USER MANUAL

Matrix-12 V

ARTURIA
YOUR EXPERIENCE · YOUR SOUND

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1   INTRODUCTION

Arturia would like to thank you for purchasing our synthesizer model: the Matrix-12 V. We are confident it will prove to be an extremely valuable addition to your music production studio. If you've purchased our products before, you know we pride ourselves in faithfully recreating the sound and feel of the original instruments, down to the smallest detail. Matrix-12 V is no exception to this rule.

And if this is the first of our products you have owned, you are in for a treat! The synthesizer upon which this model is based was the absolute pinnacle of analog synthesizer technology at the time, light-years ahead of the competition.

1.1   Oberheim: an overview

1.1.1   Prelude

The 21st century is experiencing a Renaissance in the area of analog synthesizers, with many companies offering models of all colors, shapes and sizes. From towering modular synthesizers dripping with patch cords to Arturia's affordable and innovative MiniBrute series, fans of analog synthesis haven't had it so good for decades.

Back in the early '70s, however, there were really only two main synthesizer manufacturers with any notoriety: Bob Moog and his eponym company and ARP. These two companies rode a crest of technological and musical innovations for nearly a decade, with a rivalry akin to that of the Beatles and the Rolling Stones: some liked one, some liked the other, and the serious collectors owned both. And it seemed the more these companies 'divided the pie' of market share, the larger the pie became.

Then about 1975, riding in like something out of an American Western film, came a wave of white-faced synthesizers by the name of Oberheim. In addition to their unique appearance, they offered a different set of features and a fresh sound that ranged from creamy to brash. Bands around the world began appearing onstage and in the studio with these instruments, and many a memorable song and solo were built around them.

But we're getting ahead of ourselves. 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. Here's a bit of the background.

1.1.2   Lord of the Ring Modulators

The mid-1960s were a time when all musical boundaries began to be challenged. Unusual applications for electronic circuitry were at the heart of that, as artists looked for That Sound, the one that would set them apart from the crowd.

It was then Tom Oberheim first made a name for himself through his version of an electronic device called a ***ring modulator***. These somewhat simple circuits had been used in radio receivers for a long time, but when applied to a signal in the audio range the results were often other-worldly.

Case in point: if you have watched enough of the popular BBC show *Doctor Who*, you probably know about his nemeses, the Daleks. Their ominous, alien drill-sergeant voices owe much to the ring modulator (*"Exterminate! Exterminate!"*). And many of the bell-like tones you've heard from analog synths through the years also came from ring modulators. So these devices are still in use!

Initially Tom made a couple of these for some musician friends, one of whom was in a psychedelic band called *The United States of America*. Word got around, one thing led to another, and the next thing he knew he was being contacted by the Chicago Musical Instruments Company (CMI).

CMI wanted Tom to develop a ring modulator, which he did (the Maestro RM-1A). Good for them, and even better for us: this odd little box led directly to the formation of Oberheim Electronics.

The Maestro RM-1A, circa 1969

The RM-1A was followed by the Maestro PS-1, a phase shifter which also became very popular. And for the next five years or so a steady stream of interesting and useful products were designed for CMI by Tom Oberheim.

1.1.3   Technological innovations

Our main focus in these sections will be the synthesizer products Mr. Oberheim brought to life. However, some of his other contributions to the music world are noteworthy:

1.1.3.1 Polyphony

Sometimes it's good to be reminded: Back in the "old days" it wasn't possible to play a chord on a synthesizer keyboard. You could only play one note at a time.

But after becoming the official Los Angeles dealer for ARP synthesizers, Tom set out to change this. His early discoveries with the ARP 2500 and later modifications of the 2600 allowed *duophonic* performance on these synths (i.e., two notes could be played at once).

After this a "voice race" began, with two, then four, then five or six, and eventually eight voices becoming the standard for commercially available analog synthesizers. But the roots of this modern reality trace back to the early days of Oberheim Electronics.

1.1.3.2 Sequencers

About 1975 Oberheim introduced the DS-2a, a monophonic sequencer with a 144-note memory. (Well, it seemed like a lot at the time!) The beauty of this device was that it allowed users to enter notes from a keyboard, in real or step time, as opposed to dialing them in with a knob. Memory was shared by up to three sequences, which could be played back individually or chained and played back one after the other.

