

Filterstation User's Guide

Audio Damage, Inc.

Release 2.0



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Introduction

Filterstation exists because of a simple premise: everyone needs some good filters. While filters have always been central features of analog synthesizers, they're now used on just about everything in modern music production: vocals, drums, guitar, cello, whatever. Anything that makes noise can make more interesting noise if you add a filter. A static filter can gently or radically shape the tone of an instrument, while a filter modulated by an LFO can create anything from subtle animation to wild, screeching sweeps.

Filterstation satisfies this need by providing two filters in one plug-in, each able to operate in twelve different modes. A simple but flexible low-frequency oscillator, with optional tempo sync, and an envelope follower set Filterstation's filters in motion. Filterstation's clean, direct user interface won't get in the way of applying exciting filter effects to your music.

New in Version 2.0

Version 2.0 improves Filterstation in many ways:

- A completely new user interface. No more fiddly digit-based controls.
- New sound-mangling controls: Feedback, FM, and Hack
- The envelope follower and side-chain input now operate in stereo.
- The user interface is resizable and appropriate to contemporary display technologies.
- Filterstation is now compatible with AAX hosts.
- Presets use a cross-platform XML-based file format.
- Filterstation is built with up-to-date code libraries for better host compatibility and future-proofing.

Filterstation version 2 has a different name and internal identifier than version 1. This means that you can install version 2 alongside whatever version you are currently using, and continue to use the older version in your existing projects. Version 2 cannot be directly substituted in existing projects and cannot read preset files created by version 1.

Also, direct MIDI control assignment has been removed. Host DAWs have progressed considerably since Filterstation's last major revision and it's now appropriate to leave MIDI mapping up to them.

System Requirements

Filterstation is provided as 32- and 64-bit VST2 and VST3, and 64-bit AAX plug-ins for Windows. On OSX, Filterstation is provided as Universal Binary Audio Unit, VST2, VST3 and AAX plug-ins.

Filterstation is a plug-in, not a standalone application. To use it, you'll need a host application such as Ableton Live, Steinberg Cubase, Apple Logic, Avid ProTools, etc¹. Obviously you'll also need a computer capable of running one of these applications. We assume that you are familiar with using plug-ins with your host. If you have general questions about using plug-ins with your host, please refer to its documentation.

We support the use of Filterstation under Microsoft Windows 8 or newer, and Apple OS X version 10.8 or newer.

Installation

To install Filterstation, double-click the Filterstation Installer icon, and follow the instructions. You can choose which plug-in formats to install and, for some formats, the plug-in destination folder.

To un-install from OS X, simply delete the plug-in from your VST folder, which is usually located at `/Library/Audio/Plug-Ins/VST/Audio Damage`, and your Audio Units folder, which is located at `/Library/Audio/Plug-Ins/Components/`. To un-install from Windows, delete the file named `Filterstation2.dll` from your VST folder(s). The presets are stored in separate files which you can also delete, although they occupy very little space. On OS X, they're in `/Library/Application Support/Audio Damage` . On Windows, they're in the hidden folder `C:/Program Data/Audio Damage/` .

¹ Product names are copyrighted by their respective owners.

Filters: The Basics

If you're reading this manual you probably already have at least a basic understanding of what filters do and why you need them. We're going to cover the basics here anyway, just to make sure that we have a common vocabulary. We won't be offended if you skip this section altogether.

Most filters, at their core, have just two simple controls: frequency and resonance. The frequency control usually sets the frequency at which the filter starts to change the tone or timbre of the signal. If you hear a filter effect that you might describe as making the sound "move up and down", you're probably hearing the filter's frequency control increasing and decreasing. The exact effect of the frequency control depends on the filter type, which we'll get to shortly.

The resonance control usually makes the filter's overall effect more dramatic, sharper, and/or more electronic sounding. A high resonance setting usually makes the filter boost the level of frequencies near the filter's frequency value, creating distinct overtones and sometimes a whistling-like sound. Some filters will oscillate, generating an output with no input signal, at their highest resonance settings.

Filters come in several different types which differ in how they respond to different audio frequencies. The *low-pass* filter is the filter type most commonly found in analog synthesizers. A low-pass filter attenuates or reduces signal frequencies greater than its cutoff frequency and passes signal frequencies below its cutoff frequency without alteration.

