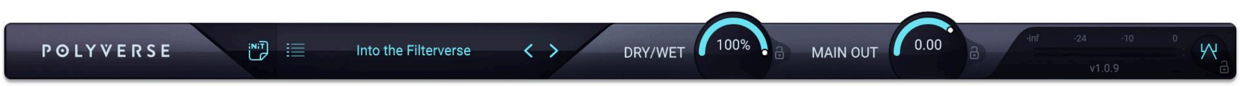
Click the question mark icon in the top right corner of the interface to open the manual in your browser for quick reference (requires an internet connection).

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



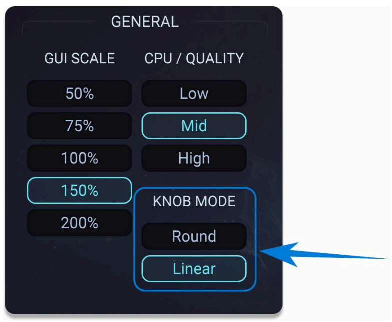
The star of the show here is the preset browser. Beyond simply stepping through presets, the browser allows you to search for presets by text (e.g. part of a name) or tagged attributes such as “dramatic,” “subtle,” and so on. The master dry/wet mix, main output level, and clipping style controls are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

2. 

3. 

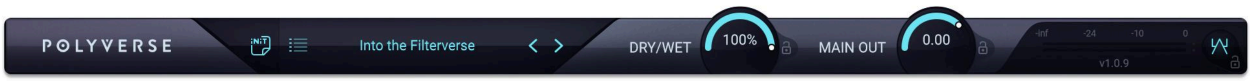
1. First, make sure Filterverse is set up to receive MIDI [in your DAW](#).
2. Right-click on any parameter to display its MIDI learn pop-up.
3. Now, click the Learn button to switch the display to activate recording mode.
4. Move the control you want to use on your MIDI device, and the display should 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:
Different DAWs 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

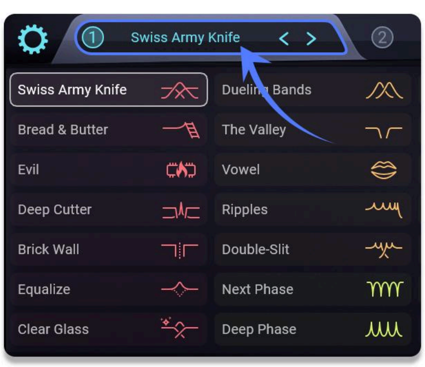
Filter tabs

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.

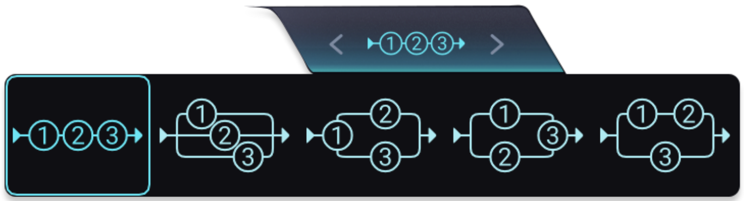
Selecting a filter



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



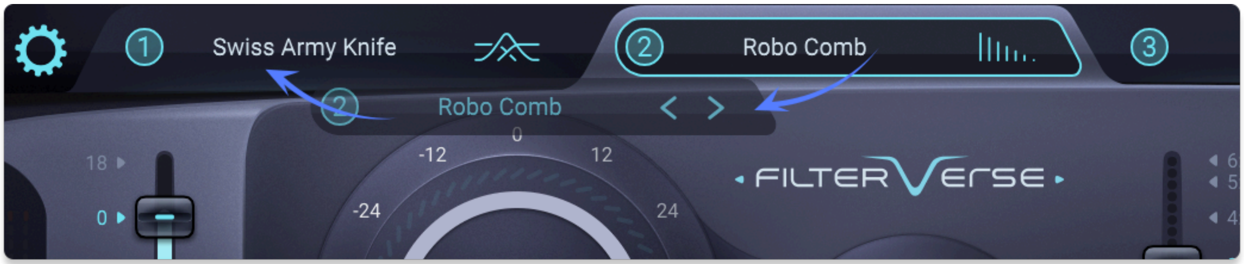
Click the filter routing tab to bring up the above options. From left to right, these are:

- 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.

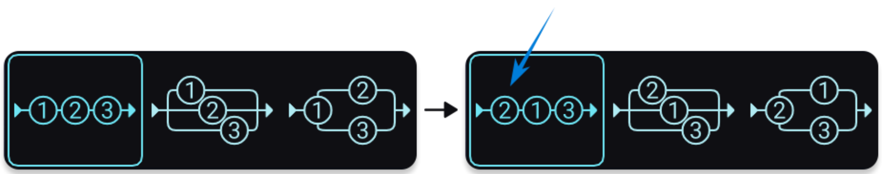
You can also click the left and right arrows in the routing tab to step through the routing schemes serially, without opening the drop-down menu below.

Why are we saying things like “filter in the first tab” and not simply “Filter 1”? There’s a good reason, which we explain directly below.

Filter position in routing



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.

You may also copy a filter's setting to another by dragging while pressing the alt key. This will duplicate all of the filter's parameters, including the assigned modulations.

Filter Level and Pan sections

Filter In

Adjusts the input volume of the filter and offers up to an 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 Filterverse. At the minimum (dry), no processing 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

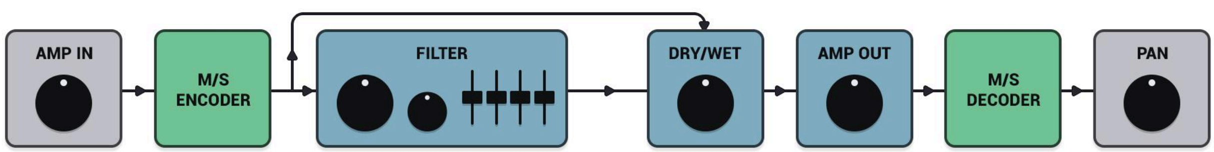
The pan control allows the audio signal to be positioned between the left and right channels. The pan control ranges from -100 percent to +100 percent, where -100 percent is full left and +100 percent is full right. At the center position, both channels are balanced. The modulation of the panning uses 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 the musical note *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!

Right click the coarse text to see note names instead of semitones in the display.

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 nasal, 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.

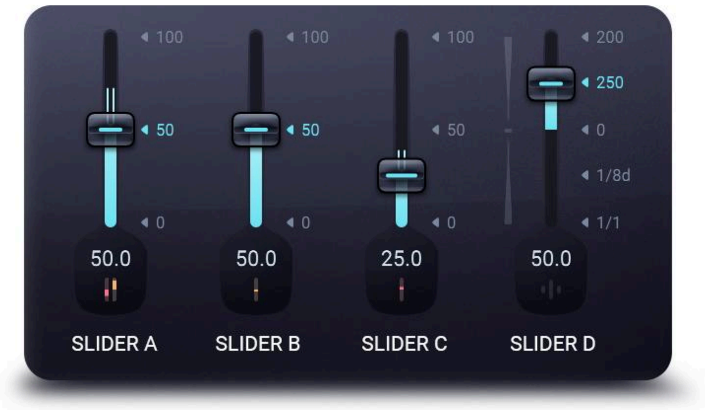
Beyond Cutoff and Resonance

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 similarly 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 dampening 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.

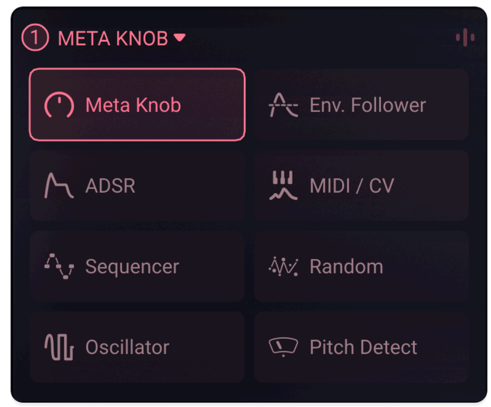
You can mix and match sources in the slots any way you like, including having multiple slots with the same type of source. Each slot has a color locked to its left-toright order onscreen; slot 1 is always red, for example. So, you might have a "red sequencer," a "green sequencer," etc., and be able to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

Copy/Swap

Drag the modulation title to swap its place with another slot. Alternatively, drag while pressing the alt key to copy all of the settings between slots.

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

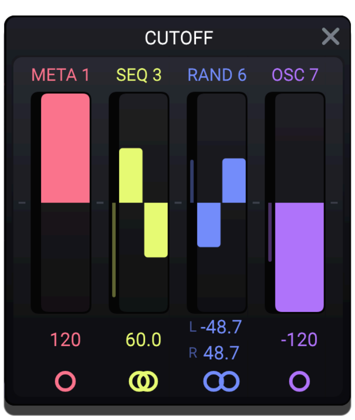
To modulate a parameter, first load a modulation source into one of the slots by clicking on its title and selecting from the choice of [available modulators.]

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

Assigning a destination

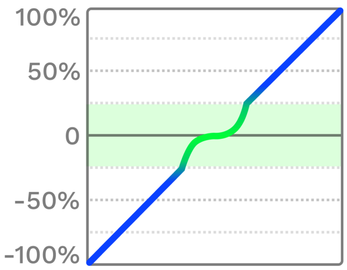
Click the Modulation Pop-up Button for the parameter you want to modulate, to open the Modulation Pop-up Sliders. This button also has a monitor allowing you to see the amount of modulation on the parameter at a quick glance.

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.

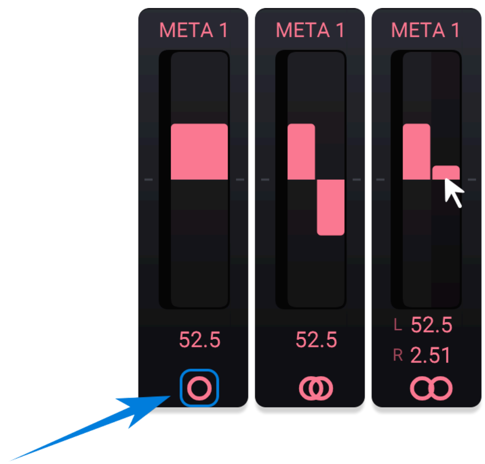
The controls are bipolar, so sliding them up to the maximum will result in positive modulation, while sliding them down to the minimum will result in negative modulation. All controls will display modulation sliders for all active sources, all the time. To ensure a given source is not modulating a parameter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

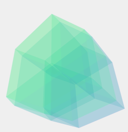


Most modulation amount sliders have a **curve** — which enables more resolution in values that are closer between zero and the 25 percent mark, and then linear action afterwards.

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.

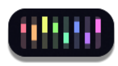
Modulation at a glance

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

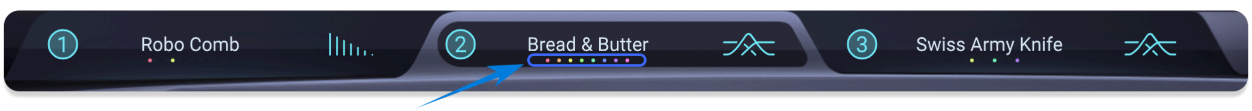
Once a slot is on, **Modulation pop-up buttons** will appear underneath any main control that's eligible to be a modulation source. These appear as small vertical lines that match the color of the modulation sources. Note that you will see only as many colored vertical lines as you have modulation sources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

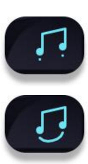
When a slot is modulated, its level will change according to the behavior of the source(s) modulating it. For example, when an envelope is being modulated by MIDI velocity, it will act like a classic velocity sensitivity slider, where more modulation means more velocity range.

