M12-Filter

ARTURIA®
YOUR EXPERIENCE • YOUR SOUND

Special Thanks

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Thank you for purchasing M12-Filter!

This manual covers the features and operation of Arturia's **M12-Filter**, the latest in a growing line of powerful plug-in effects by Arturia.

Be sure to register your software as soon as possible! When you purchased M12-Filter you were sent a serial number and an unlock code by e-mail. These are required during the online registration process.

Special Messages

Specifications Subject to Change:

The information contained in this manual is believed to be correct at the time of printing. However, Arturia reserves the right to change or modify any of the specifications without notice or obligation to update the hardware that has been purchased.

IMPORTANT:

The software, when used in combination with an amplifier, headphones or speakers, may be able to produce sound levels that could cause permanent hearing loss. DO NOT operate for long periods of time at a high level or at a level that is uncomfortable.

If you encounter any hearing loss or ringing in the ears, you should consult an audiologist.

Introduction

Congratulations on your purchase of Arturia's M12-Filter!

We'd like to thank you for purchasing M12-Filter, our latest filter modeling plug-in effect. It is based on the astounding filter section of the Oberheim Matrix-12, a massive analog synthesizer from the 1980s. The level of respect held by synth aficionados for the Matrix-12 ranks it among the most popular synthesizers of all time.

Arturia has a passion for excellence, and M12-Filter is no exception. We have not only emulated the sound and behavior of the original filter section; we have expanded its capabilities far beyond what the original could do. We are confident you will turn again and again to the M12-Filter when you are looking for ideas that are fresh and original.

Be sure to visit the www.arturia.com website for information about all of our other great hardware and software instruments. They have become indispensable, inspiring tools for musicians around the world.

Musically yours,

The Arturia team

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1. WELCOME TO M12-FILTER!

M12-Filter is the newest addition to our growing family of analog filter modeling plug-in effects. Not only does it emulate the performance of this legendary filter perfectly, it provides two of them! The filters may be run in parallel for stereo processing, or in series for even steeper filters and more resonant peaks. And since there are 15 filter types per filter, that means there are 225 potential combinations when running them in series! The possibilities are nearly endless.

M12-Filter runs as a plug-in in all major formats inside your DAW. It has easy MIDI learn functionality for hands-on control of most parameters, and also allows parameter automation for greater creative control.

1.1. Oberheim: an overview

A lot of brainstorming and hard work had to happen before musicians were able to get their hands on a polyphonic Oberheim synthesizer like the Matrix-12. The founder of the company, Tom Oberheim, got his start in the 1970s by designing a series of groundbreaking products like a ring modulator, a phase shifter, a sequencer, a monophonic synthesizer expansion module, and some drum machines.

But the reputation of the company grew by leaps and bounds as Oberheim introduced a family of synthesizers with an ever larger voice count and sound. The monophonic SEM module grew into a two-voice synthesizer with a keyboard (the TVS-1), doubled its polyphony and added more keys (the FVS-1), and then doubled its voice count again in the Oberheim 8-voice, which was released in 1977.

More synths followed hot on the heels of the 8-voice, and the designs became more sleek with each one: OB-1 (1978), OB-X (1979), OB-Xa (1980), OB-8 (1983), and the Xpander (1984).

Finally, Oberheim introduced their flagship, the Matrix-12, in 1985. It was based largely on the same voice architecture as its predecessor, the Xpander, with the addition of some important features.

1.2. Enter the Matrix

There is no argument that the pinnacle of the Oberheim synthesizer family was the Matrix-12. With a retail price of $5,000 it was only a dream for many musicians. But its impressive feature set provided a temptation that was difficult to resist: 12 dual-oscillator voices, 5 envelopes, 5 low-frequency oscillators (LFOs), a Lag processor (for portamento, etc.), 3 tracking generators, and 4 ramp generators (i.e., modulation delays), all of which added up to a lot of programming options. On top of that, the Matrix Modulation section allowed almost any imaginable sort of control over the final sound, from small refinements to radical routings that could completely disfigure the sound if desired.

But the one feature that was most responsible for shaping the sound of the Matrix-12 was its innovative filter section, which boasted 15 different filter modes ranging from subtle to extreme. And this amazing filter section makes up only one half of Arturia's M12-Filter plug-in: the other half is a second, identical filter section, complete with all 15 different filter modes. Finally, the Matrix-12 filter section has met its match: itself!

And once you are completely enamored with this astounding filter plug-in, you might want to check out Arturia's Matrix-12 V. It captures the powerful sound of the entire Matrix-12 synthesizer perfectly. And as usual with Arturia's software instruments, Matrix-12 V provides even more features and programming flexibility than the original device.

1.3. Arturia's secret ingredient: TAE®

TAE® (True Analog Emulation) is Arturia's outstanding technology dedicated to the digital reproduction of the analog circuits used in vintage synthesizers.

TAE®'s software algorithms result in spot-on emulation of analog hardware. This is why Matrix-12 V offers an unparalleled quality of sound, as do all of Arturia's virtual synthesizers.

TAE® combines three major advances in the domain of synthesis:

1.3.1. Aliasing-free oscillators

Standard digital synthesizers produce aliasing in high frequencies, especially when using Pulse Width Modulation (PWM) or Frequency Modulation (FM).

TAE® enables the generation of oscillators which are completely free of aliasing in all contexts (PWM, FM…), and at no extra CPU cost.
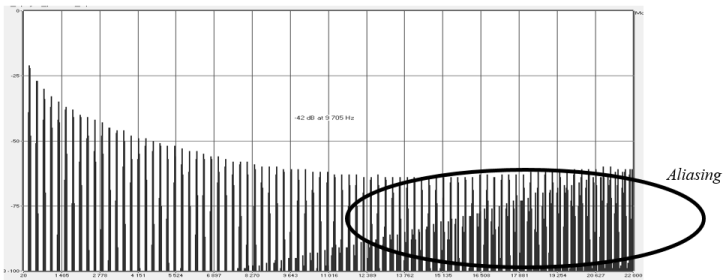
Linear frequency spectrum of a current well-known software synthesizer
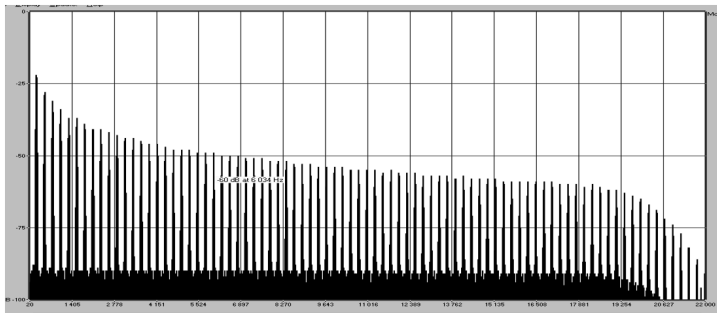
Linear frequency spectrum of an oscillator modeled with TAE®

1.3.2. A better reproduction of analog oscillator waveforms

The waveforms produced by the oscillators in analog synthesizers are affected by the presence of a capacitor in the circuits. The discharge of a capacitor results in a slight 'bend' in the original waveform (most notably for sawtooth, triangular and square waveforms). TAE® reproduces the result of this capacitor discharge in software.

Below is the analysis of a waveform from one of the five original instruments Arturia's software emulates, followed by one made by TAE®. They are both equally deformed by the low-pass and high-pass filtering.
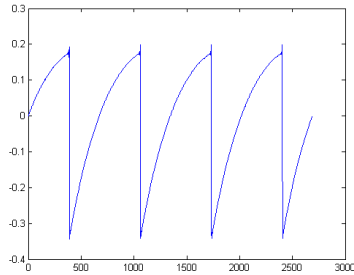
Temporal representation of the "sawtooth" waveform of a hardware synthesizer
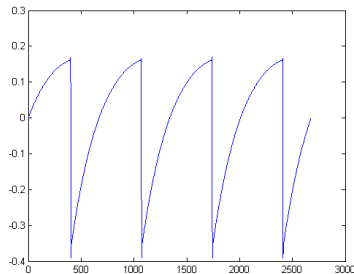
Temporal representation of a "sawtooth" waveform reproduced by TAE®

What's more, the hardware analog oscillators were unstable. In fact, their waveforms vary slightly from one period to another. If we add to this the fact that the starting point for each period (in Trigger mode) can vary with the temperature and other environmental conditions, we see why vintage synthesizers have such a typical sound.

TAE® reproduces the instability of oscillators, resulting in a fatter and "bigger" sound.

1.3.3. Direct Filter Circuit Modeling

Due to advances in computer processing power, TAE® can now employ direct filter modeling techniques to achieve unprecedented accuracy in the emulation of a hardware synthesizer's filter. By modeling the operation of the individual hardware components of the filter circuit, the warm nuances synonymous with analog sounds are recreated.

The following graph shows a single example of direct circuit modeling in action. The peaks represent the generation of harmonics at multiples of the resonant frequency when a particular filter is in self oscillation mode. These harmonics are characteristic of hardware synthesizer filters and are due to the non-linear behavior inherent to their analog circuitry. Anomalies such as these add to the richness and warmth of the sound produced by the filter.