Unfortunately, there was no way to preserve the sequences for later use; all memory would be lost when the unit was powered down.

The Oberheim DS-2a, circa 1975

Another limitation with the DS-2a is that it would "hijack" the synth to which it was attached, rendering it unplayable by the user during playback. But this limitation led Oberheim to develop his Synthesizer Expansion Module, the beloved Oberheim SEM.

And again, the DS-2a paved the way for sequencers with more memory and polyphonic capabilities, including the Oberheim DSX and products from other manufacturers.

1.1.3.3 Presets

A famous, perhaps apocryphal story about a well-known multi-keyboardist puts the early days of analog synthesizers in perspective: it is said that whenever he came up with a sound he really liked on one of his Minimoogs, he would leave the controls on that synthesizer exactly as they were and go buy another Minimoog.

True or not, it is true that back in the day, people had to write down the values for every knob and switch if they wanted to recreate a sound later (and didn't want to buy another synth). Here again Oberheim led the pack in the development of the Programmer (1976), a device that could store and recall any one of 16 presets, each containing the parameters for up to eight SEMs at a time. And the SEMs didn't have to have identical settings, either, so each preset could contain wildly different sounds from SEM voice to SEM voice.

But the parameter controls on the Programmer also allowed the user to program all eight SEMs at the same time so they would sound like a single instrument if desired. Cassette backup was possible too, which allowed for the creation of a potentially infinite patch library.
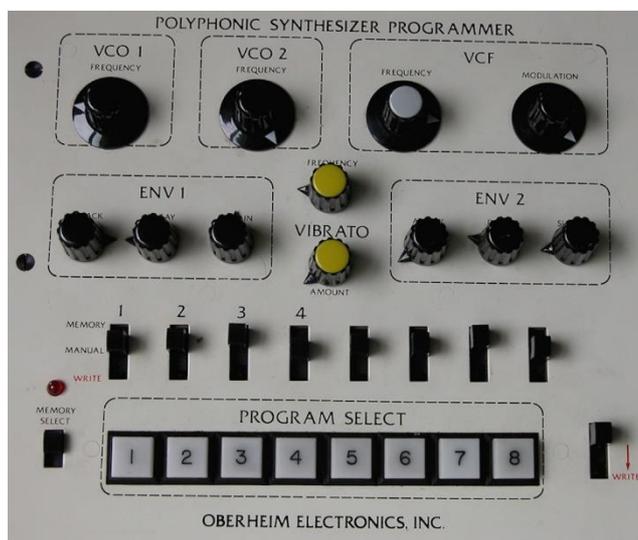
The Oberheim Programmer, circa 1976

1.1.3.4 Drum machines

Oberheim introduced the DMX (1981) and DX (1983), which became the foundation for scores of hit records in the '80s and beyond. They weren't the first sample-based drum machines, but soon they added a new twist: The Oberheim Prommer (1986) effectively turned the DMX and DX into samplers, in all their 8-bit glory. All the user had to do was open the drum machine and swap a chip (an EPROM), being careful to avoid static electricity and not to bend or break any pins!

The Oberheim DMX and DX drum machines

So the Linn LM-1 may have arrived first, but the features, the Prommer and the price point quickly established Oberheim drum machines as a significant presence in the music industry.

1.1.3.5 MIDI

Yep. MIDI. Tom Oberheim helped drive the development of the Musical Instrument Digital Interface protocol, a.k.a. MIDI, in three ways.

First, he brought the concept to life by implementing a digital communication bus for his own products (the Parallel Buss). This enabled devices such as the OB-8, the DMX and the DSX to become a synchronized, musically useful sequencing system. The home project studio was born!

Second, when two other manufacturers took notice of Oberheim's system, discussions began regarding a universal protocol that could be adopted by all manufacturers.

Third, and possibly most importantly, those three companies pitched this new idea to other major instrument manufacturers and convinced them it was a commercially viable concept. The course of the music world was forever changed through the efforts of Tom Oberheim and his colleagues.

1.2   The Oberheim synth family: a genealogy

Few product lines are as diverse as this one, even when only considering the synthesizers. From a small, single-voice expansion module to the behemoth Matrix 12, within the space of ten years this company covered a lot of ground.

But it all started with an innovative little box called the SEM.