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.

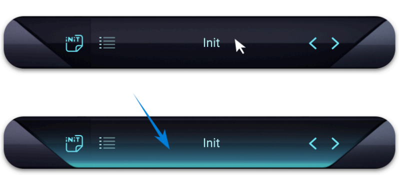
Legato 

Some modulators in Filterverse (such as the ADSR, Sequencer, Random, MIDI/CV and Oscillator) receive gate signals. The legato feature selects between two states:



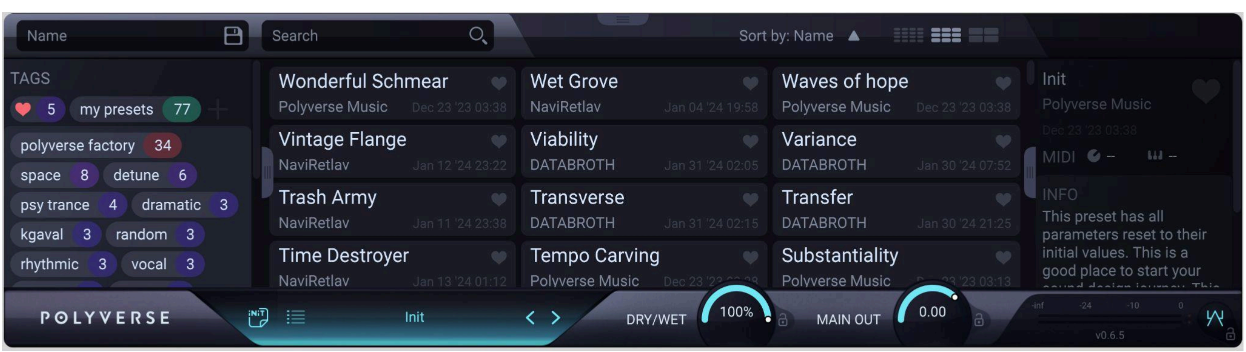
- Staccato: Every new note sends a gate signal.
- Legato: Notes played before the previous note(s) are fully released do not send a new gate signal.

Preset Browser



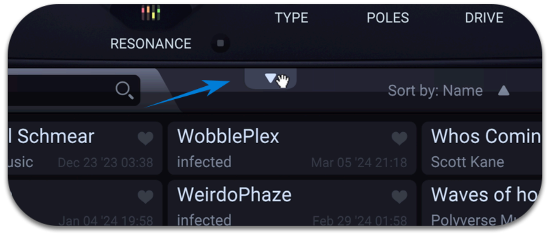
To step through Presets, click the Left/Right arrows located on either side of the Presets Bar. Clicking directly on the Preset Bar will expand the bottom half of the Filterverse interface into a full-featured Preset browser:

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

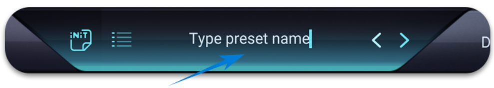


Once the button is clicked, it is possible to **browse the Presets using the arrow keys** on the keyboard.

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.

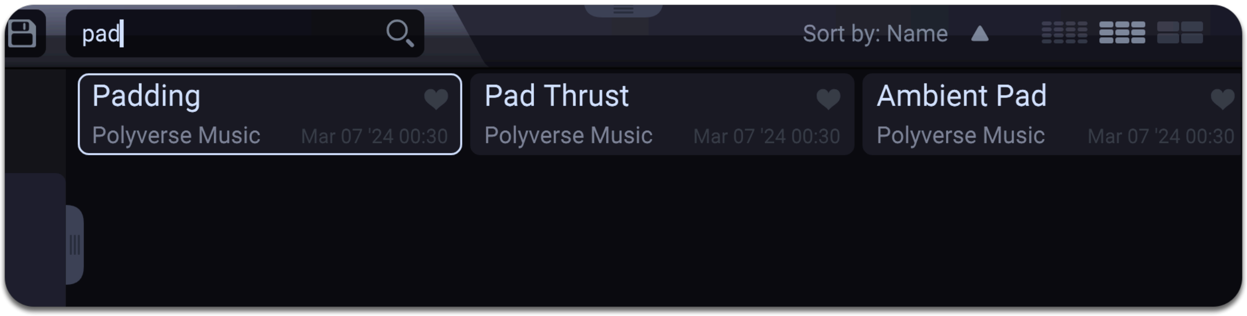


Word to the Wiz:
Saved presets can be found in the following locations:
MacOS:
Users/[user-name]/Library/PolyVerse/Filterverse/Presets
Windows:
C:\Users\%USERNAME%\AppData\Roaming\Polyverse\Filterverse\Presets

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.

Searching Presets

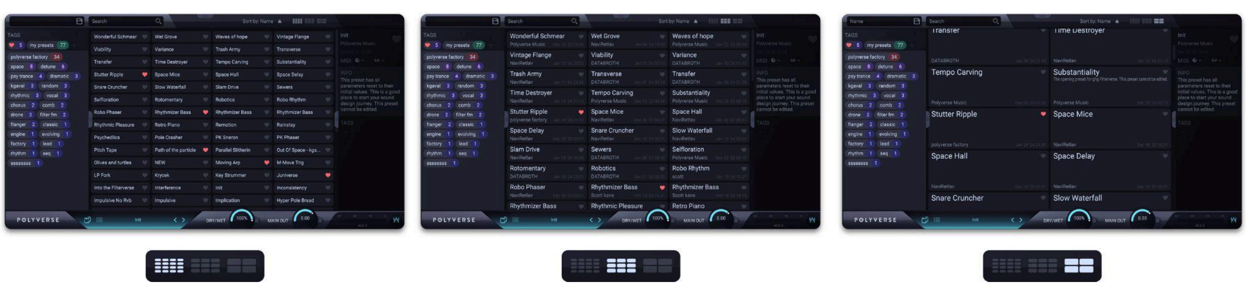


There are two text fields at the top of the browser: one for saving Presets and one for searching. To search for a Preset by text (in its name or author's name), type the text into the field on the right. The results will appear in the center of the browser directly below, updating in real time as you 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.

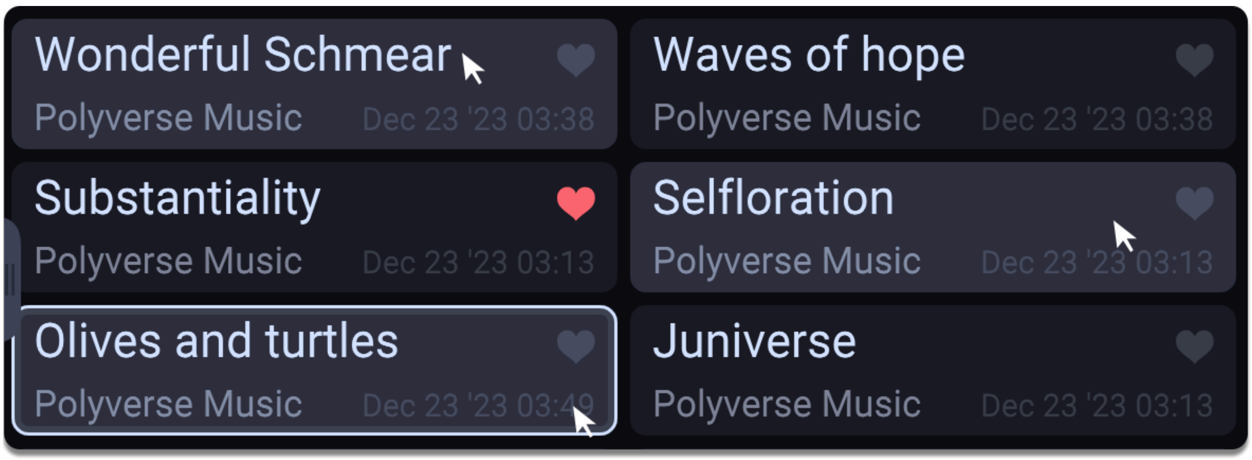


To select and load a Preset simply click on it in the results area.

Favoriting Presets

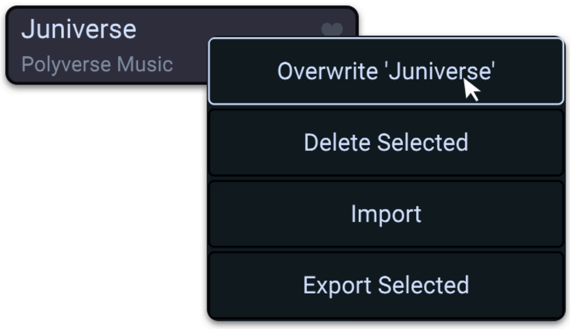
To “like” or make a Preset a favorite, click the heart icon next to its name in the results area or the [Preset Info](#) section.
If any Presets are favorited, the heart will show up as a tag in the [Tags](#) sidebar, allowing you to display only favorite Presets with a click.

Selecting multiple 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.) to select all presets, press Control+a (Windows) or Command+a (macOS).

Preset Operations Menu



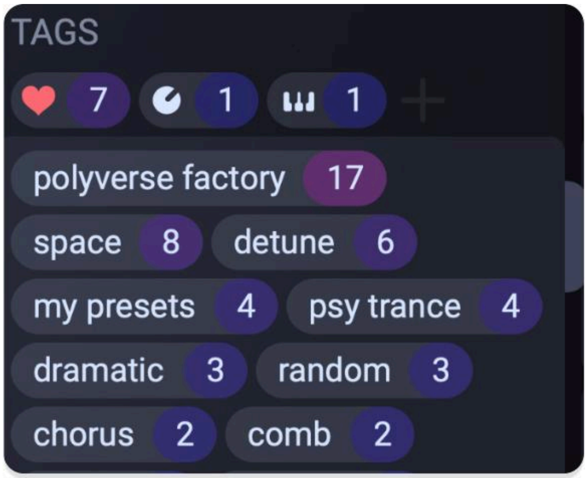
Right-click on any Preset in the central results area to display a pop-up menu offering four options for Preset management:

- **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.

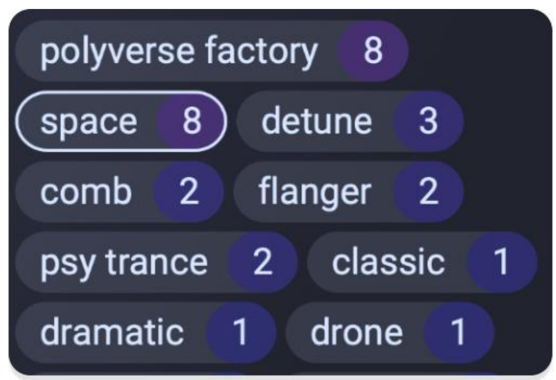
Notice the number on the right side of any tag. This refers to the number of Presets that have that Tag.

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.