A *high-pass* filter works in a manner opposite to that of a low-pass filter: it reduces signal frequencies less than its cutoff frequency and passes signal frequencies above its cutoff frequency without alteration.

A *band-pass* filter allows a range of frequencies centered on the cutoff frequency to pass and attenuates higher and lower frequencies. The width of the band of frequencies is controlled by the Resonance control. A higher resonance setting narrows the width of the band, letting through a narrower range of frequencies.

A *notch filter*, sometimes called a *band-reject* filter, is the opposite of a band-pass filter. It attenuates a range of frequencies centered on the cutoff frequency, and allows higher and lower frequencies to pass unaltered. In other words, the filter cuts a notch in the frequencies of the signals that pass through it. The width of the notch is controlled by the resonance control; a higher resonance value makes the notch narrower, cutting fewer of the frequencies. This means that, unlike the other filter types, increasing the resonance setting of a notch filter usually creates a less-dramatic filtering effect.

Filters are also often characterized by their number of *poles*². The more poles a filter has, the more it attenuates frequencies. For instance, a 4-pole low-pass filter will attenuate frequencies above its corner frequency more than a 2-pole low-pass filter with the same frequency setting. While Moog synthesizers are justifiably famous for their 4-pole low-pass filters, filters with fewer poles also have their uses. Most early Korg and Oberheim analog synthesizers, for instance, employed 2-pole filters.

Filterstation contains all the aforementioned filter types, most with variants with different numbers of poles. There are also a few that have the same basic type and number of poles, but individual sonic characteristics, i.e. different variations on a theme.

Operation

If you're at all familiar with synthesizers, either the hardware or software variety, you'll find Filterstation familiar. Briefly, Filterstation is a pair of filters controlled by an LFO and an envelope follower. This simple description belies its flexibility, which lies in its three different routing modes, twelve different filter types, and independent modulation paths.

In the following pages we'll describe each of Filterstation's controls. Filterstation is not a terribly complex plug-in, but some of its controls are interrelated and explaining one necessarily involves referring to others. Hence you will find that sometimes we refer to a control in passing before later describing it in detail.

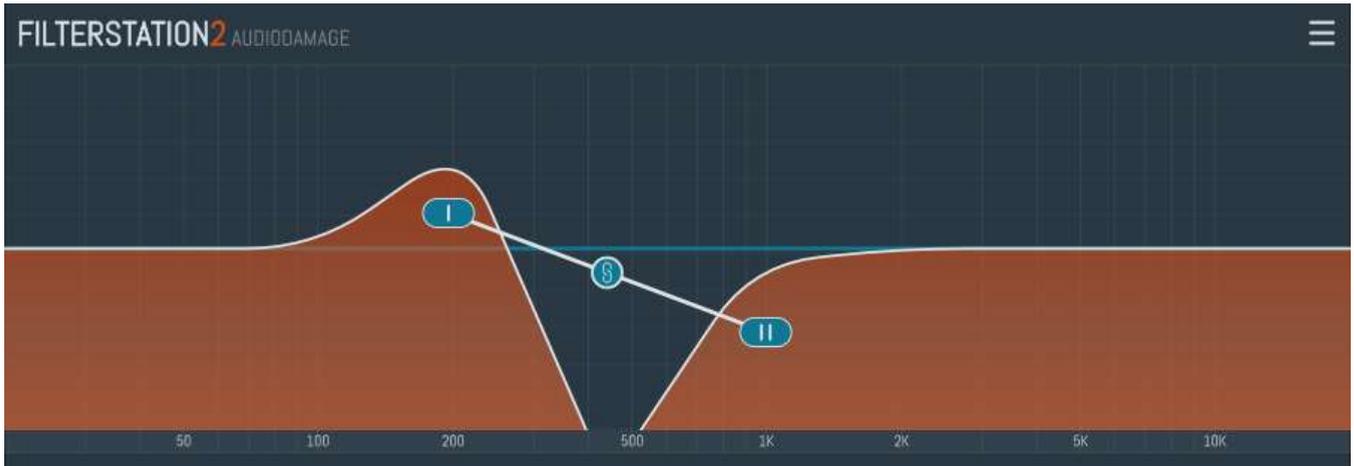
Filters One and Two

Filterstation's two filters are represented graphically in the upper half of the plug-in's window. The window shows the frequency responses of the two filters, with frequency running horizontally (lower frequencies on the left) and amplitude running vertically. The orange regions represent the filters. A scrolling waveform display of Filterstation's output signal appears in the background of this area.