But you'll notice there are two lines on the graph: Those are the superimposed frequency domain plots for both one of Arturia's virtual instruments and the hardware filter being emulated. They are practically indistinguishable, both on the graph and to the human ear. The direct recreation of this analog circuitry causes the same characteristics of the sound to be present, thus giving the user a truly analog sound.
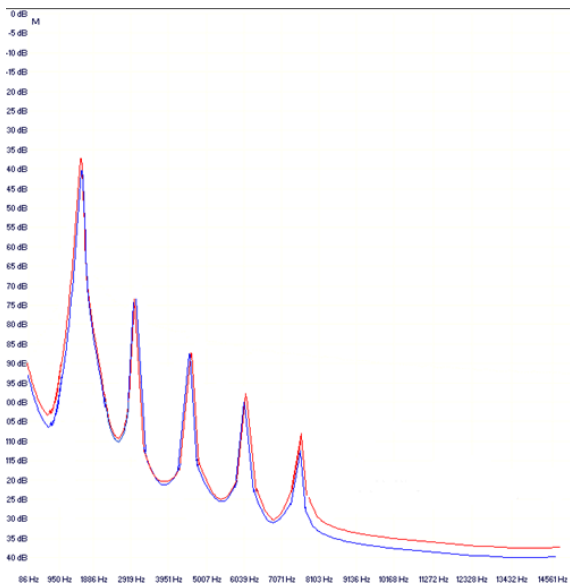
Comparison of harmonics generated by the filter circuits
in self-oscillation of TAE® and a hardware synthesizer

So here's the bottom line: when you bring together a bunch of music lovers who also have a deep understanding of the characteristics of electronic circuits, you wind up with Arturia. And Arturia now offers you our most impressive analog filter model yet, the M12-Filter.

We take great satisfaction in knowing this great filter plug-in will help you explore previously unknown musical territory.

1.4. Features of M12-Filter

- Two multi-mode filters, configurable in series or parallel
- Fifteen resonant filter modes per filter
- 5x8 Modulation Matrix
- Three Envelope Generators, which can be used as LFOs, envelopes, or step sequencers via the Modulation Matrix
- Envelopes can be looped and are always synchronized to the master clock
- Envelopes may be triggered and retriggered by incoming audio
- Random source generator, patchable through the Modulation Matrix
- Modulation oscillator with a range from 0.01 Hz – 10 kHz, patchable through the Modulation Matrix
- MIDI-assignable parameter control
- Factory presets and Envelope templates, with the ability to create and save your own
- Preset browser with search & filter options

We're very proud of the M12-Filter, and are certain your creative process will be unlocked in new and exciting ways as you explore its many features.

2. ACTIVATION & FIRST START

M12-Filter works on computers equipped with Windows 7 or later and macOS 10.10 or later. You can use M12-Filter as an Audio Units, AAX, VST2 or VST3 plug-in.

2.1. Activate the M12-Filter license

Once M12-Filter has been installed, the next step is to activate your license for the software.

This is a simple process that involves a different software program: the Arturia Software Center.

2.1.1. The Arturia Software Center (ASC)

If you have not already installed the ASC, please go to this web page:

Arturia Updates & Manuals

Look for the Arturia Software Center at the top of the page, and then download the version of the installer that you need for your system (macOS or Windows).

Follow the installation instructions and then:

- Launch the Arturia Software Center (ASC)
- Log into your Arturia account
- Scroll down to the My Products section of the ASC
- Click the Activate button

That's all there is to it!

2.2. M12-Filter and your DAW

M12-Filter comes in VST, AU and AAX plug-in formats for use in all major DAW software such as Cubase, Logic, Pro Tools and so on. You can load it as a plug-in effect and its interface and settings will work right away:

- Time-based features will synchronize to your DAW's host tempo/bpm rate
- You can automate nearly every parameter using your DAW's automation system
- You can use more than one instance of M12-Filter in a DAW project
- Any additional audio effects your DAW has available may be used to process the M12-Filter's audio output, and vice versa
- You can route M12-Filter's audio outputs freely inside your DAW using the DAW's own audio routing system.

♪: M12-Filter exists only as a plugin, not as a standalone instrument. Once it has been installed, open your favorite DAW and add it as a plugin effect to one or more of the tracks in a song.

2.3. Start using it!

The best introduction to M12-Filter is to load it up and start tweaking. You'll be able to get going pretty quickly on your own, because it's easy to get great results right away.

Here's how to get started. We'll use Logic Pro X for our examples, but the procedures are similar for most DAWs:

- Load a clip into an audio track in your DAW.
- Loop a section of the audio track. Two bars might be best at first, depending on the clip.
- Load an instance of M12-Filter as an insert on that same track.
- Make the interface of the M12-Filter visible by clicking on its name in the track window.
- In the Routing section of the M12-Filter window there are four buttons. Click the last one on the right (the one with A and B stacked on top of each other). This is the parallel routing configuration; it will allow you to hear both filters immediately.
- If the DAW isn't running, start the loop.
- Click and drag some knobs in the top three sections: Filter A, Master Cutoff, Filter B. You should hear results right away.
- For extra fun, click the Mode windows for Filters A and B. A pop-up menu will show 15 filter types. Make a selection and the window will close.

That's just the tip of the iceberg! As soon as you're ready to learn more about the features, proceed to the next chapter or jump to the one that piqued your curiosity: the User Interface [p.11], the Tool bars [p.15], the Modulation Matrix [p.39], the Filters [p.36], or the Modulators [p.43]. Click here [p.51] for an example of how to use the Envelope Generators to automate the modulation of M12-Filter parameters.

3. USER INTERFACE

M12-Filter is packed with great features, and in this manual we'll make sure you know what each one does. We think you'll be amazed by the power and versatility of this plug-in.

And while M12-Filter is very flexible, there's nothing complicated about it. That will always be the main focus of every Arturia product: to maximize your creativity while remaining easy to use.

3.1. Tool bar windows

Immediately above and below the Filter effects window are two very useful strips known as tool bars.

3.1.1. Upper tool bar
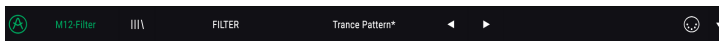
The Upper tool bar holds buttons and menus related to preset selection and library management, MIDI mapping, and other utility features.

See the Tool bars [p.15] chapter for specific descriptions of each feature.

3.1.2. Lower tool bar
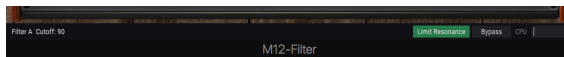
The Lower tool bar has a master Bypass [p.24] button, a Limit Resonance [p.23] button, and a CPU meter [p.24]. It also displays the names and values of parameters as they are accessed.

See the Lower tool bar [p.23] section of the Tool bars [p.15] chapter for specific descriptions of each feature.

3.2. The Filter Effects window

All of the audio processing features of the M12-Filter are contained within this window. Some features have pop-up menus that allow you to do things like choose the filter modes, select Envelope Generator presets, route sources and destinations in the Modulation Matrix, and more.

The Filter Effects window

1. Output section [p.12]
2. Filters A & B [p.13]
3. Modulators [p.13]
4. Modulation Matrix [p.14]

3.2.1. The Output section

Between the two Filter windows are three knobs and four buttons that control the overall output level and sound of the M12-Filter.

3.2.1.1. Master Cutoff

The central Master Cutoff knob changes the cutoff frequencies of both filters at the same time. This is a bipolar control with a range from -100% to + 100%. The center value of 0 has no effect on either filter.

3.2.1.2. Dry / Wet

This knob determines the balance between the processed and unprocessed audio signals. A setting of 0% is completely unprocessed (dry), which means no filtering is taking place. A setting of 100% means the audio source is only being heard through the combined filter settings.

If you want to hear the dry signal for a moment and then toggle the filter back in, there's a MIDI-assignable Bypass button [p.24] in the Lower tool bar [p.23] that will remove the M12-Filter completely from the audio chain.

3.2.1.3. Out Volume

The Out Volume is the master output control for M12-Filter. Click and drag the knob to make a change. This is a bipolar control with a range of -120 dB to +24 dB. Double-click the knob to reset the knob to the center value of 0 dB.

3.2.1.4. Routing

The M12-Filter features two instances of the Matrix-12 V filters. Audio signals may be routed through these filters in four different ways:

- Filter A only
- Filter B only
- Filter A –> Filter B –> Output: The filters are routed in series; the output of Filter A is routed through Filter B
- Filter A –> Output, Filter B –> Output: The filters are routed in parallel, and will process the incoming signal without affecting each other. If the input signal is mono it is duplicated and run through each filter to generate a stereo signal.

> ♪: When the filters are routed in series the output of Filter A is heard through Filter B.

3.2.2. Filters A & B

M12-Filter offers two identical filter sections, each of which is a complete model of the original Matrix-12 filter section. The parameters for each section are the same:

- Mode menu: select one of 15 filter modes
- Cutoff: adjust the corner frequency of the filter
- Resonance: adjust the emphasis level at the corner frequency
- Pan: modify the stereo balance of the filter
- Out: control the output level of the filter

See the Filters [p.36] chapter for more details about each control.