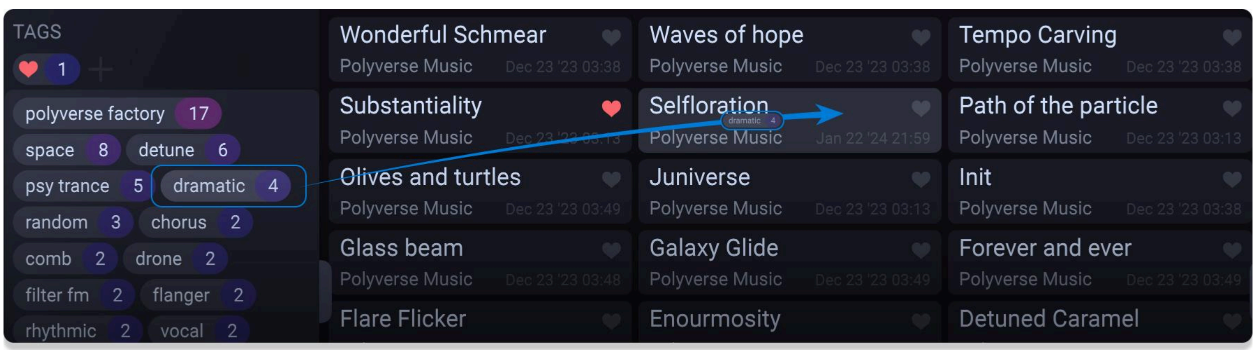
Excluding Tags

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

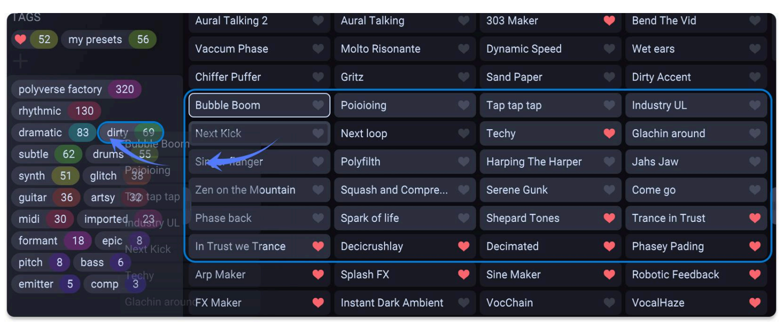
- Tags you've actively selected are outlined in white.
- Tags that remain as a result of your selections are not outlined.
- Tags you've excluded are outlined in red.
- Tags are displayed, left to right and then top to bottom, in descending order of the number of Presets that have a given tag. When two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

Adding a Tag to a Preset

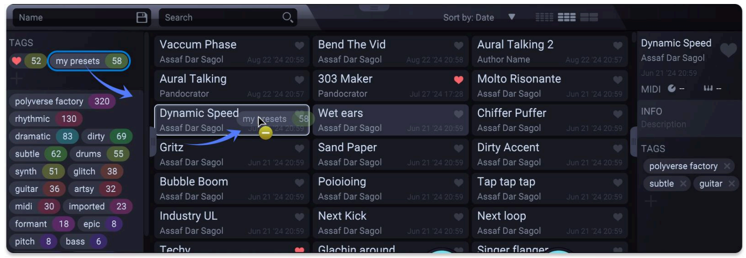


SImply drag and drop any Tag from the Tags sidebar onto a Preset or a selection of multiple presets in the central results area.

You can also drag presets onto the tag itself — either single presets or multiple ones using your OS-level multiple selection routine. There is yet another way to add tags in the Preset Info area.

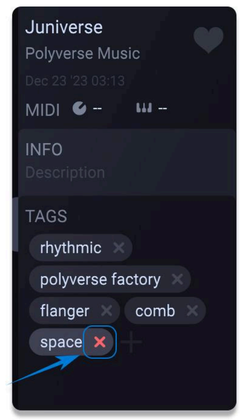


Removing a Tag from Presets



Drag presets (single or multiple) or a tag while pressing “alt” to remove a tag from those presets.

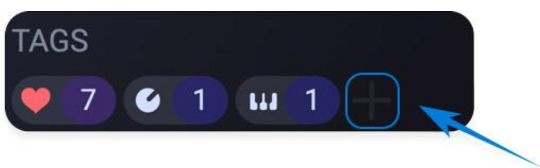
Removing a tag can also be done in the [Preset Info] sidebar on the right side of the Preset browser.

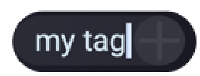


Hover over the X next to the tag name and it will turn red. Click the X to remove the Tag from the currently selected Preset.

You can also hold the Alt (Windows) or Option (macOS) key while dragging a tag to a preset, or a preset to a tag, to remove the tag.

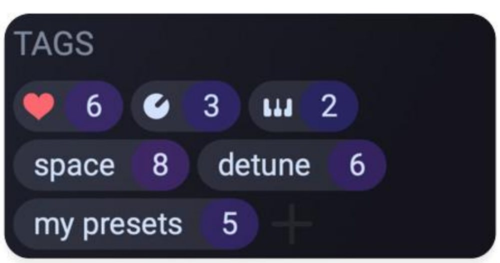
Creating a Tag





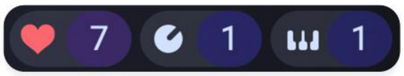
You can create your own tags. Click the **+** sign in the header of the Tags sidebar and it will expand into a field where you can type text. Name your tag and press Enter/Return.

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.

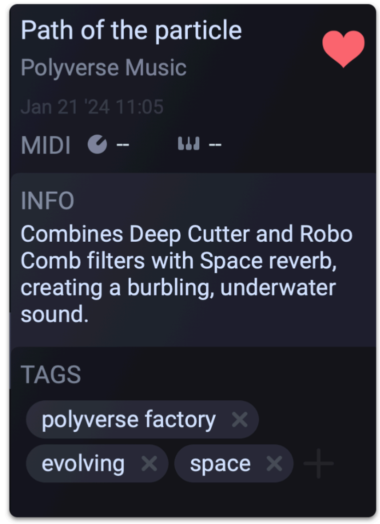
- **Import:** Imports a Preset file (or ZIP file containing Presets) and applies the selected tag to all Presets imported.
- **Export:** Exports all Presets bearing the selected Tag as a ZIP file.

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

Freshly imported presets will be selected within the preset browser, and loaded in the plug-in.

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

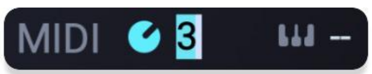
To the right of the Preset name and author is a heart icon for favoriting that Preset. This is a mirror of those in the Preset list in the search results area.

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.

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.

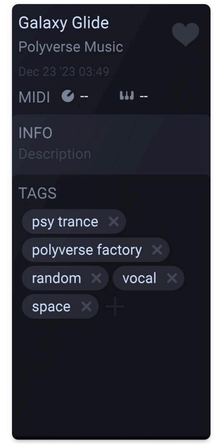
Remember, the number you’re typing in is *not* a CC — it’s the trigger **value** for a program change. Now, whenever Filterverse receives that value from your controller, it will load the Preset.

Via MIDI 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.

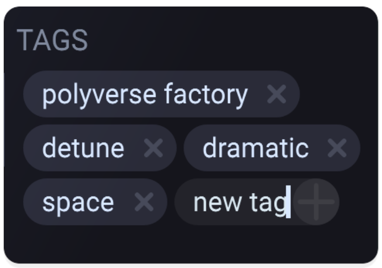
Info



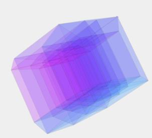
Here, you can enter descriptive notes about the current Preset. Just type in the desired text and hit enter/return to save.

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.

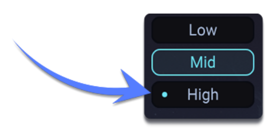
Settings

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

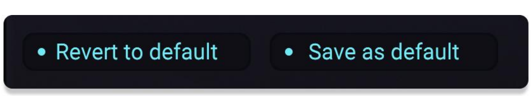
These are global settings, meaning they affect all Presets equally and do not change at the Preset level.

Settings can be saved within a DAW project or as the default. When settings are saved as default, the next time an instance of Filterverse is loaded, it will load with the default settings.

If you change a setting, a blue dot will appear next to the *current* default value, and two buttons appear below:

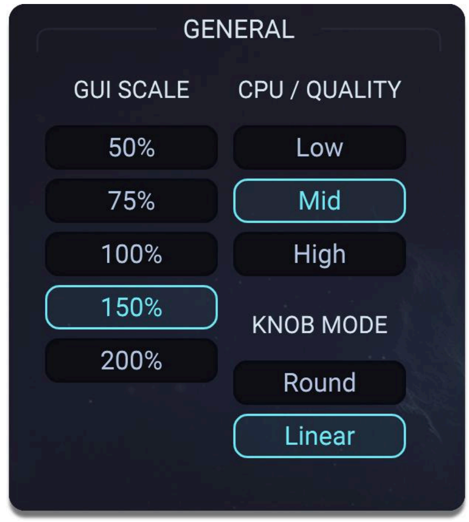


- Revert to default: Overwrites current settings with default settings.
- Save as default: Saves the current settings as the new default settings.



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 percent to 200 percent 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

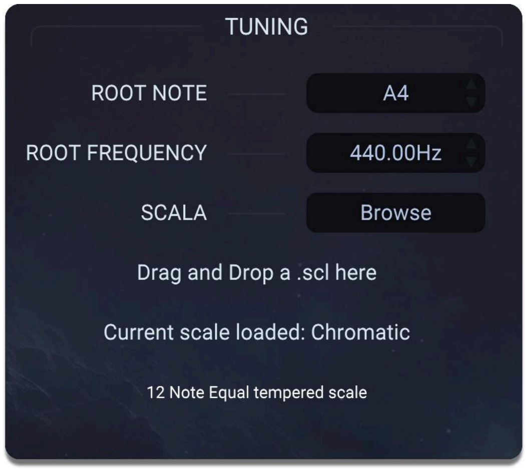
This governs the response of the main Cutoff knob as well as any other knob-shaped control in Filterverse’s interface (such as those in the modulation sources.

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 tuning

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.

Scala files can determine any or all of these three things:

- *Tuning*: The frequency in Hertz of a root note, e.g. *A*=415Hz for Baroque music, *A*=432Hz for New Age music, or middle *C*=256Hz for scientific work (used because the cycles per second corresponding to C in any octave are always a whole number).
- *Temperament*: The ratios of frequencies of notes in the chromatic (12-note) scale to the root note and to each other, e.g. just, equal, Pythagorean, etc.
- *Musical scale*: Series of notes that may employ fewer notes than the chromatic scale, e.g. blues, pentatonic, whole-tone, Indian ragas, etc. Or, the scale may have *more* notes, as with microtonal, 24-tet, and other experimental and alternative scales.

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.

What do Scala tunings affect?

Loading a Scala file in Filterverse affects the note and frequency waypoints available on the main Cutoff knob. The Quantize parameter in the Meta Knob and Random modulation sources provides access to a built-in list of scales, but Scala files can expand the possibilities for Cutoff tuning much further.