1.2.1   SEM: the little synth that could

While selling ARP synthesizers, Oberheim began to design a device to help solve a small problem created by his DS-2a sequencer: the performer had to surrender control of the synthesizer to the sequencer while it was doing its thing. Few could

afford a second modular synth, not to mention the added bulk and complexity of carrying around two systems.

So Tom realized that a small, self-contained module could provide a cost-effective solution, complete with oscillators, filter and input/output connections. This unit could be connected to the sequencer while the user played the main synth. And so the Oberheim Synthesizer Expansion Module, better known as the SEM, came into being.

He enlisted the help of Scott Wedge and Dave Rossum, engineers who were pioneers in their own right as founders of E-mu Systems. Together they unveiled the SEM at the Audio Engineering Society Convention in Los Angeles in May, 1974.

The SEM was also a great way to expand the sound of a Minimoog or an ARP Odyssey by using their Control Voltage (CV) and Gate connectors to trigger the SEM.

Pictured below is Arturia's SEM V, our reverently rendered DSP model of the SEM:

The center panel of Arturia's SEM V modeling software

Patch cables had been replaced by internal connections, giving the unit a clean and simple appearance. But on close inspection you will see there is a lot of power behind that pretty face.

Among other things, people began to take notice of the SEM because of its innovative filter section. It had lowpass, highpass and bandpass filters like the Big Boys, but there was something new: a continuously variable filter control with lowpass on one side, highpass on the other, and a notch filter at the 12:00 setting. This became known as a "multi-mode filter", and it truly set the Oberheim sound apart from the competition.

The SEM proved to be sort of a "Cinderella synth": too beautiful to stay in the background for long. And so the humble expansion module took center stage, as Oberheim Electronics combined an increasing number of SEMs with a keyboard in a single, portable package.

1.2.2   The polysynths: rapid growth

Synth followed synth in quick succession, each standing on the other's shoulders: the two-voice TVS-1 (1975), the four-voice FVS-1 (also 1975), and the Eight-voice (1977), which are pictured below (note the inclusion of the Programmer on the left side of the two larger models):

The TVS-1

The FVS-1
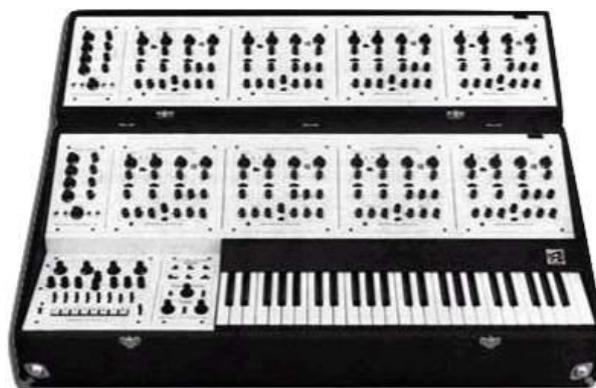
The last of the Great White Synths: the Oberheim 8-voice

Not pictured are the increasingly popular OB-1 (1978), OB-X (1979), OB-Xa (1980), OB-8 (1983), and Xpander (1984). Each is legendary and worthy of a section of its own!

Finally, Oberheim introduced their flagship, the Matrix 12, in 1985.

1.2.3   Crowning achievement: the Matrix 12

Though based on the same voice architecture as its predecessor, the Matrix 12 was definitely more than just two Xpanders soldered together. In addition to the 5-octave keyboard and two modulation levers, some important new features were added while a few significant hardware features were not included. But at an original retail price of almost $5,000, no doubt some shortcuts were necessary to keep this marvelous synth within the reach of more musicians.

So on the downside, the Matrix 12 lost the individual CV/gate and output jacks per voice (the latter being available as an upgrade). But on the plus side, the Matrix 12 gained some extra MIDI functionality and a per-voice detune parameter in the Multi Voice setup section, which you will be very glad to know we included in Matrix-12 V.

The Oberheim Matrix 12

To maintain basic patch compatibility with the Xpander, each of the Matrix 12's dual-oscillator voices boasted the same 15-mode voltage controlled filter (VCF), five envelopes and five low-frequency oscillators (LFOs) as well as a Lag processor (for portamento, etc.), three tracking generators and four ramp generators (i.e., modulation delays), 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: mods could be modulating the sources of other mods, etc.

And as usual with Arturia, our Matrix-12 V offers you all of the original features and many, many more.