3.2.3. The Modulator sections

The Modulator sections include the three Envelope Generators [p.43], the Random source [p.54], and the Modulation Oscillator [p.55]. Each of these may be used to modulate one or more destinations when routed through the Modulation Matrix [p.39].

3.2.4. The Modulation Matrix

This is a virtual 5x8 patchbay that allows you to select one of five modulation sources and route it to as many as eight different destinations at one time. It is also possible for multiple sources to modulate the same destination.

All modulation routings are visible on the main page at the same time, so it's easy to see at a glance what is happening. For a full description, see the Modulation Matrix [p.39] chapter.

3.3. Parameter editing

As you work with the M12-Filter there are several methods you will use to alter parameters:

- Click and drag: Knobs, value fields, and points/slopes in an Envelope Generator
- Click to toggle: On/off/mute buttons
- Pop-up menus: Click the menu and it will open; make a selection from the menu
- Unipolar controls: Values range between zero and a positive number
- Bipolar controls: Values may be set positively or negatively. A value of O is neutral (no change).

> ♪: For finer value adjustments when using the click/drag method, hold the Ctrl key and then drag the cursor.

4. THE TOOL BARS

The Filter effect window is framed by the Upper and Lower tool bars. These provide an array of functions for preset management, MIDI mapping of parameters, and other features.

4.1. The Upper tool bar

The tool bar that runs along the top edge of the window provides access to many useful features. Let's look at them in detail.

The first group of options can be found by clicking on the M12-Filter section at the top left hand corner of the plug-in window.

We'll go through each of these functions in the following sections.

4.1.1. Save Preset

> ! : This option will overwrite the active preset with any changes you have made, so if you want to keep the source preset also, use the Save As option instead. See the next section for information about this.
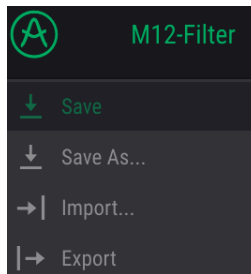
Save Preset

4.1.2. Save Preset As...

If you select this option you are presented with a window where you can enter information about the preset. In addition to naming it you can enter the Author name, select a Bank and Type, and even create your own Bank, Type, and Characteristics. This information can be read by the preset browser and is useful for searching the preset banks later.

You can also enter freeform text comments in the Comments field, which is handy for providing a more detailed description.
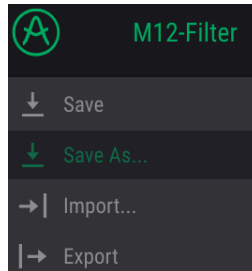
The Save As window

4.1.3. Import...

This command lets you import a file that was originally exported by M12-Filter. It can be either a single preset, an entire bank of presets, or a playlist. Presets and banks are stored in the **.mxfix** format, while playlists are given the extension **.playlist**.

After selecting this option the default path to these files will appear in the window, but you can navigate to whichever folder you prefer to use.
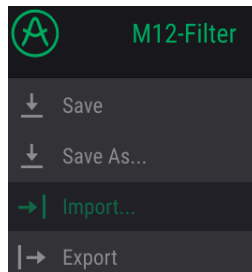
The Import Preset window

4.1.4. Export menu

The Export menu has several options for exporting files from M12-Filter, which enables you to share your sounds and playlists with other users. You could also use these options to transfer files to another computer.

4.1.4.1. Export Preset

You can export a single preset using this command. The default path to these files will appear in the window, but you can create a folder at another location if you like.
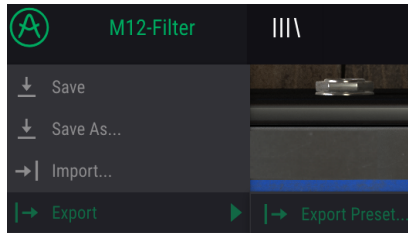
4.1.4.2. Export All Playlists

Playlists allow you to select which sounds to use for a particular gig or session. With this command you can export all of your playlists and import them into another computer that also has M12-Filter installed.
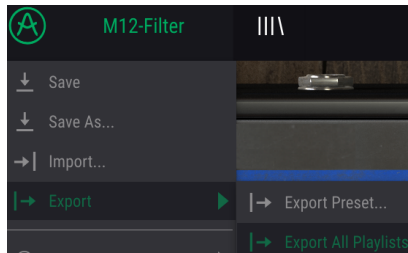
4.1.4.3. Export Bank

This option can be used to export an entire bank of sounds from the instrument, which is useful for backing up or sharing presets.
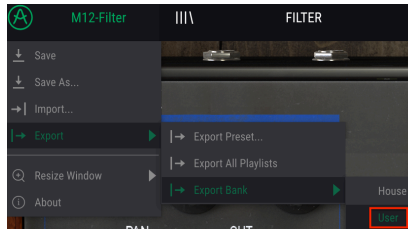
Selecting a Bank to export

4.1.5. Resize Window options

The M12-Filter window can be resized from 60% to 200% of its original size without any visual artifacts. On a smaller screen such as a laptop you may want to reduce the interface size so it doesn't dominate the display. On a larger screen or a second monitor you can increase the size to get a better view of the controls. The controls work the same at any zoom level but the smaller ones can be harder to see at the smaller magnification values.
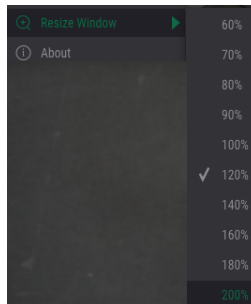
The Resize Window menu

4.1.6. About window

The final option in this drop-down menu is the About window. This is where the software version is located, along with other information about M12-Filter.

4.1.7. Preset browser overview

The Preset browser can be opened by clicking the library symbol in the tool bar. The Filter, name field and left / right arrows located in the upper tool bar can assist with preset selection.
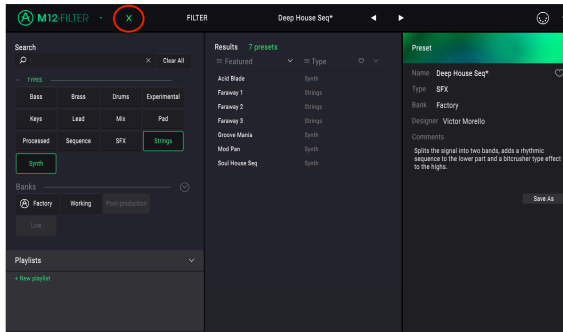
The Preset Browser

More details about this window are found in the Preset Browser [p.25] section.

4.1.8. MIDI Learn assignment

> ♪: Some DAWs do not allow MIDI control of a plug-in on an Audio track. In most cases the plug-in can still be used, but the settings must be adjusted manually.

The MIDI plug icon at the far right side of the tool bar places the instrument into MIDI learn mode. MIDI-assignable parameters will be shown in purple, which means you can map physical controls to those destinations inside the instrument. A typical example might be to map a physical expression pedal to the Out Volume control, or buttons on a controller to the Preset selection arrows so you can change the preset from your hardware keyboard.

MIDI Learn mode (top section)

In the image above one of the knobs is red. That means it has already been assigned to an external MIDI control. It can be reassigned [p.21], though.

Note that there are some MIDI-assignable parameters in the Modulation Matrix [p.39] and the Lower tool bar [p.23] also.
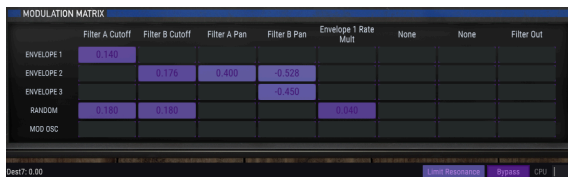
MIDI Learn mode (Modulation Matrix and lower tool bar)

> ♪: Modulation levels are also MIDI-assignable parameters. However, the boxes containing the levels will only turn purple or red when a non-zero value is present. Follow the link to learn more about the Modulation Matrix [p.39].

4.1.8.1. Assigning / unassigning controls

If you click on a purple area you'll put that control into learning mode. Move a physical knob, fader, or button and the target goes red, indicating that a link has been made between the hardware control and the software parameter. There's a popup window that displays which two things are being linked and an Unassign button that will disconnect the two.
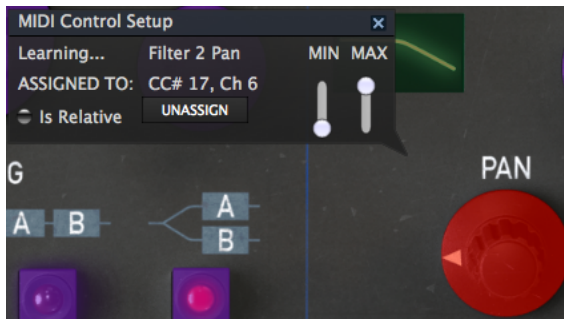
The Filter B Pan control selected and assigned

You can also right-click on a control to unassign it.

4.1.8.2. Min / Max value sliders

There are also minimum and maximum value sliders ("Min" and "Max") that you can use to restrict the range of the parameter response to something other than 0-100%. For example you might want the Out Volume knob to move within the range of 30-90%. If you made this setting (Min set to 0.30 and Max set to 0.90) your physical knob would be unable to alter the volume lower than 30% or higher than 90%, no matter how far you turned it. This is very useful for making sure you can't accidentally make the sound too quiet or too loud when performing.