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

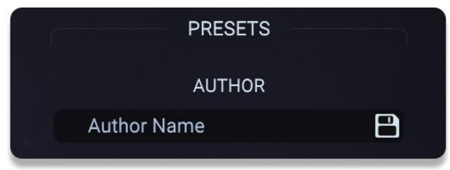
Pitch Wheel

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

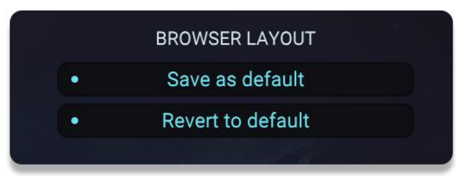
Author

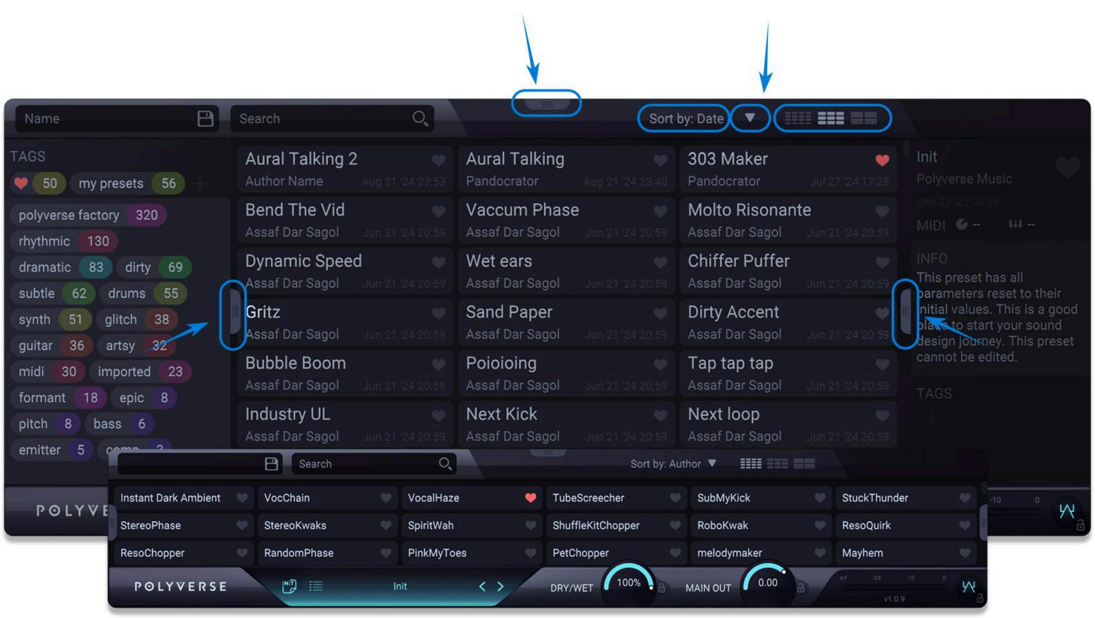
Here, you can set the default author name that will be recorded in the Preset Info when you save a Preset.



Browser Layout

You can save important aspects of how the Preset Browser looks and acts as the default settings that will load when you open an instance of Filterverse. These are:





- Height of browser when opened via the horizontal tab
- Width of Tags and Information columns when opened via their vertical tabs
- Sort by name, author, or date
- Sort in ascending or descending order
- Number of presets displayed via three “panes” options

This is accomplished using the following simple steps:

1. Open the Preset Browser
2. Adjust the various sizes and parameters to your liking
3. Close the Browser
4. Open Settings by clicking on the Gear icon
5. Separate *Revert to default* and *Save as default* options are now available
6. Click *Save as default* to save the Browser configuration you just created
7. Click *Revert to default* to abandon these changes and stay with the previous configuration

Note: You can also use the Revert and Save options at the lower left. However, the twin options in the Preset Browser Arrangement area of the Settings panel will save *only* the configuration of the Browser without affecting anything else.

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 still 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. Slope is expressed as a number of decibels (dB) for each octave Here 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: lowpass, highpass, bandpass, band-reject (notch), and peak. 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 or clone classic hardware versions of these filters. Instead, we have 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, and highpass.

Poles

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

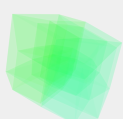
Drive

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, while based on the ladder topology in the digital domain, does not aim to emulate the Moog-style sound. Instead, it leverages our innovative new technology to achieve the best possible sound from this classic design.

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.

Evil

Self-oscillation: Yes
Saturation: Yes
Types: Lowpass, bandpass, highpass
Poles: 1, 2, 3, 4, 5, 6

Evil is dedicated to dirty, grungy filter sounds. This filter uses a Sallen-Key topology, enhanced with carefully crafted non-linearities to deliver a gritty, character-rich sound. Each of the six pole settings provides a different tonal experience. From fizzy saturation to a deep bass boost, Evil can add a little nastiness or a lot. Changing the input level can drastically alter how the filter sounds, because Evil reacts differently depending on the input signal level. Adjusting the gain isn't just about making it louder — it actually changes the tone in interesting ways. It’s important to note that in Evil, increasing gain is not the same as adding drive; gain and drive work together in unique ways to create a wider range of wicked, unpredictable sounds.

Type

On slider A, the Type slider parameter continuously morphs between lowpass, bandpass, and highpass filter responses. That makes Evil a true state-variable filter.

Poles

Slider B offers six different poly options, each providing a unique character ranging from fizzy resonance to sharp, pronounced filtering.

Drive

The Drive control on slider C introduces a unique distortion for each pole, distinct from simply increasing the filter’s input gain. Think of it like an elaborate, custom distortion pedal for each filter, adding harmonics and increasing the dirtiness and edge of the sound.

Damage

The Damage control on slider D allows the user to impart various imperfections to the filter. Each pole exhibits its own type of “damage”: One might emulate transistors with a deep “dead zone,” another may have malfunctioning capacitors, and still others might evoke circuit bleed or even some tasteful circuit bending. This adds character by introducing unpredictable, flawed behaviors that make each pole sound distinctively broken yet musically engaging.

Character table

The following table is our take on what each pole setting sounds like, and how each interacts with input gain, Drive, and Damage settings.

Name	Poles	Character	When pushed	Drive	Damage
Fizz	1	Fizzy, gritty	Gently saturates while keeping resonance	Dense, dynamic	Crosstalk between the components leads to unexpected results
Spit	2	Squelchy, spitty	Beefs up, resonance becomes warbly	Warm yet assertive, loves to interact with resonance	This setting is already naturally warbly. Damage makes it unnaturally so
Howl	3	Fat, juicy	Inflates and displaces resonance	Screaming high gain, loves to latch on to overtones	Transistors are of lesser quality at higher values
Glug	4	Bubbly, slightly wet	Gurgles and burbles	Vintage “smokey” dynamic; gets darker with the filter	The capacitors in this filter are failing. Can you smell it?
Phat	5	Sub shaker	Fattens, no resonance displacement	Raspy on the top end, heavy on the low	A solar storm is adding quantum disturbances
Warp	6	Hot and Nasal	Grunges up displacing resonance	Adds wavefolding	Soft bit-crusher

Word to the Wiz:

A Sallen-Key filter is so named because the original design is credited to R.P. Sallen and E.L. Key of the Massachusetts Institute of Technology. In 1955, they used vacuum tubes as operational amplifiers (op amps). Their filter produced a multiple feedback path and notably could avoid clipping during self-oscillation. The Sallen-Key topology was famously used in the Korg MS-20 synthesizer.

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.

Equalize

Self-oscillation: No
Saturation: No
Types: Low shelf, bell EQ, high shelf
Boost/cut: ±18dB

Equalize is a precision filter designed to give you exacting control over a signal's tone, allowing adjustments from subtle to dramatic. It features a low shelf, a high shelf, and a bell EQ type filter, each working across the entire frequency range but shaping the sound differently, making it versatile for various tonal adjustments. This filter has an amplitude range of ±18 dB.

This filter does not add any distortion to the sound and maintains a linear amplitude response, meaning it affects only the frequency balance without adding any color or character to the signal.

Frequency

The Frequency control (Cutoff knob) sets the center or cutoff frequency for the filter, determining which part of the audio spectrum is affected. This allows you to target specific tonal areas with precision.

Q

The Q control (Resonance knob) adjusts the width of the bell filter, or how steep the slope is for the low and high shelves. A higher Q value results in a narrower and more focused adjustment, while a lower Q value produces a broader, more gentle change.

Type

Slider A smoothly transitions between three filter types: low shelf, bell, and high shelf. This allows for seamless changes in the character of the equalization, making it easy to find the exact tonal adjustment you need.

Gain

The Gain control (slider D) is bipolar, allowing you to boost or cut frequencies by up to 18dB. This provides flexible control over the intensity of the tonal changes, whether you need subtle enhancement or a more drastic adjustment.

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 into 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.

The Valley

Self-oscillation: Yes
Saturation: Yes

The Valley is a high-slope band-reject filter that carves out a canyon of silence in the frequency spectrum. With steep 36dB/octave filters on both sides, it's perfect for aggressively cutting a swath through midrange mud or isolating spectral extremes. But what makes The Valley truly powerful is the creative control you get over the shape, motion, and behavior of that "gap."

At its core, The Valley is made of a highpass and a lowpass in parallel. The cutoff frequency sets the center of the rejected band, while the resonance sharpens its edges. From there, you can sculpt its contour with dynamic modulation or carefully dialed automation to create movement and space in complex soundscapes.

Unlike traditional notch filters, The Valley gives you asymmetric control over its shape. The **Anchor** and **Spread** parameters let you choose whether the rejection zone expands symmetrically or pivots from one side—useful for musical sweeps, shifting formants, or ducking specific harmonic content without losing the surrounding energy.

Drive this filter hard, and the **Drive** control adds post-filter saturation that creates harmonic splatter and midrange grunge. Combined with the **Balance** control, which lets you favor one filter over the other, The Valley can morph from a clean notch to a biting highpass or lowpass with extra attitude.

Cutoff

Sets the center frequency between the lowpass and highpass filters.

Resonance

Sharpens the edges of the rejected band. Adds emphasis to the transition points.

Anchor

Slider A sets the pivot behavior of the filters when spreading. At the center position, both the lowpass and highpass filters move symmetrically away from the cutoff frequency. At minimum, the lowpass remains fixed while the highpass moves upward. At maximum, the highpass stays in place while the lowpass moves downward. This control defines how the rejected band expands when adjusting the Spread parameter.

Spread

Slider B sets the distance between the two filters. Higher values widen the rejection band, from a narrow notch to a massive frequency gulf (up to 48 semitones, or 4 octaves).

Drive

Slider C pushes extra gain into the output stage post-filtering. This introduces rich intermodulation grunge and harmonic breakup, especially when the rejection band intersects with resonant or modulated signals.

Balance

Slider D sets the output mix between the highpass and lowpass filters. At the center position, both filters are heard at full amplitude. Pushing the slider upward reduces the lowpass level, emphasizing the high-frequency content. Pushing it downward reduces the highpass level, emphasizing the low-frequency content.

Vowel

Self-oscillation: Yes
Saturation: Yes

The vowel filter models a mouth by combining four different filters (see Type, below). The Cutoff knob controls the “formant shift” of the filter. For a larger sounding vocal tract, lower the cutoff knob. Adding more resonance makes the vowels more defined and easy to recognize.

Lips and Tongue

This filter models a mouth and allows it to position it in a way that it can produce most known vowels.

Slider A opens and closes the lips, while Slider B moves the back of the tongue forward and backward.

Tongue→ Lips↓	**Front**	**Mid**	**Back**
Open	a (hat)	ɐ (nut)	ɒ (not)
Mid	e (bed)	ə (bird)	o (or)
Close	y (free)	ɨ (goose)	u (boot)

Drive

Slider C pushes extra gain into the output post-filtering. As opposed to saturating the filters using the input gain, the effect combines a separate drive for each formant with an overall drive, for a more legible vocal distortion sound.

Type

These are all four-pole bandpass filters except for the following:

- LP: The lowest filter is lowpass
- PK: The lowest filter is lowpass and the highest is highpass
- BP: All filters are bandpass
- HP: The highest filter is highpass

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 the double-slit experiment famous from quantum physics. 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

Phasers comprise a type of audio effect that creates sweeping, swirling sounds by manipulating the phase relationships of an audio signal. This effect is achieved through a series of all-pass filters that alter the phase of different frequencies at varying rates. As the phase-shifted signal is mixed back with the original signal, it produces a series of peaks and notches in the frequency spectrum.