1.2.4 Where the sound is found…

The list of albums made using the various Oberheim synthesizers could fill an encyclopedia. They have certainly filled the airwaves over the years:

Styx – The Grand Illusion	Jean Michel Jarre – Magnetic Fields
Rush – Moving Pictures	Bruce Hornsby & The Range – The Way It Is
The Police – Synchronicity	Madonna – Madonna
Prince – 1999	Tangerine Dream – Exit
Queen – The Game	Van Halen – 1984

This is only a small percentage of the songs, styles and artists fueled and driven by Oberheim products. They have clearly proven themselves as great creative tools, to say the least.

1.3 All good things come to an end…for a while

It's a sad truth that no matter how excellent a company's products are, they sometimes find it hard to stay afloat. Alas, this was true of Oberheim Electronics as well, which changed hands a number of times beginning in 1985. Tom stayed on

board for a couple of years and then struck out on his own, founding another company called Marion Systems.

Then in 2009 he announced he would be revising and reissuing some of his most famous early synthesizers, starting with the SEM. He followed that in 2014 with the Two Voice Pro.

Look for these products and more at tomoberheim.com.

1.4   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:

1.4.1   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.
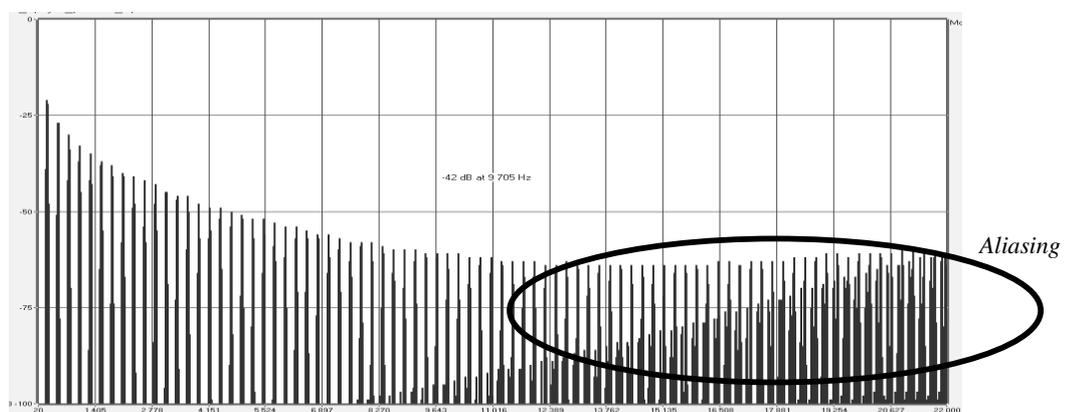
Linear frequency spectrum of a current well-known software synthesizer
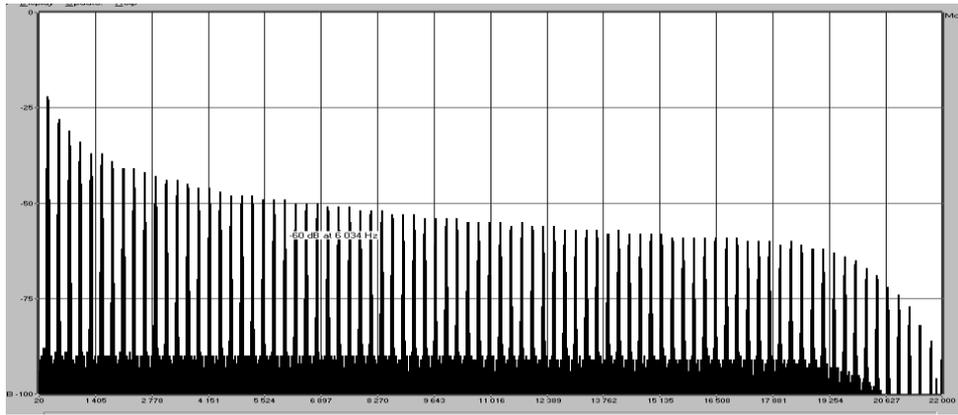
Linear frequency spectrum of an oscillator modeled with TAE®