In the case of switches which only have two positions (on or off), those would normally be assigned to buttons on your controller. But it is possible to toggle those with a fader or another control if you like.

4.1.8.3. Relative control option

The final option in this window is a check box labelled "Is Relative". It is intended for use with a specific type of control: one which sends only a few values to indicate the direction and speed at which a knob is turning, as opposed to sending a full range of values in a linear fashion (O-127, for example).
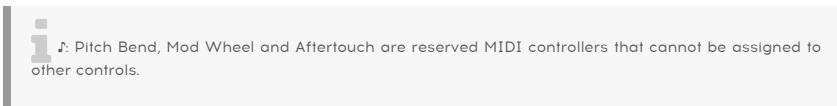
To be specific, a "relative" knob will send values 61-63 when turned in a negative direction and values 65-67 when turned in a positive direction. The turn speed determines the parameter response. Refer to the documentation of your hardware controller to see if it has this capability. If so, be sure to switch this parameter on when setting up its MIDI assignments.

When configured this way, movements of the physical control (usually a knob) will change the software parameter by starting at its current setting, rather than being an "absolute" control and snapping it to some other value as soon as you start to move it.

This can be a great feature when controlling things like volume, filter, or effect controls, since you won't usually want them to jump noticeably from their current setting when they are modified.

> ♪: Pitch Bend, Mod Wheel and Aftertouch are reserved MIDI controllers that cannot be assigned to other controls.

4.1.9. MIDI controller configuration

There's a small arrow at the far right hand side of the tool bar that opens the MIDI controller configurations menu. This allows you to manage the different sets of MIDI maps you may have set up for controlling the instrument's parameters from MIDI hardware. You can copy the current MIDI assignment setup or delete it, import a configuration file, or export the currently active one.

This is a quick way to set up different hardware MIDI keyboards or controllers with M12-Filter without having to build all the assignments from scratch each time you swap hardware.
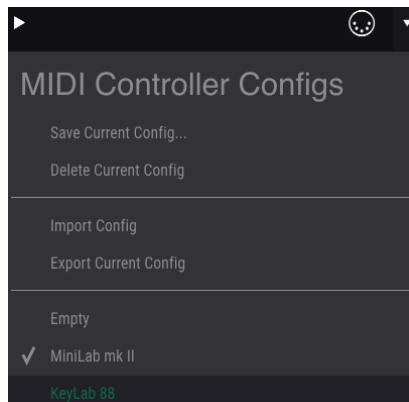
Note the check mark next to one of the controller names: that indicates that the MiniLab mk II configuration is currently active. Select the "Empty" configuration to start with a blank slate with all controllers unassigned.

4.2. The Lower tool bar

At the left hand side of the Lower tool bar you will see a readout showing the value or state of whatever control you are modifying. It will also display the current value of a parameter without editing it: just hover the cursor over the related control and the value will appear as pictured below.

Displaying the current control's value

4.2.1. Utility options

At the right hand side of the Lower tool bar are several small windows and buttons. These are very important features, so let's take a closer look at them.
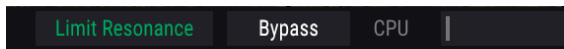
4.2.1.1. Limit Resonance button

Each of the filters in the M12-Filter is capable of self-oscillation. In simpler terms, when you set Resonance to maximum the filter will become an oscillator. By default, self-oscillaton is off. This option allows you to turn it on.

> ♪: The status of the Limit Resonance button is stored with the preset.

4.2.1.2. Bypass button

Activating the Bypass button will disable the M12-Filter. The difference between this button and the on/off button in the upper left corner is that the Bypass button is MIDI-assignable. Also, the status of the Bypass button is stored with the preset; the status of the upper on/off button is not.

> ♪: Remember that each of the filters within the M12-Filter plug-in has an independent On/Off button.

4.2.1.3. CPU meter

The CPU meter is used to monitor how much of your computer's CPU is being used by the plug-in.

4.3. The Preset browser

The preset browser is used to search, load and manage presets in M12-Filter. It has several different views but they all access the same banks of presets.

To access the search view, click the library symbol in the upper tool bar. To close it again, click the X that appears in the same location.

The Preset Browser button

4.3.1. Searching presets

The Search screen has a number of sections. By clicking on the Search field at the top left you can quickly enter any search term to filter the preset list by patch name. The Results column is updated to show the results of your search. Press the Clear All button in the Search field to clear the search.
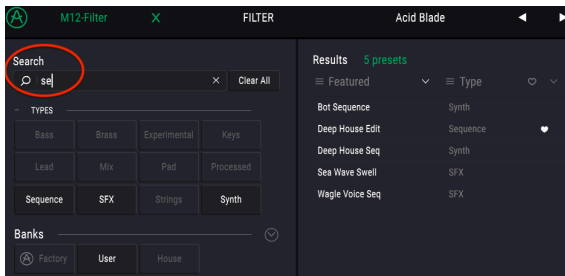
Filter by typing text in the Search field

In the example above the letters "s" and "e" were typed into the Search field. This selects all presets that have those two letters next to eachother in the preset name.

4.3.2. Using Type as a filter

You can also search using different Types. So for example by clicking on the Pad button in the Types field you can show only presets that match that Type.

Several types may be selected at once to make the search results even more specific. Both selecting and deselecting are done by using Ctrl + click (Windows) or Cmd + click (macOS).

You can use multiple search fields to perform narrower searches. So by entering a text search and also specifying the Type and Bank options you will see only the presets that match those exact criteria. Deselect any tag in any area to remove that criteria and widen the search without having to go back and start again.
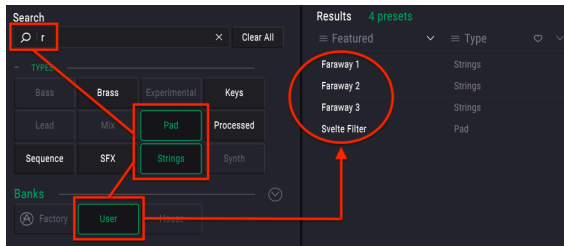
Search using multiple fields to filter the results

4.3.3. Tag category windows

The Tag category windows may be collapsed and expanded using the arrows near their names.
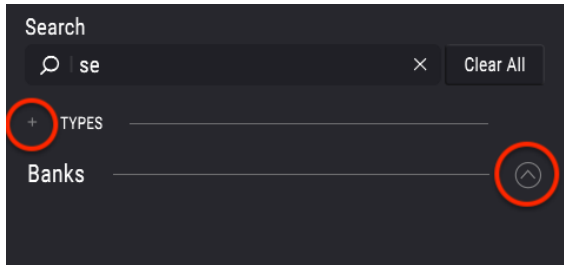
Tag category windows closed
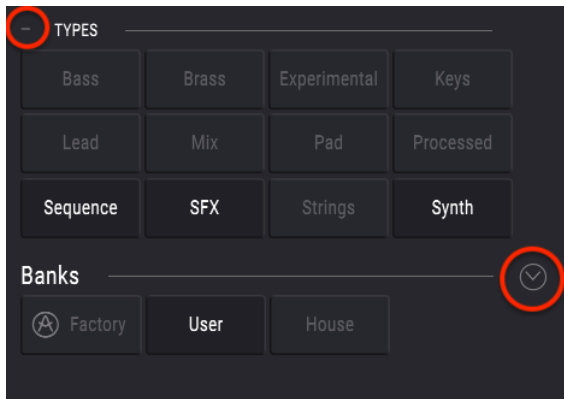
Tag category windows open

4.3.4. Search Results window

Click the options menu button in the first Results column to specify whether you want to view the presets by **Featured** or by **Name**. Click the sort arrow to reverse the alphabetical order.
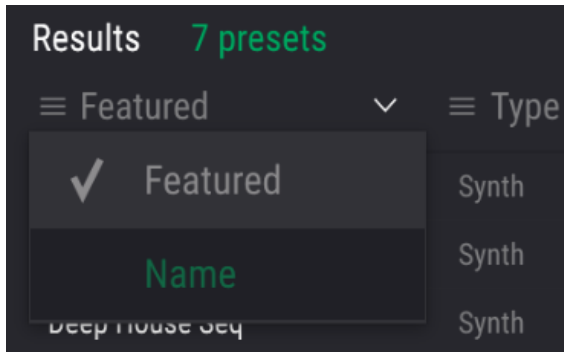
Sort options for column 1

Similarly, click the options menu button in the second Results column to order its display results by **Type**, **Sound Designer**, or **Bank** tags. Click the sort arrow to reverse the alphabetical order.
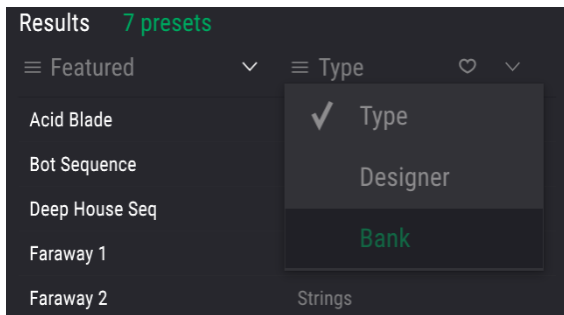
Sort options for column 2

4.3.5. The Preset Info section

The Info column on the right side of the search field shows specific information about each preset. The information for User presets may be changed here: Name, Type, Favorite, etc.