The phasers in Filterverse take this concept a level further with special types of phasers such as nested and contrary phasers. Each filter alters phase relationships in the audio signal, offering different effects from subtle phase shifts to more complex transformations.

What is an all-pass filter? As its name implies, it allows all frequencies to pass through, but towards the cutoff/center/tuned frequency, they're increasingly phase-shifted.

Word to the Wiz:

Classic musical examples of phasers include their use on analog string synths by such keyboard-centric artists as Gary Wright ("Dream Weaver") and Jean Michel Jarre. Phasers are also abundant on electric pianos across "yacht rock" — think "Minute by Minute" by The Doobie Brothers or "Babylon Sisters" by Steely Dan. But that represents only the beginning of a good phaser's capabilities.

Next Phase

Next Phase is a modern phaser that allows for dynamic and expressive modulation effects by creating multiple peaks and notches around the cutoff frequency. The resonance emphasizes the peaks of the filter, enhancing the effect's intensity and sharpness. Next Phase has Internal saturation that adds warmth and character to the signal by softly overdriving it.

Next Phase loves to be audio-rate modulated for some intriguing phase modulation effects.

Peaks

Slider A adjusts the number of resonant peaks within the phaser's effect, ranging from 2 to 8. By increasing the number of peaks, you can intensify the phaser effect and create more complex sound textures.

Spread

Slider B changes the distance between the peaks of the phaser effect. Adjusting this control changes the phase relationships and the spacing of the peaks, allowing the phaser to affect a wider or narrower range of frequencies.

Depth

Ranging from -100 to +100, Slider D sets the depth of the notches formed by the phaser. At +100, the effect doubles the number of notches and places them in opposite positions compared to -100, creating a more profound and distinct phaser effect. At 0, the phaser functions as an all-pass filter, providing a subtle phase shift without altering the overall tonal balance.

Deep Phase

Self-oscillation: No
Saturation: Yes
Type: Contrary dual phaser
Peaks: 2–8

Deep Phase consists of two phasers—tuned closely, but not identically—and mixes them with opposite polarity. This contrary-phase configuration causes overlapping frequencies to cancel out, leaving only the frequencies where the two phasers differ. The result is a very narrow set of bandpass filters, with the number and spacing of peaks determined by the user.

Unlike traditional phasers, which produce a series of moving notches, Deep Phase creates a set of isolated resonance peaks. These remaining peaks respond well to modulation and can create highly articulated motion within a sound.

Deep Phase loves to be audio-rate modulated for some intriguing phase modulation effects.

Peaks

Slider A adjusts the number of resonant peaks within the phaser's effect, ranging from 2 to 8. By increasing the number of peaks, you can intensify the phaser effect and create more complex sound textures.

Spread

Slider B changes the distance between the peaks of the phaser effect. Adjusting this control changes the phase relationships and the spacing of the peaks, allowing the phaser to affect a wider or narrower range of frequencies.

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 Deep Phase adds intermodulation grunge to the character of the output.

Fractazer

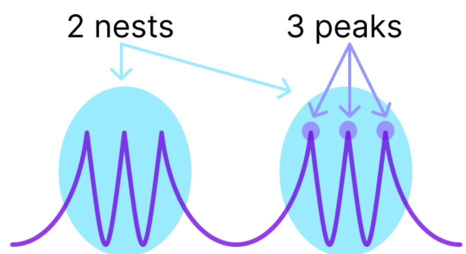
Fractazer is a fractally nested phaser — a phaser within a phaser within a phaser. Within the first all-pass filter there is a further chain of all-passes. Those nested filters determine the amount of “major” peaks. Within each of those nests there is yet another nested chain of all-pass filters. This provides for 66 frequency peaks in total.

The Cutoff knob shifts the frequencies of the peaks up and down.

Resonance adjusts the width of those peaks. This filter is a bit finicky when it comes to audio-rate modulation, so be careful!

Nests

Slider A adjusts the number of resonant peaks within the outer layer of the phaser, ranging from 1 to 6. By increasing the number of nests, you can intensify the phaser effect and create more complex sound textures.

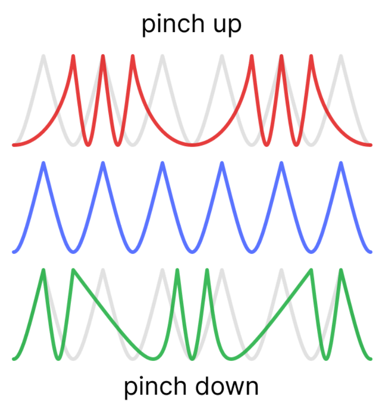


Peaks

Slider B adjusts the number of resonant peaks within the inner layer of the phaser, ranging from 1 to 11. By increasing the number of peaks, you add a peak to each of the nests, multiplying the complexity to the signal.

Depth

When Slider C is at zero, Fractazer is presented as an all-pass filter. Given that it can reach a very high order, it can add dramatic dispersion to the signal. Increasing Depth mixes the signal with the original to reveal the valleys and peaks in the frequency spectrum. Setting depth to maximum creates a deep phaser effect.



Pinch

Slider D squeezes the frequencies of multiple peaks within a nest together. At zero, the peaks are spaced evenly. High values bunch up the peaks; low values spread them out.

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.

Flange

Self-oscillation: Yes
Saturation: Yes
Types: Positive, Negative

Flange is a classic flanger, derived from a comb filter. This filter creates its effect through a delay that is recursively fed back into itself. The flanger is responsible for the "jet engine" effect sometimes heard in classic rock songs, though the one in Filterverse can do much more.

Unlike typical comb filters, Flange has five distinctive features:

1. **Feedback morphing:** It can seamlessly transition between positive and negative feedback. Positive feedback produces a saw-like tone, while negative feedback generates a square-like, hollow sound.
2. **Frequency range handling:** When increasing feedback, the Flange maintains the balance of the frequency spectrum, preventing the low end from becoming overly pronounced.
3. **Damping with phase correction:** It incorporates a damping filter that adjusts the phase to maintain the filter's pitch, ensuring the pitch remains consistent even with significant damping applied.
4. **Self-resonance:** It is capable of self resonance when pushing the Feedback parameter beyond 10 and even up to 12.
5. **Cross-feedback:** It's a delay after all, so why not cross feedback paths between the sides for some stereo subharmonics!

To achieve a classic flanger effect simply modulate the cutoff with an LFO, with the cutoff knob tuned at higher frequencies. For a chorus effect, tune the cutoff to lower frequencies.

Type

Slider A allows for smooth transitions between two types of feedback loops. The positive feedback loop enhances the sound with a saw-like quality, while the negative feedback loop gives a square-like, hollow characteristic to the sound.

Damp

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

Cross Feedback

Slider C crossfades between self-feedback per side and cross-feedback (left feeding right and vice-versa). This control is only relevant when the cutoff between the sides is different.

Vacuumb

Vaccumb is a multi-pole comb filter with inverted feedback. In subjective terms, this variant of a comb filter sounds like air being sucked through an empty tube. To change the tube's size just change the cutoff. This filter loves to be audio-rate modulated. It also has a very pronounced cutoff even with no resonance.

Poles

Slider A allows for continuous and smooth transition from 1 to 8 poles. The higher the pole count, the more pronounced the filter effect becomes.

Damp

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

Strings

The Strings filter utilizes a physical modeling technique called a waveguide to simulate the natural behaviors of three interacting strings. This approach allows for precise manipulation of the strings' physical properties through four primary parameters: Position, Damp, Timbre, and Detune, each designed to shape the sound authentically.

Warning: This filter might self-resonate if audio-rate modulation is applied.

Position

On Slider A, the Position parameter adjusts the excitation point along the string, akin to selecting where a string is plucked or bowed. Setting this parameter to zero percent places the excitation at the center of the string, producing a robust and resonant tone with a hollow quality. At 100 percent, the excitation moves close to the bridge, yielding a sharper, more piercing sound. This parameter allows for fine control over the tonal balance and timbral character of the strings.

Damp

Slider B affects the decay rate of higher frequencies, thus shaping the string's timbre by controlling the resonance duration. A higher Damp setting shortens the resonance of high frequencies, leading to a more subdued and darker tone.

Timbre

Slider C modifies the string's geometric uniformity, influencing the harmonic structure of the sound. At zero percent, the string is perfectly uniform, resulting in harmonic overtones. As Timbre is increased to 50 percent, the string exhibits greater geometric irregularity, introducing inharmonic (metallic and bell-like) overtones. At 100 percent, the string returns to uniformity.

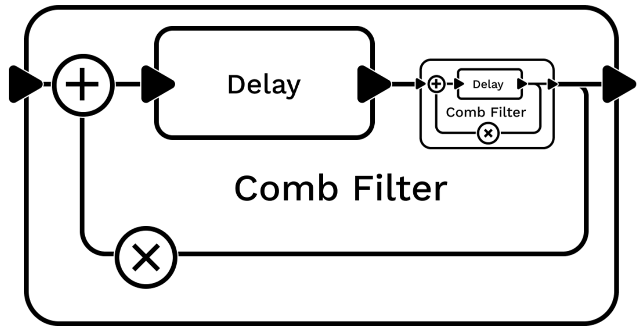
Detune

Slider D spreads the pitch of the three simulated strings in a stereo field, enhancing the spatial depth and width of the sound. The central string remains at its original pitch, while the left and right strings are detuned symmetrically away from the center. This parameter ranges from unison (no detune) to a full octave apart, allowing for subtle thickening effects to dramatic tonal variations.

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 lowpass 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!

Fibonacci Chorus

Self-oscillation: No
Saturation: Yes

Fibonacci Chorus is a dense, multi-voice modulation effect built from twelve delay lines, each modulated by its own low-frequency oscillator. All LFOs share the same frequency, controlled by the Frequency knob, but their phase relationships vary according to the Scatter parameter.

At minimum Scatter, the LFO phases fall into three rationally spaced groups, creating rhythmic patterns and gentle beating. As Scatter increases, the phase offsets shift toward the most irrational spacing possible: the inverse square of the golden ratio (≈0.3819). This eliminates pattern repetition, creating a blurred, detuned texture with no perceptible beats.

Each delay line includes internal feedback, controlled by the Resonance knob. When combined with modulation, this creates subtle resonant tails or warbly artifacts depending on the settings. The modulator waveforms are slanted to maintain balanced detune, and their shape can be smoothly morphed using the Crystal control.

Frequency

Sets the LFO modulation rate for all twelve delay lines, from slow to fast.

Resonance

Controls the amount of feedback in each delay line. Higher values produce resonant or metallic effects depending on the modulation depth and phase spacing.

Scatter

Slider A adjusts the phase distribution of the LFOs. At minimum, the twelve LFOs are grouped into three equally spaced phase clusters (rational 1/3 cycle). At maximum, their phases follow an irrational distribution based on $1/\varphi^2$ (≈0.3819), eliminating periodicities and creating a hazy, beatless detune.

Damp

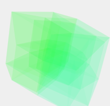
Slider B rolls off high frequencies in the feedback paths. Higher settings produce a darker, more subdued chorus. The effect is intensified when Resonance is high.

Crystal

Slider C morphs the LFO waveform from a smooth sine (left) to a sharp triangle (right). This shapes the modulation behavior and alters the perceived texture of the chorus.

Depth

Slider D sets the modulation depth. Higher values increase pitch modulation and detune intensity; lower values produce a more subtle chorus effect.