Each filter has a blue oval which responds to mouse clicks and dragging. To change a filter's frequency, drag the oval horizontally. To change the filter's resonance, drag the filter vertically. In the following screenshot, Filter 1 is a low-pass filter with a frequency setting of about 200Hz and a medium-high resonance, and Filter 2 is a high-pass filter with a frequency of about 2KHz and a low resonance setting. Note that these

² Why the term "poles"? It has to do with the mathematics used to design filters, and it's something of a long story. See for example http://en.wikipedia.org/wiki/Pole-zero_plot if you're curious.

representations are abstracted renditions; while they show the filter's type and frequency, they do not reflect the exact frequency response of Filterstation's different filters or their idiosyncrasies.

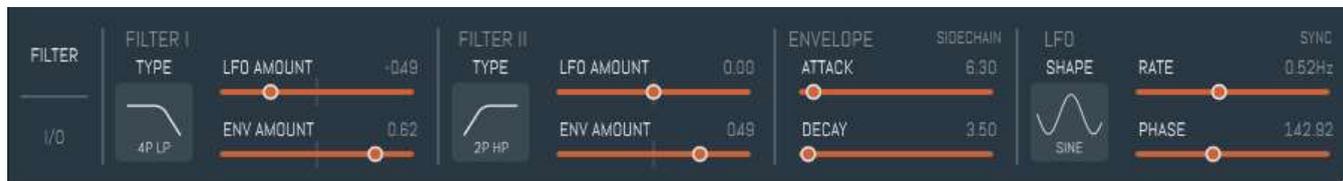


The blue circle with the linked-chain symbol lets you control both filters simultaneously. Dragging it back and forth will change the frequencies of both filters, while dragging it up and down changes their resonance settings.

As with all of Filterstation's parameters, the positions of these controls generate, and will respond to, host automation information. Hence you can record your dragging actions with your host, and/or control the filters with a hardware controller.

Filter Controls

The lower section of Filterstation's window contains two sets of controls. The first set includes controls for selecting the types of the two filters, and for the LFO and envelope follower which modulate the filters. It looks like this:



The second set contains controls for how signals are routed through the filters, input and output level controls, and additional signal-processing functions. We'll get to those later.

Filter Types

The two filters can be independently set to one of a dozen different types. To change the filter type, click the icon labeled **TYPE** and then click one of the icons which appears in the pop-up selector, shown below:



The icon labels use the conventions of "P" for Poles and "LP", "HP", and "BP" for Low-Pass, High-Pass, and Band-Pass. For example, the icon labeled 4P BP selects a four-pole band-pass filter. Here are descriptions of each of the filter types:

2P LP, 2P HP – plain-vanilla two-pole filters. If you’re looking for somewhat milder-sounding filters, these are your best bet.

3P LP – this filter started life as an emulation of the filter in the TB-303, and has some of its squelchy sound³.

4P LP – Audio Damage’s own 4-pole low-pass filter. Loosely based on the famous Moog filter, this is the one to use for overtly electronic-sounding filter effects.

4P HP – a high-pass filter with a steeper cutoff slope than its 2-pole counterpart. If you want to kill the bass in your sound, this will do the job.

NOTCH – that’s pretty much what it is. Notch filters can create phase-shifter-like effects, particularly if you use a couple of them with different frequency settings.

MS20 LP, MS20 HP – emulations of the low-pass and high-pass filters found in the Korg MS-20 analog synthesizer. These filters have more personality than the **2P LP** and **2P HP** modes.

914 BP – the bandpass filter from Audio Damage’s 914 filter bank plug-in. An accurate recreation of the filters in Moog filter bank modules, this is a bandpass filter with distinctive phase response characteristics.

4P BP – a band-pass filter with less resonance and coloration than the **914 BP**.

FILTERPOD – the filter from one of Audio Damage’s first products, Filterpod. It’s essentially a low-pass filter with an odd hump near its cutoff frequency, and some unusual resonance and overdrive characteristics.