Click inside one of those fields and then type the new name or type, etc. You can also switch the Type or Bank by using a pull-down menu on the right side. Click Save when you are done.
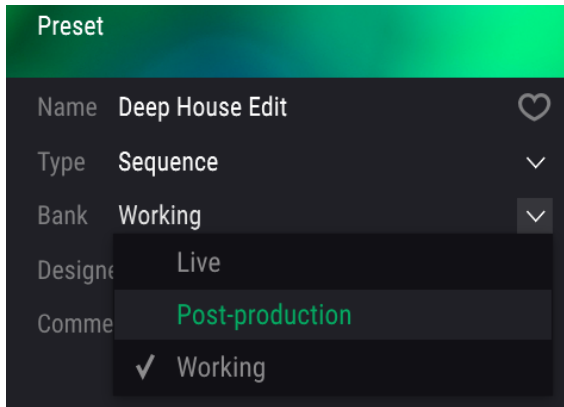
Select a new Bank for the preset

4.3.5.1. Edit the info for multiple presets at one time

It's possible to edit the Preset Info for multiple presets at the same time. For example, if you want to move 5 presets from the User bank to a different bank:

- Select those 5 presets inside the Search window
- Click the Edit Info button
- Use the Bank selection menu to select the target bank for those presets
- Click the Save Info button and those 5 presets will be moved to the new bank.

4.3.6. Delete a bank

ℹ !: The following process will delete both the Bank and all of the Presets it contains. If there are Presets you do not want to delete, edit the Bank selection inside the Preset information window before following these instructions.

The process of deleting a Bank is simple:

• Right-click on the Bank name
• A pop-up window will display the Delete Bank command and the name of the Bank
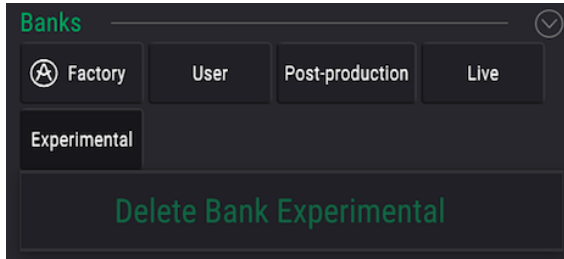
Deleting a Bank

• If the information is correct, click the pop-up window
• A second window will ask you to confirm the process and warn you of the dangers
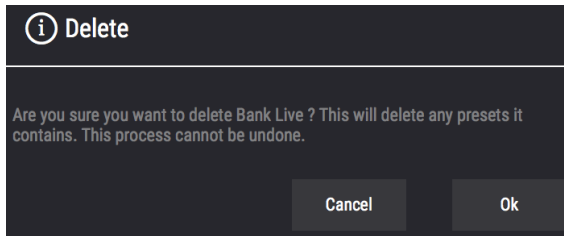
Deleting a Bank: The final confirmation window

• Click OK to delete the Bank and all of its Presets.

4.3.7. Preset selection: other methods

The pull-down menu to the right of the Search menu provides a different way to select presets. The first option in this menu is called Filter, and it will display the presets that fit the search terms you used in the Search field. So if you searched for the word **Form** in the main search area, the results of that search will appear here.
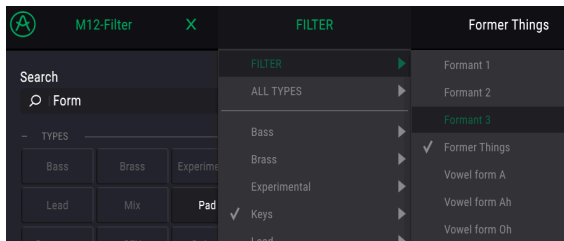
Filter results may differ based on Search criteria

Similarly, if you previously selected **Type: Pads** and **Bank: Post-Production** in the Search field you would see the results of that search in this area instead.
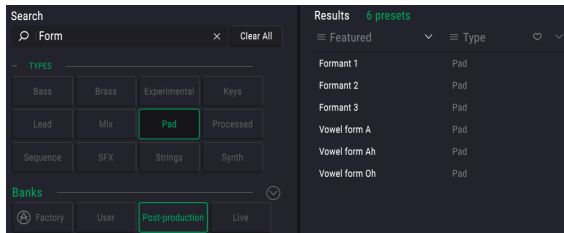
Filter results may differ based on Search criteria

Selecting the All Types option in the pull-down menu will bypass the Search criteria and show the entire list of presets.

The Categories below the line also ignore the Search criteria and display the presets based on their Type: *Bass, Brass, Keys,* and so on.
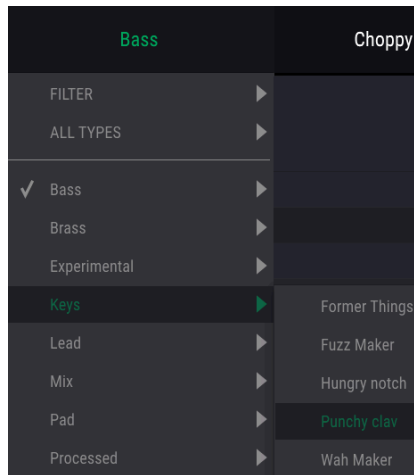
Selecting a preset by its Type

Clicking on the name field in the center of the tool bar will show you a list of all available presets. The list will also take into account any selections you have made in the Search field. So if you have pre-selected a Type such as "Funky" this shortcut menu will only show you presets that match that tag.

The left and right arrows in the tool bar cycle up and down through the preset list: either the full list, or the filtered list that resulted from the use of one or more search terms.

4.3.8. Playlists

In the lower left corner of the Preset Browser window is a feature titled Playlists. This is used to collect presets into different groups for different purposes, such as a set list for a particular performance or a batch of presets related to a particular studio project.

4.3.8.1. Add a playlist

To create a playlist, click the words "+ New playlist" at the bottom:
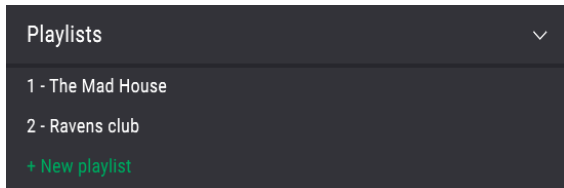
The Playlists menu

Give the playlist a name and it will appear in the Playlists menu. You can rename the playlist at any time; just click the pencil icon at the end of its row.

4.3.8.2. Add a preset

You can use all of the options in the Search window to locate the presets you want to have in your playlist. Once you have found the right preset, click and drag it onto the playlist name.
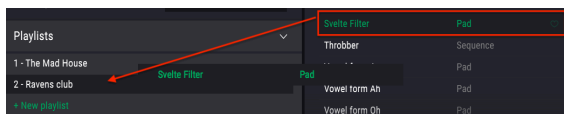
Click and drag from the Search Results list onto one of the playlists

To view the contents of a playlist, click on the playlist name.

4.3.8.3. Re-order the presets

Presets may be reorganized within a playlist. For example, to move a preset from slot 1 to slot 3, drag and drop the preset to the desired location.
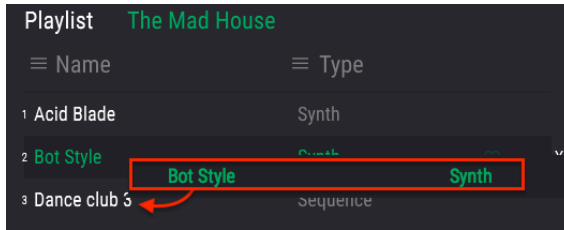
This will cause the other presets to be bumped up in the list to accommodate the new location of the preset being moved.
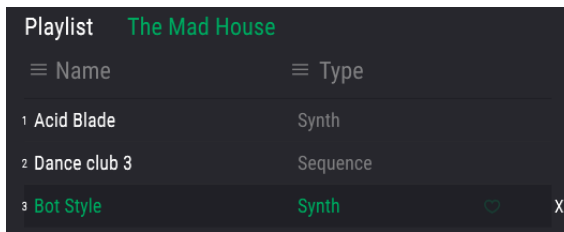
4.3.8.4. Remove a preset

To delete a preset from a playlist, click the x at the end of the preset row.
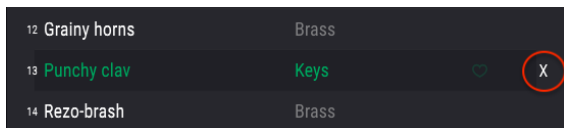
Click the X to remove a preset from a playlist

4.3.8.5. Delete a playlist

To delete an entire playlist, click the x at the end of the playlist row. This will only delete the playlist; it will not delete any of the presets inside the playlist.
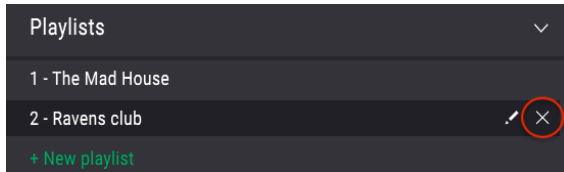
Click the X to delete a playlist

5. THE FILTERS

M12-Filter contains two independent filters, each of which is a perfect model of the original Matrix-12 filter section.