Word to the Wiz:

The golden ratio (φ ≈ 1.618) is often associated with aesthetics and natural growth patterns, such as those seen in sunflowers or shells (called phyllotaxis). Its inverse square (1/φ² ≈ 0.3819) has a unique mathematical property: it's maximally irrational, meaning it avoids alignment or repetition better than any rational number. When used to space LFO phases in Fibonacci Chorus, it ensures the modulation never loops back on itself, avoiding rhythmic overlaps and creating a smooth, non-cyclic detune. This gives the chorus its uniquely organic and diffuse sound.

Phyllotaxis:

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!

Defuse

Slider A 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.

Damp

Slider B 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 resonance values intensify the damping effect.

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.

Coils

This non-filter type models the behavior of the coils of a physical spring by summing multiple detuned comb filters together. When the comb filters' frequencies are close together, a characteristic sweep is heard thanks to the rate of the frequency difference between the filters.

Twist

Slider A adjusts the twist of the virtual springs, analogously to holding a coil spring in your hands and twisting the ends to make it more or less tense. Higher twist results in a more detuned sound with a faster downsweep and more reverb-like behavior. Lower twist values approximate a more common comb filter behavior.

Damp

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

Stereo

Slider C pans alternating coils in the virtual spring to the left and right, creating a wider stereo image.

Type

Slider D models different coil shapes. The inverse feedback combs (negative values) produce evenly shaped coils while the forward feedback combs (positive values) produce oddly shaped coils.

Decimate

Self-oscillation: No
Saturation: Yes

Decimate reduces the sample rate of incoming audio signals using various sample-and-hold techniques. It is capable of introducing quantization noise and aliasing, which can impart a gritty or “crunchy” texture characteristic like that of vintage digital hardware.

The Decimate filter includes several unique features that enhance its versatility and creative potential in sound design:

1. **Interpolation algorithms:** This filter offers a range of interpolation methods, from those that increase noise and aliasing effects, introducing sharp discontinuities, to smoother algorithms that help soften transitions between samples. Some algorithms even utilize complex waveforms to create continuous, character-rich sounds.

2. **Sampling techniques:** Decimate provides multiple sampling options, including traditional sample-and-hold, averaging, and filtering. These techniques allow detailed control over the type and intensity of aliasing and noise introduced into the signal, enabling precise textural adjustments.

3. **Bit crushing:** The integrated bit crusher feature in Decimate extends its functionality by offering linear and exponential crushing options. Users can apply bit crushing either before or after the decimation process, allowing for a wide range of effects from subtle saturation to aggressive distortion.

4. **Feedback mechanism:** A distinctive feature of Decimate is its feedback mechanism, which adds a resonant component to the effect, mimicking some properties of filters. This functionality broadens the scope of Decimate, enabling it to produce filtering effects without a conventional filter algorithm.

Decimate is ideal for adding distinct, textured elements to any musical piece. When used in conjunction with MIDI and audio-rate modulation, it unlocks exceptional possibilities for innovative sound design.

Sample Rate

The **Cutoff** knob adjusts the sample rate, from 21.10kHz at "open" all the way down to 20.60Hz. The lower you go, the more mangled the sound generally becomes.

Feedback

The **Resonance** knob adds the feedback mentioned above (feature 4). Turning it up increases the amount of processed signal fed back into Decimate's input. Do so, and you will hear a filter resonance-like effect of emphasized frequencies in the vicinity of the sample rate frequency. Some of the interpolators change the resonating frequency.

Interpolation

On Slider A, Decimate offers the unique ability to smoothly transition between ten different interpolation types, each designed to manipulate how samples are connected and transitioned within the audio processing. This allows for dynamic sound shaping and the ability to blend characteristics of different interpolations to achieve nuanced audio effects. Here's an overview of the interpolation types available:

1. **Square:** This method switches from an older sample to a recent sample mid-step, creating a pronounced, step-like discontinuity that results in a harsh sound texture.
2. **Saw:** Starts from an older sample and linearly ramps into the recent sample. Though smooth between individual samples, this method produces a harsh discontinuity when transitioning between pairs of samples.
3. **Step:** Implements the classic sample-and-hold technique, resulting in a staircase pattern that audibly mimics a stepping effect.
4. **Curve:** Adds a brief, smooth transition between samples, maintaining the step-like characteristic but softening the harshness of discontinuities.
5. **Linear:** Connects samples with a direct line, forming a continuous, triangle-like waveform that is smoother than the curve interpolation.
6. **Smooth:** Uses cubic interpolation to ensure a very smooth transition between samples, providing the softest transitions among the options.
7. **Sine:** Draws a half-sine wave between samples, introducing a unique, wavy transition that adds a subtle tonal quality.
8. **Sine X3:** Extends the sine interpolation by drawing one and a half sine cycles, enhancing the third harmonic and adding an extra wriggly transition between the samples.

9. **Sine X5:** Similar to Sine X3 it Extends the sine interpolation by drawing two and a half sine cycles, emphasizing the fifth harmonic adding an extra-bextra wriggly transition between the samples.
10. **Sine X7:** Dramatically extends the sine wave approach by drawing 3 and a half sine cycles between samples, adding an extra-bextra-schmextra, wriggly-schmigly transition between the samples.

Sample & Hold

Decimate allows selection among four sample-and-hold methods via Slider B, each varying in the degree of aliasing and frequency-mirroring effects they produce:

1. **S&H:** The traditional sample-and-hold approach that samples the signal at set intervals without pre-filtering, resulting in maximum aliasing.
2. **S&H + Pre-filter (SHF):** Adds a pre-filter before the sampling process, reducing the aliasing effect slightly.
3. **Average (AV):** Averages all sample values between steps, acting as a rudimentary filter to further decrease aliasing.
4. **Average + Pre-filter (AVH):** Combines averaging with a pre-filter to provide the most control over aliasing and frequency-mirroring effects.

Crush

Slider C controls Decimate's bit crusher, which can be adjusted from off (no effect) to a significant reduction in bit depth from 16-bit to 1-bit.

Position

Slider D adjusts where the bit crushing occurs in relation to the decimation process, and also the type of bit crushing algorithm used. Two factors figure into this:

Algorithm Selection:

- **Linear:** Applies a uniform bit reduction across all amplitude levels.
- **Exponential:** Provides more resolution to quieter sounds, enhancing perceived audio quality (Similar to μ-law encoding).

Position:

- **Pre:** Applies bit crushing before the decimation process, smoothing out the discontinuities and grit.
- **Post:** Applies bit crushing after decimation, emphasizing a grittier sound texture.
- **Both:** Combines pre and post crushing for extensive sonic manipulation.

Slider D steps through combinations in the following order from bottom to top:

- **PrL:** Pre Linear
- **PrE:** Pre Exponential
- **PoL:** Post Linear
- **PoE:** Post Exponential
- **BoL:** Both Linear
- **BoE:** Both Exponential

Word to the Wiz:

In digital audio, the higher the sample rate, the higher the frequencies that can be reproduced. Bit depth, on the other hand, is about dynamics: the higher the bit depth, the finer the gradations between loud and soft that are captured. Intentionally reducing the sample rate and/or bit depth can create the "lo-fi" effect of vintage samplers, drum machines, and video game consoles.

BarberPole

Self-oscillation: No
Saturation: Yes
Direction: PH, UP, DN, UP +, DN +

BarberPole is a resonant phase shifter that can operate as a musical allpass filter, a Shepard tone phaser, or a frequency shifter—depending on the selected mode. Unlike traditional phasers or frequency shifters, BarberPole allows precise control over the spacing of resonant peaks in semitone intervals, enabling harmonically tuned textures and complex spectral illusions.

In **PH** (phase) mode, BarberPole acts as a static phase shifter. The phase wraps around a full cycle from -360° to +360°, but due to its circular nature, the effect is perceptually identical at minimum, center, and maximum settings. When **Resonance** and **Depth** are set to zero, the filter functions as a true allpass. Audio rate modulation at this mode allows for phase modulation (Yamaha-style FM) on any audio source.

In **UP** and **DN** modes, the phase is continuously rotated forward or backward in a **tempo-synced** cycle. This produces a Shepard tone-like effect—an auditory illusion of an infinitely rising or falling phaser. These modes are particularly useful for evolving textures or sweeping filter effects with no perceived reset point.

In **UP +** and **DN +** modes, BarberPole becomes a **frequency shifter**. Unlike pitch shifting, frequency shifting adds or subtracts a fixed frequency value, which transforms harmonic content into enharmonic structures.

BarberPole's filter peaks are spaced in musical intervals, defined by the **Interval** control. This allows the effect to follow harmonic steps like whole notes or octaves imparting a uniquely musical character to this phaser.

Cutoff

- In **PH** mode: cutoff sets the static phase position in degrees (–360° to +360°). This parameter can also be modulated by pitch-detection or MIDI-note modulators for harmonic tracking.
    - When **Interval** is set to 12 (one octave), a cutoff modulation amount of 60% results in a repeating octave cycle—each octave sounds identical.
    - When **Interval** is set to 6 (a tritone), a modulation amount of 120% completes a full cycle every tritone, creating more exotic harmonic motion.

- In **UP / DN**: selects tempo divisions for the rotation speed.
- In **UP + / DN +**: sets the frequency shift amount.

Resonance

Adds resonant peaks to the filter response.
In **PH** peaks remain static. In **UP / DN**: peaks move slowly upward or downward. And in **UP + / DN +**: peaks move rapidly, creating complex ringing timbres.

Direction

Slider A Selects the mode of operation:

- **PH (Phase):** static phase shift.
- **UP (Up Tempo):** slow upward phase rotation (Shepard rise).
- **DN (Down Tempo):** slow downward phase rotation (Shepard fall).
- **UP + (Up Shift):** high-rate upward frequency shift.
- **DN + (Down Shift):** high-rate downward frequency shift (through-zero shift).

Interval

Slider B Sets the spacing between resonant peaks in semitone intervals.
Higher values (e.g., 12 semitones) yield fewer peaks. Lower values (e.g., 2 semitones) create dense harmonic clusters.

Stereo

Slider C Applies inverse phase modulation between the left and right channels. At maximum, the sides are 180° apart. Even small values create noticeable spatial movement.

Depth

Slider D mixes the dry signal with the phase-shifted signal.
At 0: in PH mode, the effect is inaudible without resonance; in **UP / DN**, it produces mild frequency shift artifacts; in **UP + / DN +**, it functions as a classic frequency shifter.
At higher values it emphasizes cancellation notches in PH and tempo modes, and creates a more pronounced, warbly character in frequency-shifting modes.

Word to the Wiz:

Frequency shifting adds a fixed amount to all frequencies, breaking harmonic relationships—unlike pitch shifting, which multiplies them. BarberPole stands apart by allowing the shift amount to track incoming pitch, either via MIDI or pitch-detection modulators. This transforms what's usually a metallic, enharmonic effect into something strikingly musical. Combined with the Interval control, this makes BarberPole the most harmonically aware frequency shifter we've ever heard. Try modulating it with melodies—you might be surprised where it takes you.

Cabinator

Cabinator is a guitar/bass amp cabinet emulator built using a 16-node graphic EQ, tailored to replicate the frequency response of various classic (and not so classic) amplifiers. Cabinator features eight guitar amplifiers and two bass amplifiers, giving you a wide range of tones to choose from. Additionally, Cabinator includes a microphone/room simulation, lending a sense of depth and ambience. The Gain and Tone controls let you shape the sound even further.

Size

The Cutoff knob alters the pitch of the amplifier's formants in a parallel fashion, allowing you to simulate a larger or smaller speaker cabinet or even mimic the effect of sound moving faster or slower through air. This gives you a dynamic way to craft your ideal cabinet tone. With this control, a little bit goes a long way! The zero position of the knob reflects the original amplifier size; higher values produce smaller cabinets with more high-frequency response. Lower values correspond to larger, darker cabinets.