VCA – not really a filter as such. Like the module commonly found in synthesizers, the VCA mode changes the loudness of signals, rather than changing their timbre. Changing the filter’s frequency parameter

³ It’s a common misconception in synthesizer folklore that the TB-303 has a three-pole filter. This is incorrect; it has a four-pole filter, albeit a somewhat strange one. Back when we wrote the first version of Filterstation, we were young and impressionable and accepted this erroneous gospel without question. We’re older and possibly wiser now, but we still like the way this three-pole filter sounds, so it is preserved without alteration in this version.

changes the level of the signal. This means that Filterstation is quite capable of creating a variety of volume-based effects, such as tremolo, gating, and auto-panning.

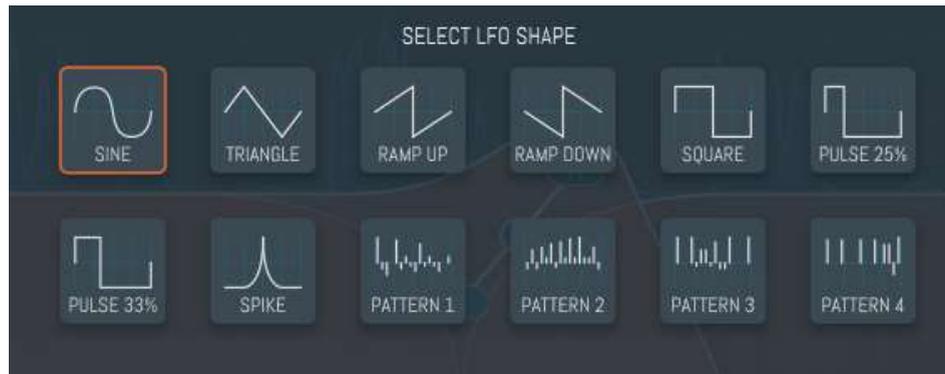
NONE - if you choose this mode, the filter is removed from the signal path altogether. This setting is useful if you need to focus on the sound of the other filter while adjusting it, or if you simply find that just one of Filterstation's filters is enough to do the job.

Low-Frequency Oscillator (LFO)

Filterstation has a single low-frequency oscillator (LFO) for controlling the filters. The LFO can modulate the frequency of either or both filters.

Shapes

The LFO has a selection of different wave shapes. Click the icon labeled **SHAPE** to invoke the shape selector, then click one of the shape icons as shown here:



The following shapes are available:

SINE - a sinusoidal wave which rises and falls smoothly, with no corners or sudden jumps.

TRIANGLE - a wave which moves up and down in straight lines, rising at the same speed that it falls.

RAMP UP - a wave which rises slowly from zero to its maximum value, then jumps back to zero.

RAMP DOWN – the opposite of **RAMP UP**: it jumps from zero to its maximum value, then returns to zero.

SQUARE – a wave which jumps between zero and its maximum value, staying at both values for the same amount of time.

33% PULSE – like the **SQUARE** wave, but it stays at the maximum value for 1/3 of its total time and at zero for 2/3 of the time.

25% PULSE – like the **SQUARE** wave, but it stays at the maximum value for 1/4 of its total time and at zero for 3/4 of the time.

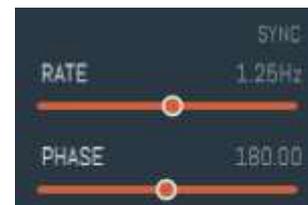


SPIKE – this wave rises quickly from zero to the maximum value, and then falls back to zero.

PATTERN 1-4 – these settings produce 16-step rhythmic patterns for creating burbling filter sequences, trance-gate patterns, etc.

LFO Amount

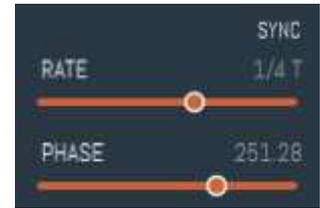
Each filter has its own **LFO AMOUNT** control, which sets how much the LFO changes the filter's frequency. These controls have a range of -1.00 to +1.00. Larger values, either positive or negative, have a greater effect on the filter. If you set the **LFO AMOUNT** control to a positive value, the LFO will increase the filter's frequency. (For low-pass filters, this is usually called "opening" the filter, since it allows more harmonic content to pass through.) If you set the **LFO AMOUNT** to a negative value, the LFO will decrease the filter's frequency. Although Filterstation has one LFO which is shared by the two filters, you can set their LFO amounts to opposite polarities and cause the filters to move in different directions.