The secret to the spectacular sound of the Matrix-12 is that in addition to the more familiar Low Pass / High Pass / Band Pass filters in 2- or 4-pole options, it offers some very unusual filter types as well. There are 15 filter modes in all.

Having two of these amazing filter sections in a single plug-in makes the M12-Filter an unrivalled sound-sculpting tool. Let's take a look at each of the parameters.

5.1. Filter A / Filter B

Since the filter sections are identical, we will go through each parameter only once. The descriptions apply equally to each filter.

5.1.1. Filter Mode

Sometimes it helps to have a visualization of something that affects the audio realm. The following graphic is revealed when you click in the Filter Mode value field. It shows all 15 of the M12-Filter modes:
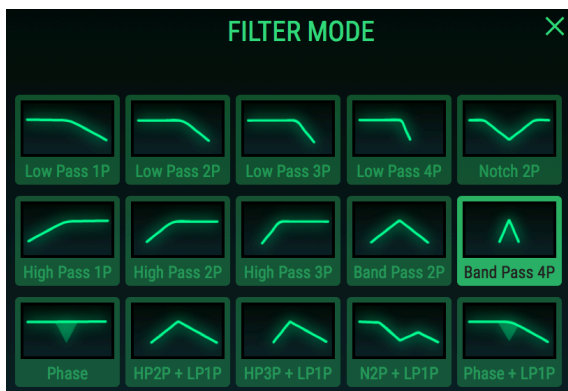
One of the M12-Filter mode selection windows, with Band Pass 4P selected

The graphic helps illustrate the concept that a 2-pole filter is "steeper" than a 1-pole filter of the same type, and a 3-pole filter is steeper than a 2-pole filter, and so on. The 4-pole LP filter, labeled "Low Pass 4P", rolls off the audio frequencies more quickly than the Low Pass 1P, for example. A High Pass filter scoops out the audio starting at the bottom, and the 3-pole HP filter has the strongest impact that way. But sometimes that isn't what you want, so we've given you two "kinder, gentler" HP filters in the 1-pole and 2-pole versions.

The other options affect the output signal in various ways, so be sure to try them all. And once you've become familiar with the different filter modes you'll probably be able to select one without referring to the built-in graphic. In that case, you might like to dedicate a couple of knobs on your controller to selecting the modes for each filter. Then you could simply turn the assigned knobs to dial up the filter modes you want. See the MIDI Learn assignment [p.20] section to learn more about this.

Select one of the options and the window will close, or click the X to close the window without making a change. You can also click anywhere outside the filter mode selection window and it will close.

5.1.2. Cutoff Frequency

This is probably the first knob anyone tries on a synthesizer: "How do the filters sound?" The amount and type of change you hear depends on a number of factors, but rest assured the M12-Filter provides just about anything you could want from an analog filter.

Turning the knob changes the location of the corner point(s) of the filter, revealing and/or hiding frequencies in the audio range depending on the selected filter model.

5.1.3. Resonance

To emphasize the corner point(s) at which the audio signal is being filtered, raise the resonance level. The higher it goes the more obvious a filter sweep becomes, to the point of "whistling" as certain frequencies are emphasized by the more powerful filter modes. This "whistling" is known as self-oscillation.

We should draw your attention the Limit Resonance [p.23] button at this point. When it is disabled, self-oscillation will happen at the maximum resonance values for most filter types. The status of the Limit Resonance button is stored with the preset, so it is possible to disable this feature selectively. As a general rule you should limit the resonance except for special presets. And if you do plan to experiment with this, please be careful not to destroy any audio components (like your ears!).

5.1.4. Pan

The further this knob is turned to the left or right, the more dry signal is heard on the opposite side. Full left means the right channel is completely dry (i.e., not filtered at all), and vice versa. This is a bipolar control with a range from -100 to +100.

5.1.5. Out

This bipolar knob controls the output level of the selected filter. The range is from -80 dB to +24 dB. Double-click the knob to return it to 0 dB (the center position).

> ♪: The highest levels of a filter's Out parameter can overload the output of the M12-Filter and cause distortion. This is especially true when the filters are routed in parallel. But when the filters are routed in series, the output of Filter A can be made to overdrive the input of Filter B without clipping the output of Filter B into your DAW. This can lead to some interesting results.

5.2. Output section

These master output controls [p.12] are described more completely in the User Interface [p.11] chapter. They are included here because two of them directly affect filters A & B.

- **Master Cutoff [p.12]**: This control enables you to offset the corner frequency of each filter at once.
- **Dry / Wet [p.13]**: This knob balances the effected and non-effected signal.
- **Out Volume [p.13]**: This is the overall output control for the plug-in.
- **Routing [p.13]**: Filters can be "soloed" or routed in series or in parallel by making a selection here.

> ♪: When the filters are routed in series the output of Filter A is heard through Filter B.

5.2.1. More about the Routing section

One of the things the Routing buttons allow you to do is bypass the filters individually so you can gauge the effect each one is having on the sound. For example, to bypass Filter A you would click the Filter B button. It may help to think of those buttons as "Solo" buttons for the filters.

There's also a MIDI-assignable Bypass button [p.24] in the Lower tool bar [p.23] that will remove the M12-Filter completely from the audio chain.

6. MODULATION MATRIX

A modulation matrix is a virtual "patch bay" that allows you to route one or more sources to one or more destinations. M12-Filter offers up to 40 modulation routings in each preset, instantly visible within a 5x8 grid.

The Modulation Matrix with example routings

The Modulation Matrix allows up to 5 different sources to be routed to a single destination, or as many as 8 different destinations to be modulated by a single source.

The mod sources are listed along the left side of the Modulation Matrix window:

Mod sources are listed vertically

The mod destinations are listed along the top of the Modulation Matrix window:

Mod destinations are listed horizontally

♪: The screen shot above shows one example of destination routings. There are many possibilities.

6.1. Sources

The modulation sources are "hard-wired" and cannot be changed. Only the mod destination and level can be altered. To use a particular source in a mod routing, first you'll need to find an empty box in its row.

The modulation destinations for Envelope 1 are in its row

For example, to set up a modulation routing between Envelope 1 and the master Out Volume parameter, you must first select Out Volume as the destination in one of the columns. For this example we will use the first row, first column of the Modulation Matrix.

*Envelope 1 will modulate
the Out Volume*

> ♪: For this example we must use the first row because it is hard-wired to Envelope 1, but we could have used any one of the columns to select the destination.

Once the mod destination has been chosen, click and hold the rectangle in the matrix that represents the connection between the source and the destination. As you drag the cursor up and down the mod routing value will be adjusted. For finer adjustments, hold the Control key and then click/drag the cursor in the value field.

In the example below, the Random source is controlling three different destinations.

A source can be routed to multiple destinations

In this example, as new values are generated by the Random source the cutoff frequencies for Filters A and B will be sent in opposite directions. This is because one mod routing is positive and the other negative. At the same time the Random source will modulate a different Envelope (Envelope 2), which itself has been routed to the output volume of Filter B.

6.2. Destinations

Click on one of the mod destinations along the top of the Modulation Matrix and a selection window will open.
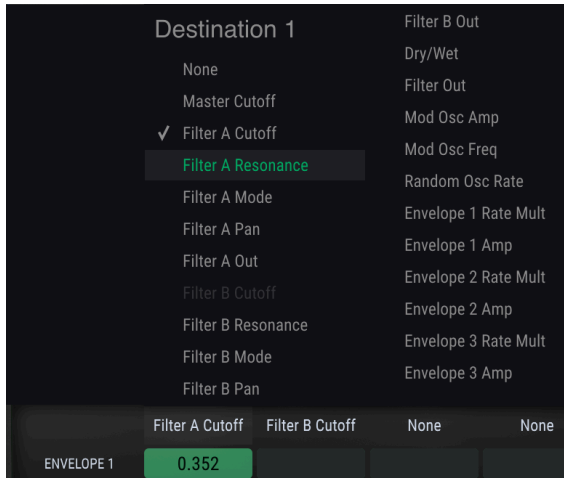
The mod destination selection menu

Note that Filter B Cutoff is not available in the list above: this is because it is already selected as a destination for another mod routing. A check mark indicates the current selection. Make a selection and the window will close.

Once the mod routing is complete, click and hold the rectangle in the matrix that represents the connection between the source and the destination. As you drag the cursor up and down the mod routing value will be adjusted. For finer adjustments, hold the Control key and then click/drag the cursor in the value field.

In the next example the Rate Multiplier of Envelope 1 is being controlled by three different sources.
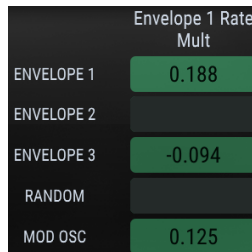
A destination can be modulated by multiple sources

In the example above Envelope 3 will modulate the Multiplier Rate of Envelope 1 negatively and the Modulation Oscillator will increase its speed. Keep in mind that Envelopes and the Modulation Oscillator are bipolar, so depending on their settings it is possible they will alternate between increasing and decreasing the Multiplier Rate. We'll cover those concepts in the Modulators [p.43] chapter.