Cabinate

The Cabinate (resonane) control adjusts the intensity of the cabinet emulation by boosting the peaks and deepening the valleys of the selected amplifier's frequency response. At 100%, the cabinet sounds as it was originally captured. Reducing the value moves the tone closer to a direct signal, as if the amp were bypassing the speaker entirely. Increasing the value above 100% exaggerates the cabinet's tonal imprint—similar to running the signal through the same cabinet multiple times. This control allows you to dial in anything from subtle coloration to an overemphasized, stylized cabinet tone.

Type

On Slider A, the Type selector allows you to select from ten different cabinet models, with the "*Bmes*" and "*Bamp*" types designed for bass and the remaining eight selections intended for guitar.

Mic

Slider B adjusts the proximity of the virtual microphone to the amplifier. The greater the distance, the more diffuse the sound. This adds a touch of ambience and stereo liveliness to the overall tone.

Gain

Slider C lets you drive more level into Cabinator, enhancing its punch and presence. When pushed, it can overdrive the cabinet for a more aggressive and gritty tone.

Tone

Slider C is a simple cut/boost that balances highs and lows, letting you deepen or brighten the overall sound of Cabinator.

Delays

In its simplest form, a delay applies a time lag to an incoming signal. This can happen once or multiple times in succession. In musical applications, a delay is usually mixed with the original undelayed signal. Multiple delays can combine to create more complex audio effects, from repeating echoes at longer delay times (above 30ms or so) to flanging and reverberation at shorter time settings.

The earliest forms of delay recorded a portion of audio onto a tape loop. The loop could then be played back alongside the original signal; changing the position of the playhead adjusted the length of the delay. Later, analog “bucket brigade” delays would pass the signal from one capacitor to the next, with the time it took for a capacitor to charge and discharge creating the delay. Later still, digital delays sampled the snippet of audio into memory. It could then be played back indefinitely without degradation.

A delay in its most basic form is an all-pass filter. When using a short delay time and mixed with the original, it becomes a comb filter. The border between delays and filters is very blurry, and we are happy to blur this border even more!

Filterverse has several delays as options for any of its three filter tabs. We believe that in many cases, they offer more sophisticated options than dedicated delay plug-ins.

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 in 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. This parameter is quantized, ensuring that the delay is always on beat.

Feedback

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

Low Damp

Slider A a highpass filter in the feedback path. It reduces the low-frequency content in every repetition of the delay.

High Damp

Slider B adjusts the lowpass filter in the feedback path. This reduces the high-frequency content in every repetition of the delay.

Cross Feedback

Slider C crossfades between self-feedback per side and *cross* feedback, i.e. the left side of the signal feeding right and vice-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.

Tempong

Tempong is a tempo-synced ping-pong delay. “Ping-pong” is a term for a delay where successive repeats alternate between the left and right sides of the stereo picture. Tempong expands upon this with a number of parameters that let you determine the ping-pong behavior with great precision.

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 icon can change if Time is being modulated. This parameter is not quantized and allows the tuning to “in-between values.”

Feedback

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

Pattern

Slider A smoothly crossfades between different delay behaviors with regard to alternating between stereo sides.

- LR: Left and right, i.e. the most common ping-pong bounce
- LCR: Left-center-right
- LCRC: Left-center-right-center
- LLRR: Left-left-right-right, i.e. two repeats/taps on each side before moving
- X: Above each pattern on the slider is a cross-feedback option where the left side of the signal feeds the right and vice-versa

The following chart illustrates what you will hear at each setting. Note that the slider is continuous and smooth with regard to “in-between” values.

LR

L	L		L		L		L		L		L		L		L	
R		R		R		R		R		R		R		R		R

LR Cross (X)

L	L		R		L		R		L		R		L		R	
R		R		L		R		L		R		L		R		L

LCR

L	L		L	L		L	L		L	L		L	L		L	L
R		R	R		R	R		R	R		R	R		R	R	

LCR Cross (X)

L	L		L	R		R	L		L	R		R	L		L	R
R		R	R		L	L		R	R		L	L		R	R	

LCRC

L	L	L		L	L	L		L	L	L		L	L	L		L
R		R	R	R		R	R	R		R	R	R		R	R	R

LCRC Cross (X)

L	L	R		L	R	L		R	L	R		L	R	L		R
R		L	R	R		R	L	L		L	R	R		R	L	L

LRRR

L	L			L	L			L	L			L	L			L
R		R	R			R	R			R	R			R	R	

LLRR Cross (X)

L	R			L	R			L	R			L	R			L
R		R	L			R	L			R	L			R	L	

Damp

Slider B controls a lowpass filter in the feedback path, which reduces high frequency content in each repeat of the delay

Stereo

Slider C smoothly crossfades between the sides of the patterns. At its extremes, left-to-right becomes right-to-left and vice-versa. Set in the exact middle, the delay taps/repeats appear on both sides simultaneously (while still remaining stereo).

Div

Slider D smoothly transitions between “straight,” dotted, and triplet rhythmic subdivisions. This value is not quantized.

TemPitch

TemPitch is a pitch-shifting granular delay that is tempo-synchronized. It works by cutting the audio into beats and playing them in different pitches at different times. Its effects can range from tape delay and chorus all the way to twinkly granular particles, offering a truly diverse and musical effect.

Time

The Cutoff knob is repurposed as the Time control. This lets you select common tempo divisions — quarter-, eighth-, and sixteenth-notes — while the **Div** slider (see below) introduces dotted and triplet rhythmic subdivisions for extra flexibility.

Feedback

The Feedback control (Resonance knob) manages the amount of delayed signal fed back into the Tempitch filter. At higher settings, the delay can run away with itself, however an internal saturation protects it from getting out of hand.

Pitch

Slider A adjusts the pitch of the grain from -24 to +24 semitones. The change occurs within the feedback path, creating a cascading pitch-shifting effect where the pitch change accumulates with each repetition, allowing for creative manipulation of delay repetitions.

Damp

Slider B controls a lowpass filter in the feedback path, which reduces high frequency content in each repeat of the delayed signal.

Cross Feedback

Slider C crossfades between self-feedback per side and *cross* feedback, i.e. the left side of the signal feeding right and vice-versa. This control is only relevant when the delay times or pitches between the sides are different.

Div

The Div control (slider D) skews the rhythmic synchronization towards either dotted or triplet subdivisions, providing extra rhythmic flexibility for creative timing.

yaleD opmeT

yaleD opmeT is a reverse granular delay that is tempo-synchronized. It works by cutting the audio into beats and playing them in reverse, allowing you to control the timing and pitch of the reversed grains. Its effects can range from soothing ambient smears through vinyl scratches all the way to granular mayhem, offering a truly dynamic and expressive effect.
The Cutoff knob is repurposed as the Time control. This lets you select common tempo divisions — eighth-, quarter-, and sixteenth-notes — while the **Div** slider (see below) introduces dotted and triplet rhythmic subdivisions for extra flexibility.

Feedback

The Feedback control (Resonance knob) manages the amount of delayed signal fed back into the yaleD opmeT filter. At higher settings, the delay can run away with itself, however an internal saturation protects it from getting out of hand.

Pitch

Slider A adjusts the pitch of the grain from -24 to +24 semitones. The change occurs within the feedback path, creating a cascading pitch-shifting effect where the pitch change accumulates with each repetition, allowing for creative manipulation of delay repetitions.

Damp

Slider B controls a lowpass filter in the feedback path, which reduces high frequency content in each repeat of the delayed signal.

Phase

The Phase control (slider C) adjusts the phase of each grain of audio in relation to the beat-sized grain, ranging from -360° to +360°. This gives users a modulation range of 720°, allowing for precise manipulation of the phase relationship and adding unique movement to the delay effect.

Div

The Div control (slider D) skews the rhythmic synchronization towards either dotted or triplet subdivisions, providing extra rhythmic flexibility for creative timing.

Granular

Granular effects operate by dividing an audio signal into small segments called grains, typically ranging from 1 to 250 milliseconds in duration. Each grain captures a tiny snapshot of the audio and can be shaped, modulated, and reassembled in near-infinite ways. Unlike traditional delay or modulation effects, which work on continuous audio streams, granular processing treats sound as a stream of discrete particles—allowing for non-linear playback, precise manipulation, and real-time recomposition.

Once extracted, grains can be triggered at different times, played forward or backward, detuned, overlapped, scattered across the stereo field, or even crossfaded for smoother results. This level of control makes it possible to slow down audio without lowering its pitch, or stretch it out while preserving transient detail. By modulating the grain playback position, grain size, pitch, and density, granular effects can produce a wide range of results—from naturalistic time expansion to extreme sound design textures.

Granular processing is especially useful when you want to disassemble audio into textures, extract rhythmic patterns, smear or blur transients, or generate synthetic ambiences from acoustic material. In more rhythmic contexts, grains can be locked to tempo, allowing for stutter, repeat, or swing-based effects that remain musically grounded. And when grains are taken far out of phase with their source, the results can resemble granular synthesis—blending delay, filtering, and resynthesis into one cohesive tool.

Time Warp

Self-oscillation: No
Saturation: No
Type: Granular delay

Time Warp is a granular delay that allows precise manipulation of time within a fixed tempo grid. Each incoming signal is sliced into grains, which are then repositioned, reversed, stretched, or shifted in real-time. This filter enables a wide range of time-based effects including rhythmic stutters, reverse playback, swing reshaping, tape stops, scratch emulation, and granular clouds.

The Cutoff knob determines the grain size in tempo-synced increments, setting the base rhythmic resolution of the effect. Short grain sizes result in tight, comb-like filtering or glitch effects, while longer grains lend themselves to slower tape-speed manipulations and reversals.

Feedback is applied independently to the delay buffer, allowing grains to re-enter the system and accumulate over time. This creates everything from dense, washed-out textures to comb filtering or cascading pitch effects when used with modulation.

Cutoff

Sets the grain length in tempo-synced values. Short grains create tight rhythmic effects; longer grains allow for time warping and pitch-shifting textures.

Resonance (Feedback)

Controls the amount of signal fed back into the delay buffer. At low values, grains do not repeat. At higher values, feedback introduces rhythmic repetition, granular smearing, comb filtering, and cascading pitch effects when used in conjunction with **Direction** modulation.

Direction

Slider A controls the playback direction and speed of audio within each grain. At it's minimum, grains play forward at normal speed. As the value approaches the midpoint, playback slows, and pitch drops within each grain. At the center position, the audio comes to a halt, allowing for vinyl-like effects when modulating the Cutoff. Above the midpoint, playback resumes in reverse, gradually speeding up. At maximum, grains play in reverse at original pitch and tempo. This parameter does not affect grain size—only the behavior of audio playback within each grain.

Texture

Slider B adjusts the crossfade speed between successive grains. At minimum, the crossfade matches the full grain length, producing smooth and seamless blending. As the value increases the crossfade becomes shorter emphasizing grain edges for a more percussive, rhythmic result.

Time

Slider C modulates the grain playback position in time. Grains are displaced or repositioned within the delay buffer based on modulation input.
This parameter is highly responsive to modulation and is typically used with 100% depth for unipolar modulators or 50% depth for bipolar ones. The overall time range is defined by the Range control.

Range

Slider D sets the maximum time displacement range of the Time parameter. The values are quantized to tempo divisions to keep modulation rhythmic and controllable.
Combined with the Cutoff and Time controls, Range allows for precise design of effects such as stutters, pitch ramps, sudden reversals, or slow-motion playback.