Rate and Host Sync

The **RATE** control determines how fast Filterstation's low-frequency oscillators change the levels of the two audio channels, and thus the speed of the panning or tremolo effect. The **RATE** control operates either in time units—the number of cycles per second—or in metrical units, depending on the setting of the **SYNC** switch. If the **SYNC** switch is off, Filterstation's LFOs run at whatever rate you set with the **RATE** control. In this mode, the LFOs can run as slowly as 0.01Hz (one cycle every 100 seconds) to 100Hz.

If the **SYNC** switch is on, Filterstation's LFOs synchronize to the tempo and position of your host DAW's transport. The **RATE** control operates in metrical units, expressed as fractions of a measure. For instance, if the **RATE** control is set to 1/4, the LFOs will complete four cycles every measure. The **RATE** control has a range of 1/32 to 2/1 (one cycle every two measures). Dotted and triplet values are indicated with a . (dot) and **T** respectively.



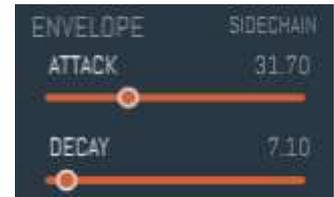
Turn sync on and off by simply clicking the word **SYNC**.

Phase

The **PHASE** control sets the starting position of the LFO's wave form, i.e. its phase, when it begins a cycle at the beginning of a measure. Phase is expressed in degrees and ranges from 0 to 360. The exact effect of the **PHASE** control varies depending on which **LFO SHAPE** you're using. If **SYNC** is off, the **PHASE** control has no audible effect.

Envelope Follower

Filterstation can respond to the loudness of the incoming signal with its envelope follower. The envelope follower generates a modulation signal that rises and falls in proportion to the loudness of the input signal, and can be used—like the LFO—to control the frequencies of either of the filters, or both.



Attack and Decay

The envelope follower has two parameters for adjusting how it responds to the signal: **ATTACK** and **DECAY**. **ATTACK** controls how quickly the envelope follower's output increases when the level of the input signal increases. **DECAY** does the opposite, controlling how fast the output decreases when the input signal's level falls. If you're using the envelope follower with a signal like drums or electric bass, you'll probably want to use an attack value near zero so that the envelope follower responds rapidly. Adjust the decay setting to smooth out or extend the response of the envelope follower as the notes die away. With sustained sounds like synthesizer pads, you can use a longer attack time to create fade-in effects that open the filters slowly as you hold chords. In conjunction with the VCA filter mode, a longer attack time can create slow-gear effects for guitar.

Envelope Amount

Each filter has its own **ENVELOPE AMOUNT** control, which sets how much the envelope follower changes the filter's frequency. These controls have a range of -1.00 to +1.00. Larger values, either positive or negative,

have a greater effect on the filter. If you set the **ENVELOPE AMOUNT** control to a positive value, the envelope follower will increase the filter's frequency when the input signal gets louder. If you set the **ENVELOPE AMOUNT** to a negative value, the envelope follower will decrease the filter's frequency.

Side-Chain Input

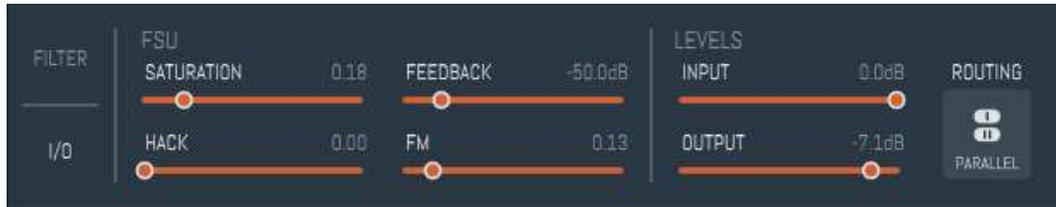
Filterstation has a pair of auxiliary inputs which will appear in your host as "SC". If the switch labeled **SIDECHAIN** is turned on, the signal from these inputs is fed to the envelope follower. This lets you use a different instrument to control the filters, similar to how a side-chain compressor lets you control the loudness of one instrument with another. For instance, you could use a kick drum to apply rhythmic filtering to a synthesizer, or use your voice to control the filter. In general, these are the steps for setting up side-chain control of Filterstation:

- 1) Insert Filterstation on the track with the instrument that will be filtered (e.g., the synthesizer in the first example we mentioned above).
- 2) Route a signal from the track with the instrument that will control the filter (e.g., the kick drum in the first example) to Filterstation's side-chain inputs. Usually you'll want to use a "send" control on that track, but you can also route the output of the track to Filterstation if you don't want to hear the controlling instrument in your mix.
- 3) Turn on the side-chain input in Filterstation by clicking the **SIDECHAIN** button.
- 4) For starters, set the **ATTACK** and **DECAY** sliders to zero.
- 5) Move the **ENVELOPE AMOUNT** slider(s) to the right or left to adjust how much the side-chain signal controls the filter(s).
- 6) Adjust the **ATTACK** and **DECAY** sliders to smooth out or lengthen the envelope follower's response.

Support for side-chain inputs varies depending on both the plug-in type and your host's signal-routing features. Side-chain routing is usually simpler to set up with VST3, AU, and AAX plug-ins. Side-chain routing isn't a fully supported feature in VST2 plug-ins. If you use the VST version of Filterstation, you may find that your host presents four inputs for the plug-in but does not label them. If this is the case, the third and fourth inputs are the side-chain inputs.

I/O Controls

Click the label **I/O** at the bottom left of Filterstation's windows to reveal another set of controls. to switch between the two sets of controls.



Routing modes

Filterstation has three different ways of routing audio through its two filters. These are chosen with the switches near the top of Filterstation's window. The three choices are:

SERIES – the signal passes through Filter 1 first and then through Filter 2.

PARALLEL – the signal is split and passes through Filter 1 and Filter 2 simultaneously, and the outputs of the two filters are added together.

STEREO – the left audio signal passes through Filter 1, and the right audio signal passes through Filter 2.

Filterstation uses separate filters for the left and right channels, so a stereo signal path is present regardless of the routing configuration. Click on the icon labeled **ROUTING**, then click one of the icons in the selector to choose the corresponding routing configuration.

Note that it's not difficult to create configurations in which no sound at all comes out of Filterstation. Suppose, for example, you use the Series routing and band-pass types for both Filter 1 and Filter 2. If the frequency settings of the two filters are far enough apart, nothing will come out of Filter 2: only a narrow range of frequencies will be emitted by Filter 1, and Filter 2 will block them. However, if you were to switch to the Parallel routing, you'd hear an interesting double-peak effect, since the raw input signal would enter both filters and their outputs would be mixed.



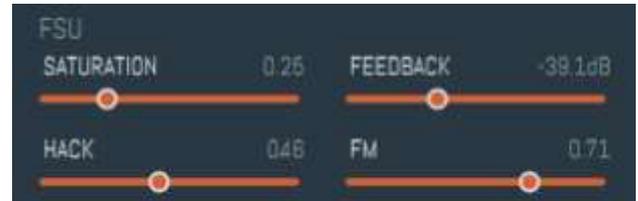
Filterstation uses separate filters for the left and right channels, so a stereo signal path is present regardless of the routing configuration.

FSU

This section contains controls which distort the signal in various ways. The effect of these controls is highly dependent on the signals entering Filterstation, the currently active filter types, and the settings of other controls. Hence the result of manipulating these controls is somewhat unpredictable. If you're using Filterstation because you need clean, subtle filtering effects, you probably won't find much use for these controls. On the other hand, if you're looking to mangle sounds, these controls are your new friends. The letters "FSU" might stand for Foul Stuff Up, or Forced Signal Unpleasantness, or, well, use your imagination.

Saturation

The Saturation slider applies a variable amount of analog-like distortion to the signal before it enters the filter. The slider has no effect at its leftmost position. As you move it to the right, the signal will become increasingly distorted.



Feedback

The Feedback slider routes some of Filterstation's output signal back into its input. The further right you move the slider, the more signal is fed back. This may cause the entire plug-in to oscillate loudly, so exercise caution.

Hack

The Hack control damages audio with noise whose character is dependent on both the input signal's loudness and the frequency setting of Filter 1. At settings near the left end of the scale, you'll hear Hack only on the louder peaks of the input signal. Moving the Hack slider to the right increases the amount of damage.