6.3. Multiple sources/destinations: keeping track

The Modulation Matrix is a grid that indicates the connections between the sources and the destinations; i.e., it shows what the sources are modulating, and also displays which destinations are being modulated by certain sources.

It can be confusing at first glance to figure out what is connected to what, though. It may be helpful at first to follow the path starting with the source, using one hand to trace from left to right, and using the other hand to trace from top to bottom, until the two hands connect in the middle of the grid.

This diagram may also be helpful:
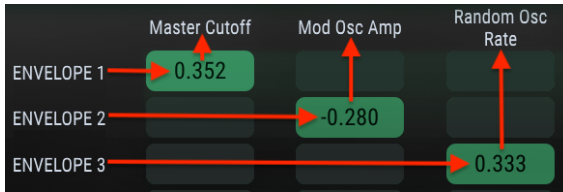
Mod routing connection diagram

In the previous picture Envelope 1 is controlling the Master Cutoff, Envelope 2 is affecting the Modulation Oscillator amplitude, and Envelope 3 is modulating the rate of the Random source.

> ♪: Once a destination has been connected to a particular source it is no longer available as a second destination for that source, and will be greyed out in the destination menu.

7.1. Envelope Generators

Three Envelope Generators are available as modulation sources. The Envelopes are sort of a mixture of an LFO, a step sequencer, and an envelope, and are capable of any of those behaviors depending on the settings you select.

Envelopes may contain up to 16 points, with adjustable levels, times, and slopes for almost every point. A Envelope may be looped continuously or triggered/re-triggered by incoming audio.

> ♪: A Envelope Generator must be routed to a destination through the Modulation Matrix in order for you to hear what it is doing.

7.1.1. Edit the Envelope

The round circles are called "points". They may be clicked and dragged to make changes. For more precise edits, click and drag the appropriate value field on the right side.

7.1.1.1. Point selection

Click a point to select it, or click the Point number field on the right side and drag it up and down to select the point to be edited. To add another point, click anywhere in the Envelope field. Up to 16 points are possible, but the first and last points are basically the same point, so they will always be the same level.

Right-click a point to remove it. If you double-click a point it will be reset to a value of 0 (the center line).

7.1.1.2. Adjust the Time

This parameter sets the length of time it will take to reach the selected point from the previous point in the Envelope. Changing this value will also affect the length of time it will take to reach the next point in the Envelope. Keep in mind that the actual duration of the time divisions is also dependent on the Rate Multiplier [p.49] setting.

When moving a point within a Envelope, or changing the value in the Time field on the right, the position of other points is not altered. For example, when moving point 4 you will adjust its position between points 3 and 5, but points 3 and 5 will stay where they are.

> ♪: A looping Envelope is always synced to the master clock, so only positions that fall within the grid resolution are permitted. If you click/drag a point to somewhere in the middle between two positions on the grid, that point will be moved left or right to the nearest grid position.

Double-click the numeric Time field to reset it to the default value.

7.1.1.3. Horizontal zoom

The Envelope Generator window can be zoomed in and back out. This enables you to view the entire Envelope or zoom in to view precise details within it.

To do this, click and hold the top line between the Envelope 1/2/3 tabs and Envelope Generator editing window. The cursor will become a "hand" icon as long as you hold down the cursor button.
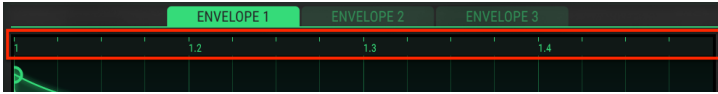
Click / drag within the timeline to zoom in / out

Now drag the hand icon up and down. You will see the timeline expand and reveal additional time division markers as you zoom in, and hide those markers when you zoom out.

At the maximum zoom level the FG editing window will span approximately 1/16th of the Envelope. To view a section of the FG window further to the left or right of the current view, grab the top line and drag it in the desired direction. The only time the left/right dragging will not work is when the FG window is zoomed all the way out.

> ♪: The depth of the FG view window cannot be changed; the vertical values are fixed.

7.1.1.4. Levels

The relative strength of the selected point is set using this parameter. The graphic will change as the level is moved. Hold the Control key while you click/drag for finer adjustment of the Level parameter.

Double-click the numeric Level field to reset it to the default value.

> ♪: Since the Envelope Generators will loop by default, point 1 and point 16 will always share the same level value.

7.1.1.5. Slopes

Each point has an adjustable slope that governs the shape of the transition into the next point. Positive values are increasingly exponential; negative values are increasingly logarithmic.

Click and drag the small arrows to change the slope:

The slope adjustment arrows

Right angle slopes are possible too, at the extreme of either value. These result in instantaneous change between two levels.

Double-click the slope within the window or its value in the numeric Slope field to reset it to a linear slope.

> ♪: A slope value is not available for the last point of a Envelope. Since the last point of a Envelope is also the first point, there is no transition between them and therefore no slope.

7.1.2. Envelope tabs

When a different Envelope tab is selected the Envelope Generator window will change to reveal the settings for that Envelope. All settings within the window are independent for each Envelope, including the Rate Multiplier and Trig Mode.

Use their tabs to select one of the three Envelopes

7.1.3. Envelope Generator Presets

A quick way to learn about the power of the Envelope Generators (FG) is to experiment with the preset templates included with M12-Filter. These presets are also great starting points for creating your own Envelopes.

> ♪: Changes you make to the FG settings will be stored when you save your preset. But there is an independent Save process for the Envelopes, and it's a good idea to use that also for two reasons:
>
> 1. If another FG preset is selected and then the Filter preset is saved, the original, unsaved FG settings will be lost.
> 2. If you want to use the unsaved FG settings with another Filter preset you won't be able to find them in the FG preset list unless the FG settings are deliberately saved as a Envelope Preset.

The presets and Save / Save As options are located in a pop-up menu on the left side of the Envelope Generator window. Click the list and the menu will open.
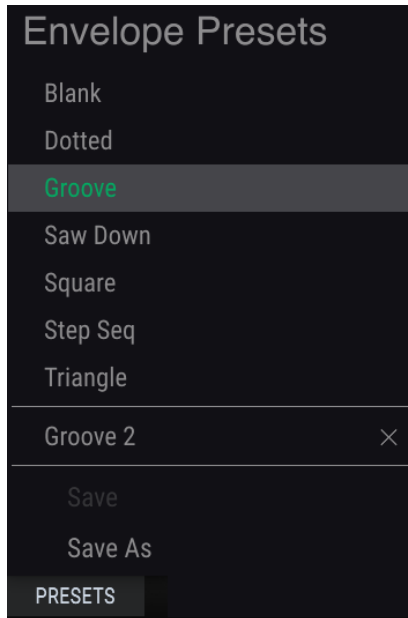
Envelope preset template selection menu

Scroll through the options and click a preset to select it. Once the selection has been made the menu will close.

7.1.3.1. Simple presets

Some of the presets are very simple, and would work well as an LFO waveform when used as a source in the Modulation Matrix.
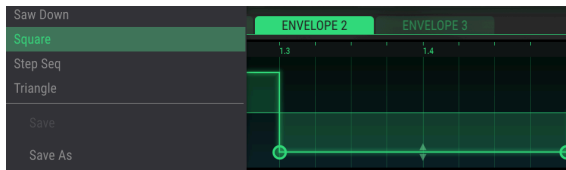
A simple Envelope, suitable for use as a basic LFO waveform

7.1.3.2. Complex presets

Other presets are more complex. These Envelopes could be used as the basis for a rhythm track, or to enhance one. They also are great starting points for your own rhythmic explorations.

A complex, rhythmic Envelope

♪: The Envelope Generators are fully adjustable, but any points that are moved will be forced to a location within the grid resolution (1/64th of the full window).

7.1.3.3. Save/delete Envelope Presets

Whenever you've altered one of the Envelopes and would like to save the changes, use the Save / Save As features inside the Envelope Preset menu.

- **Save**: this will overwrite the original preset, so if you would like to save them both use the Save As feature. (Factory templates cannot be overwritten or deleted.)
- **Save As**: With this option a window will open and ask you to give a new name to the Envelope Preset. Enter a name in the field and click the Save button.
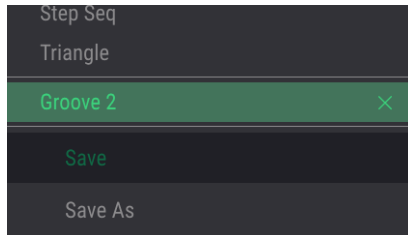
The Envelope Preset Save/Save As menu

To delete one of your Envelopes, open the Envelope Preset menu and click the X that corresponds to that preset. Factory templates cannot be overwritten or deleted.

7.1.4. Rate Multiplier

The Rate Multiplier can be used to speed up or slow down the Envelope segment times. A value of 1.0 has no effect, while values lower than 1.0 slow down the segment times and values higher than 1.0 will speed them up.