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.

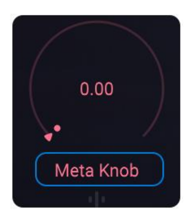
Meta Knob

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

Polarity

The button at bottom center changes the knob's type. Knobs can be *positive-unipolar* (ranging from zero to 100 percent and marked "+"), *negative-unipolar* (ranging -100 percent to zero and marked "-"), or *bipolar* (ranging -100 percent to 100 percent).

Rename the knob



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, Resonance, and other parameters in quantized amounts or according to a selection of various musical scales and intervals — which are commonly used to modulate frequency-affecting parameters.. 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 percent).

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 percent, 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 percent once a MIDI note is received.

Decay

Adjusts the amount of time it takes for the envelope signal to drop from 100 percent 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.

Vel Sens (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

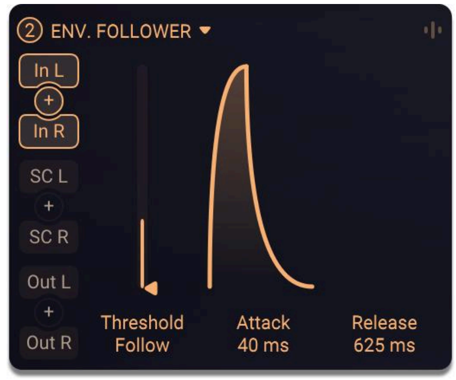
Similarly to the Glide setting in the Meta Knob source, this slows and smooths out transitions between incoming MIDI or CV and their effect on assigned destination parameters.

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



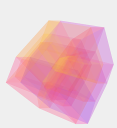
An envelope follower listens to the amplitude of an incoming audio signal and converts it into a modulation signal — often a filter cutoff in the case of envelope-following effects pedals. In Filterverse, it can modulate any destination you want.

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.

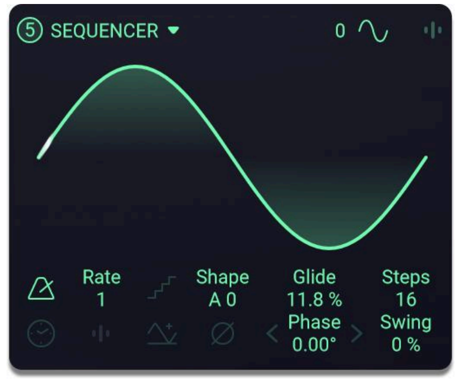
Attack

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 percent 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.

Clicking on the pattern icon selector will select the next group of shapes. Right clicking will select the previous group of shapes. Dragging it up or down will select the first of each group.

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.

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.

Rate modulation

Notice the familiar pop-up sliders button 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.

Step Mode

Step Mode determines how the Rate value applies to the sequencer's timing. When enabled, each individual step in the sequencer lasts as long as the Rate value, meaning the Rate sets the duration of a single step. When disabled, the entire sequencer cycle fits within the Rate duration, dividing that time equally among all steps. This allows for flexible rhythmic behavior — enabling complex polymetric and polyrhythmic patterns by stretching or compressing step timing relative to the Rate value.

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 Retrig is active, the LFO will sync to the beginning of each incoming MIDI note.

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).

Invert

Flips the polarity of the sequencer output. When Unipolar is off, this fully inverts the signal around zero. When Unipolar is on, the inversion occurs within the positive range only, effectively mirroring the sequence while keeping all values above zero.

Phase

Shifts the phase of the sequencer output from –360° to +360°, allowing precise control over the timing of modulation events. Small adjustments can make the sequence feel slightly ahead or behind, while larger shifts (such as ±90° or 180°) can be used to create stereo movement or phase-based interactions with other modulators. Use the arrow buttons next to the control to nudge the entire sequence forward or backward by one step at a time—rotating the sequence so that each step moves to the next position, with the last step wrapping around to the beginning.

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.

Steps

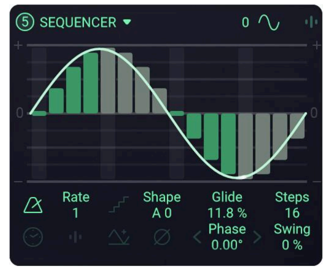
Drag on this field to set the number of steps in the sequence, from 2 to 16.

Swing

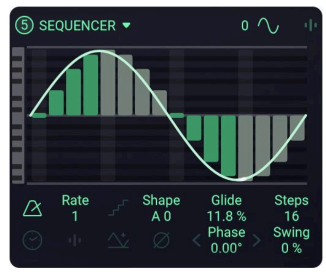
This introduces a characteristic “behind the beat” feel. Musically speaking, what’s going on is that certain rhythmic subdivisions (typically eighth- and sixteenth-notes) have alternating longer and shorter durations even though they wouldn’t be written as such in conventional music notation. Think of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

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.
- Use the number keys on your computer keyboard to switch between the main patterns in increments of 10.
- **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

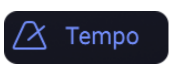
As its name implies, the Random pattern generator is capable of randomizing tempo divisions and note quantization, then converting the resulting output to a modulation source.

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 

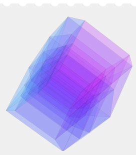
The familiar pop-up sliders button is located directly beneath the Rate control. This allows the Sequencer/LFO Rate to be modulated by any of the active sources, including the Sequencer itself. Click on the icon to open the pop-up sliders for the Rate.

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 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 percent.

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 percent. (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, or 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.

Noise

Switches between Noise mode and Oscillator mode. In Noise mode the Oscillator generates bandpassed white noise.

Coarse

The Coarse control sets the pitch of the Oscillator in semitones. Note the pop-up sliders button under the control; this lets you modulate the coarse pitch. The default pitch when Coarse is set to zero (12 o'clock) is *E4*. This also controls the white noise filter cutoff allowing you to "play" the filtered noise with your keyboard.

Pitch modulation

Clicking the icon directly below the Coarse knob opens the pop-up sliders. Adjusting these lets you modulate the coarse pitch with any active source, including the Oscillator itself.

Fine

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

Factor

Sets the natural number multiplier of the frequency (as set by the Coarse and Fine controls) for scanning between overtones. This is useful for preserving harmonic relationships between the oscillator and the source content.

Shape

Continuously crossfades the Oscillator waveform from sine (0 percent) through triangle, saw, square, and a thin pulse wave (100 percent). In Noise mode, this controls the amount of resonance of the white noise filter.

Shape modulation

Note the pop-up sliders that appear beneath the Shape control. This means the shape can be modulated by any of the active sources, including the Oscillator itself. Click on the icon to open the [pop-up sliders]() for the Shape.

Phase

Adjusts the phase of the oscillator between -360° to 360°, for a total of 720°. This can be used both for adjusting the phase of the starting point of the oscillator when retriggering, or for phase modulation and distortion techniques. When in noise mode this will set the seed of the noise when it retriggers.

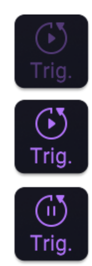
Phase modulation

Another pop-up sliders button is located beneath the Phase control. This means the phase can be modulated by any of the active sources, including the oscillator itself. Click on the icon to open the [pop-up sliders]() for Phase.

Trig.

This function only works with MIDI. it will retrigger the oscillator to its starting phase (set by the phase control) when activated. It has three modes:

1. Button dark: No retrigger
2. Button lit, play icon in center: Retrigger engaged
3. Button lit, pause icon in center: One-shot; oscillator plays one cycle from starting phase then pauses (for drum design and filter pinging)



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.

Audio Out

Turning this control up routes the oscillator audio into the filter, so you can hear and process it (in addition to using it as a modulation source).

While this doesn't make Filterverse a full-fledged synthesizer, it can act like one to some extent here. Hearing the oscillator alongside the source content a filter type is modulating makes for some interesting creative possibilities — particularly if that filter and the oscillator pitch are modulated rhythmically by the same source. So, experiment boldly!

Audio out modulation

As with most other controls in the Oscillator, the level of audio output to the filter can be modulated by other active sources, including the Oscillator itself. Click the icon below the knob to open the pop-up sliders.

Pan

The audio output of the oscillator can likewise be panned in the stereo field. To hear audible results, the Audio Out knob must be set to a nonzero value. The Pan control has no effect on the Oscillator's use as a modulation signal.

Pan modulation

Just like with the audio output, the stereo panning can be modulated by other active sources, including the Oscillator itself. Click the icon below the knob to open the pop-up sliders.

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 for the Quantize setting in both the Meta Knob and the Random Generator.

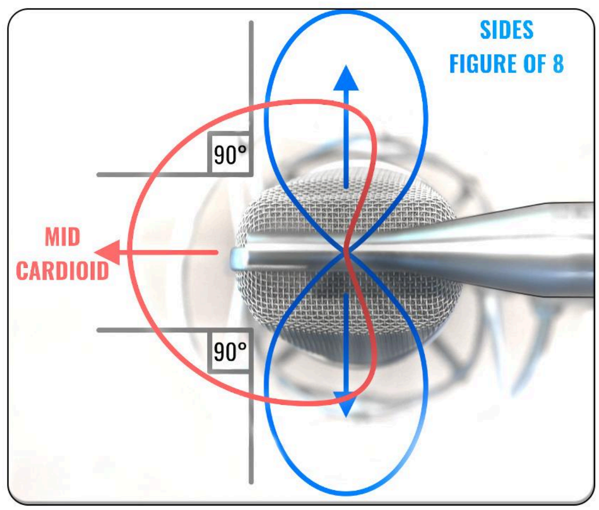
Value	**Description**	**Notes** (assuming C root)
Off	No quantization applied	None
Chromatic	12-tone chromatic scale	C, C#, D, D#, E, F, F#, G, G#, A, A#, B
Octatonic a	8-tone symmetrical scale starting with a minor second	C, C#, D#, E, F#, G, A, A#
Octatonic b	An 8-tone symmetric scale starting with 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

M/S 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 M/S 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.

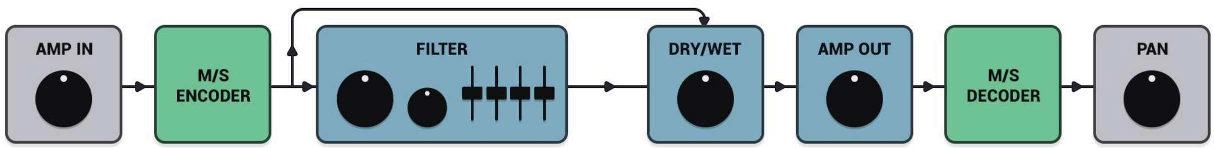
Common pitfalls and misconceptions

- **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 M/S 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 90-degree phase relationship between Mid and Side can lead to phase issues and a compromised stereo image.
- **Incorrect source material:** Using M/S processing on stereo tracks where the left and right channels are unrelated can result in an 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 M/S processing in Filterverse for a balanced and phase-coherent stereo mix.

Mid/Side in Filterverse

Each of the three filters in Filterverse has an M/S 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 M/S domain.



Before the signal reaches the filter, it is **encoded from stereo to M/S**.

Once in M/S 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 M/S 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-oscillation

Some filters in Filterverse are capable of self-oscillation (a.k.a. self-resonance), which means they can generate their own tone at high resonance settings. While this feature can be creatively useful, it requires careful handling in M/S processing, particularly with mono signals.

When a mono signal is processed in M/S 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.

Thank you for using Filterverse!

Have fun!