1.4.2   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.
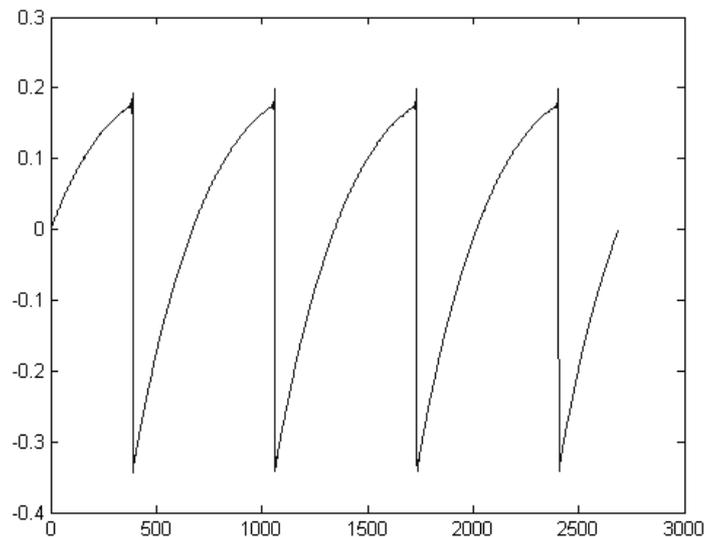
Temporal representation of the "sawtooth" waveform of a hardware synthesizer
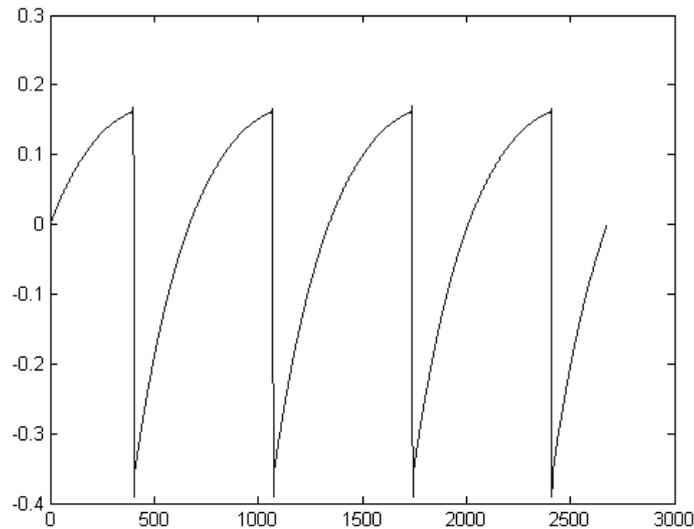
Temporal representation of a "sawtooth" waveform reproduced by TAE®

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.

1.4.3   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.
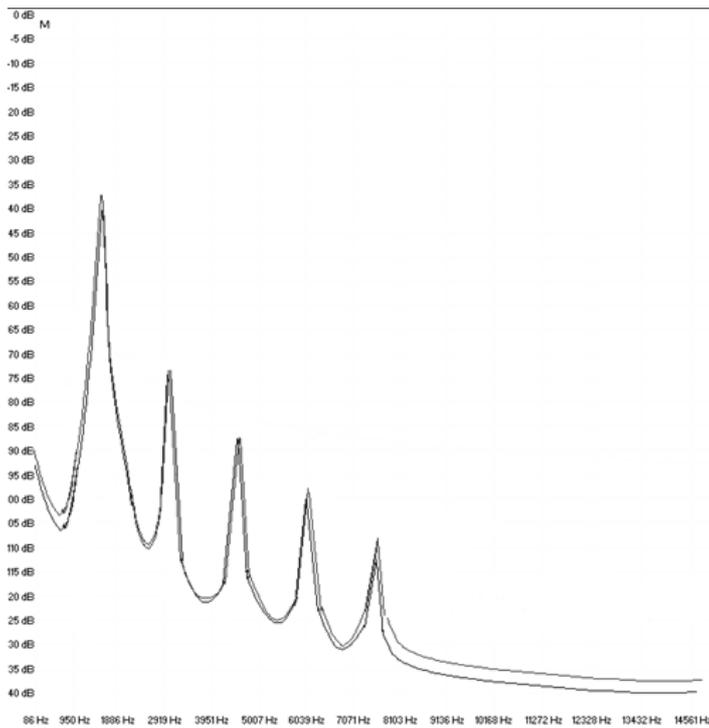
Comparison of harmonics generated by the filter circuits in self-oscillation
of TAE® and a hardware synthesizer

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 software model yet, the Matrix-12 V.