There are eight Rate Multiplier values: 0.125, 0.25, 0.5, 0.75, 1, 2,4, and 8. They are multiples of the master tempo, so a value of 2 will double the Envelope rate (i.e., cut the segment times in half) and a value of 0.5 will double the segment times (i.e., make the segment times twice as long).

The minimum value is 0.125, or eight times slower. The maximum value is 8.0, or eight times faster. Double-click the field to reset the value to 1.0.

> ♪: Each of the three Envelopes within the Filter preset has its own Rate Multiplier setting.

7.1.5. Trig Mode

The Trigger mode switch allows you to choose between two behaviors for the Envelope Generator: Loop mode and Audio Threshold mode.

> ♪: Each of the three Envelopes within the Filter preset has its own Trig Mode settings.

7.1.5.1. Loop

When Loop mode is selected the Envelope will loop endlessly after receiving a Note On message. When a looping Envelope is routed to a destination through the Modulation Matrix it will behave like a Low-Frequency Oscillator (LFO), and it will introduce cyclic modulation to the selected destination (filter frequency, pan position, etc.).

You can add points to the Envelope to make the loop more complex. The maximum number of points is 16, and the first and last points are always present, so you can add as many as 14 points. The loop start / end is always first and last points, respectively.

The overall rate of the loop is determined by a combination of the tempo of the master clock, the placement of the last point, and the Rate Multiplier for that Envelope.

> ♪: The Envelope Generators are set to Loop by default, and point 1 and point 16 will always share the same level value. The last point does not have an independent slope value, though.

7.1.5.2. Example: modify a looping Envelope

- Select the Default filter preset. It has the Master Cutoff value set at O so we'll be able to modulate it in either direction with obvious results. It also doesn't have any routes set up in the Modulation Matrix, so we'll learn a bit about that too as we work through the example.
- Click the Preset menu inside the Envelope Generator window and the menu will open.
- Select the Envelope preset Triangle from the list.
- In the Modulation Matrix, set up a modulation route between Envelope 1 and the Master Cutoff parameter.
- Set the modulation amount to roughly O.5OO.
- Make sure the Envelope Generator Trig mode is set to Loop.

The Envelope window and Modulation Matrix window should look like this:
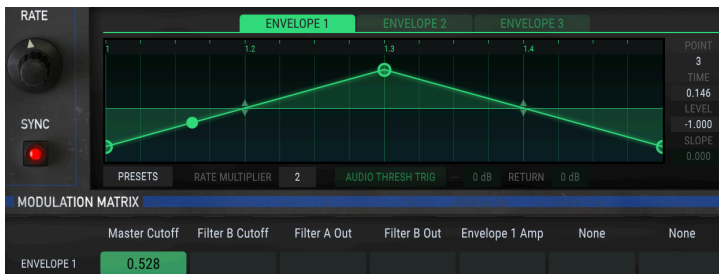
Once you have confirmed the settings, continue working through the example:

- Send a four-bar loop through the M12-Filter plugin to confirm that the audio is being processed. You should hear a distinct "wah-wah" as the Master Cutoff filter opens and closes.
- Double-click the middle point to set its value to O.
- Now grab the end point and move it slowly to the left, then slowly to the right. You'll hear the speed of the loop increase and decrease. It will also sound a bit "lumpy" because the length of the second half of the loop becomes shorter than the first half of the loop as the end point moves toward the middle.
- Return the end point to the far right side of the Envelope window.
- Drag the center point to the bottom of the Envelope window.
- Move the end point up and down, and notice that the first point moves at the same time. This is because the first and last points in a Envelope share the Level parameter.
- Return the end point to the bottom of the Envelope window.
- Add more points in the middle; the overall length of the Envelope loop will not change.
- You can remove points by right-clicking on them.
- Now set the new points to interesting levels [p.44] and adjust their slopes [p.45] and times [p.43] to suit your tastes.
- You may also want to experiment with different values for the Rate Multiplier so you can observe the difference that makes to the final sound.
- When you're ready, apply what you've learned in this example to Filtes A and B. The Envelope Generators can modulate their parameters independently through the Modulation Matrix, all while synchronized to the master clock of your DAW.

7.1.5.3. Audio Threshold

In audio threshold mode the Envelope is triggered and re-triggered when the input signal exceeds a certain threshold. The Envelope does not "loop" in this case, though it may seem to be looping when it retriggers. This behavior is dependent on the Return value (see below).

When you click the Audio Threshold Trigger button two additional parameter fields become accessible: Threshold and Return.

- **Threshold:** If the input signal exceeds this value the Envelope will be triggered. Range: -Inf (-80dB) to +6dB
- **Return:** If the input signal has already exceeded the threshold but has not yet fallen below the Return level, the Envelope will not be retriggered. However, once the input signal falls below the Return level the Envelope will be retriggered. Range: 0dB to +24dB

> ♪: In Loop mode the Threshold and Return parameter fields are not available and are greyed out.

Below this paragraph is a graphic example of an audio clip. The red line (**A**) represents the Threshold level and the green line (**B**) represents the Return level. When the audio level exceeds the red line the Envelope will be triggered. The Envelope will only retrigger after the audio level is reduced by the specified Return value (**C**) and falls below the green line again.
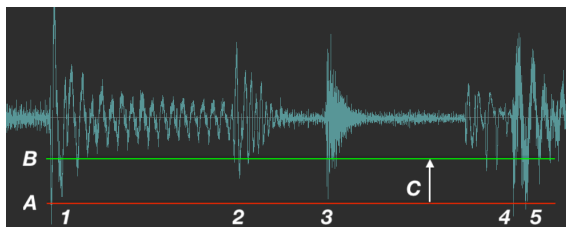
Here's what is happening at each numbered stage:

1. Audio level exceeds the Threshold level (**A**) and triggers the Envelope.
2. Audio level drops below the Return level (**B**) but has not yet risen above the Threshold level, so the Envelope does not retrigger.
3. Same as stage 2: the Envelope does not retrigger.
4. Audio level exceeds the Threshold level (**A**) and retriggers the Envelope.
5. Audio level has dropped by the Return value (**C**) between points 4 and 5 and exceeds the Threshold setting again, and so the Envelope retriggers twice in quick succession. You may want to adjust the Threshold slightly if this is not the effect you are hoping to achieve.

Here is another visual aid using a mixer channel.
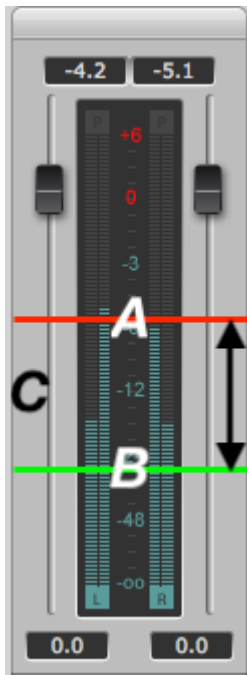
A: Threshold, B: Return level, C: Return value. Graphic: [DSP-Quattro](http://www.dsp-quattro.com)

7.2. Random source

7.2.1. Rate

This module will generate random values that can be routed through the Modulation Matrix [p.39] to any destination. The Rate knob controls the length of time between the random events.

The generation of these events can be synchronized to the master clock or left to run freely. To synchronize them to the clock, enable the Sync button beneath the Rate knob.

If Sync is enabled there are fourteen Rate parameter values available: 1/48, 1/32, 1/24, 1/16, 1/12, 1⁄8, 1⁄6, 1⁄4, 1⁄3, 1⁄2, 1, 2, 3, and 4. A value of 1 equals one bar of 4/4 time at the master tempo, while a value of 2 equals 2 bars and a value of 1/4 equals a quarter note.

The minimum value is 1/48, or a thirty-second note triplet. The maximum value is 4, or four bars. Double-click the field to reset the value to 1.0.

If the Sync button is not enabled the range of the Rate parameter is 0.01 Hz to 40 Hz.

7.2.2. Sync

When this button is active the Random source will be synchronized to the master clock. When synchronized the random values will be generated at subdivisions of the current tempo, with the frequency determined by the setting of the Rate knob (see above).

7.3. Modulation Oscillator (Mod Osc)

This module is a basic oscillator, but it is not directly tied into the audio output of the filter. It can be routed through the Modulation Matrix and used to modulate destinations at rates within the audio range.

7.3.1. Frequency

This knob adjusts the Mod Osc frequency. Range: 0.01 Hz to 10kHz

7.3.2. Wave

This field is where you will select the Mod Osc waveform. Use the left or right arrows on either side of the Wave field to cycle through the options: sine, triangle, square, saw up, and saw down.

It is also possible to click within the Wave window and a list of options will open. Make a selection and the window will close.
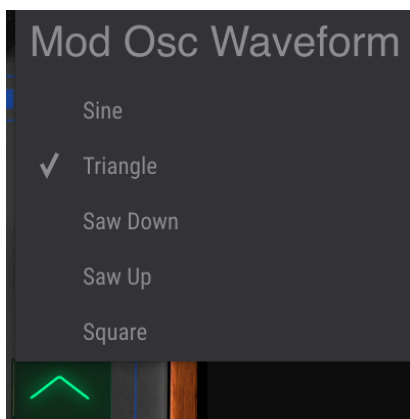
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