FM

FM stands for Frequency Modulation. The FM slider changes (modulates) the filter frequencies with the input audio signal itself. This is different than using the envelope follower for modulation in that while the envelope follower responds to the amplitude level of the signal, the FM control responds instantaneously to the shape of the signal. In other words, while the envelope follower changes the filters in response to the loudness of the signal, the FM control changes the filters in response to the frequency of the signal.

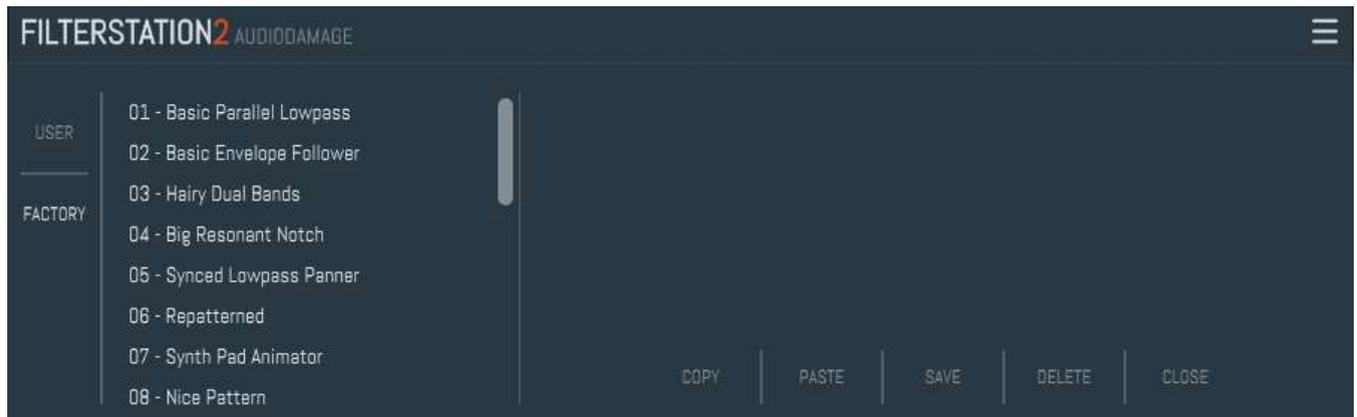
Moving the slider to the right increases the amount of FM. Small amounts of FM can create a subtle fluttering or produce sounds like tremolo effects. More FM can make sounds fizzy or flatulent.

Levels

Finally, Filterstation has a pair of sliders for controlling the level or amplitude of the signals entering and leaving the plug-in. In most cases you can probably leave these sliders at their rightmost position, which provides unity gain. If you find that the output of Filterstation is a bit loud, use the **OUTPUT** slider to reduce it. On the other hand, use the **INPUT** slider to lower the level of the signal entering the plug-in. Since the filters and FSU controls respond differently depending on the level of the signal passing through them, you might find that automating the **INPUT** slider produces interesting effects.

Presets

Filterstation includes a number of built-in presets to serve as a demonstration of its capabilities and inspirations for your own creations. To access the presets, click the menu icon at the top-right corner of the window to invoke the preset browser. The scrolling list on the left shows the available presets; click one of the preset names to load its settings.



To save your own presets, click the **SAVE** button. You can save your presets anywhere you like, but if you place these files in Filterstation's own folder (located at C:\ProgramData\Audio Damage\Filterstation\Presets\ on Windows, /Library/Application Support/Audio Damage/Filterstation/Presets/ on OS X), your presets will appear in the scrolling list on the left when you click the **USER** button.

You can delete presets from the User list by clicking their name and then clicking the **DELETE** button.

The **COPY** and **PASTE** buttons copy the current settings to the system clipboard and paste settings from the clipboard. The settings are presented in a plain-text XML format so that you can exchange them online in forums, copy them between a Windows computer and a Macintosh, etc.

The **CLOSE** button closes the preset browser.

Automation

All of Filterstation's parameters can be automated using your host's automation features. Consult your host's documentation for information on how to use these features.

And Finally...

Thanks for purchasing Filterstation. We make every effort to ensure your satisfaction with our products, and want you to be happy with your purchase. Please write info@audiodamage.com if you have any questions or comments.