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

2   ACTIVATION AND FIRST START

2.1   Register and Activate

Matrix-12 V works on computers equipped with Windows 7 or later and Mac OS X 10.8 or later. You can use the stand-alone version or use Matrix-12 V as an Audio Units, AAX, VST2 or VST3 instrument.

Once Matrix-12 V has been installed, the next step is to register the software.

The registration process will require you to enter the serial number and the unlock code you received with the product.

In order to proceed, go to this web page and follow the instructions:

http://www.arturia.com/register

Note: If you don't have an Arturia account yet, you will need to create one. The process is quick, but it does require that you can access your email address during the registration process.

Once you have acquired an Arturia account you will be able to register the product.

2.2   Initial setup

2.2.1   Audio and MIDI settings: Windows

At the top left of the Matrix-12 V application is a pull-down menu. It contains various setup options. Initially you will need to go to the menu and choose the Audio Settings option to get sound and MIDI flowing in and out.

Matrix-12 V main menu

You will then see the Audio MIDI settings window. This works in the same way on both Windows and Mac OS X, although the names of the devices available to you will depend on the hardware you are using.
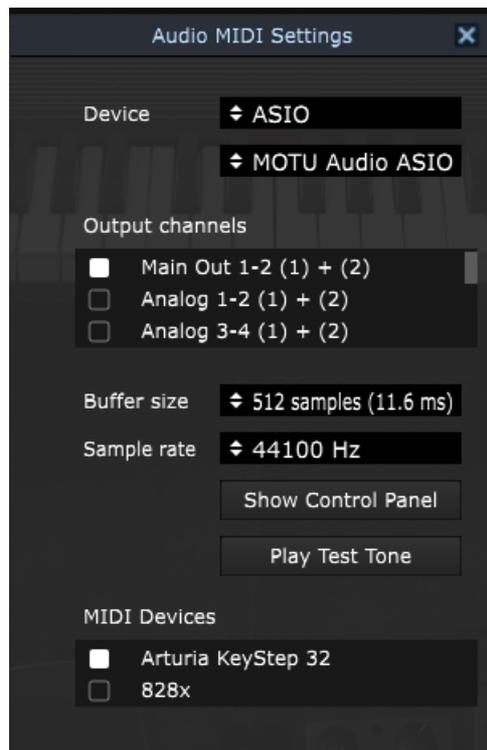
Audio and MIDI settings window

Starting from the top you have the following options:

- **Device** lets you choose which audio driver you want to use to route sound out of the instrument. This might be your computer's own driver like Windows Audio, or an ASIO driver. The name of your hardware interface may appear in this field.

- **Output Channels** lets you select which of the available outputs will be used to route audio out. If you only have two outputs, only two will appear as options. If you have more than two you can select a specific pair of outputs.

- The **Buffer Size** menu lets you select the size of the audio buffer your computer uses to calculate sound. A smaller buffer means lower latency between pressing a key and hearing the note. A larger buffer means a lower CPU load as the computer has more time to think, but can result in a small latency. Find the optimum buffer size for your system. A fast, modern computer should easily be able to operate at 256 or 128 sample buffer size without creating pops or clicks in the sound. If you are getting clicks, try raising the buffer a little. The latency is displayed on the right hand side of this menu.

- The **Sample Rate** menu lets you set the sample rate at which audio is sent out of the instrument. The options here will depend on the capability of your audio interface hardware though even most computers' own hardware can operate at up to 48kHz which is perfectly fine. Higher sample rates use more CPU power so unless you have a good reason to go up to 96kHz, then 44.1k or 48k is usually fine. The **Show Control Panel** button will jump to the system control panel for whatever audio device is selected.

- **Play Test Tone** helps you to troubleshoot audio issues by confirming whether sound can be heard through the correct device.

- Your connected MIDI devices will appear in the **MIDI Devices** area. Click the check box to accept MIDI from the device you want to use to trigger the instrument. In standalone mode, Matrix-12 V listens for all MIDI channels so there's no need to specify a channel. You can specify more than one MIDI device at once.

2.2.2 Audio and MIDI settings: Mac OS X

The process is very similar to initial setup for Windows and the menu is accessed in the same way. The difference is that OS X uses CoreAudio to handle audio routing and the audio device selection is made in the second dropdown menu. Apart from that, the options work the same way as described in the Windows section.
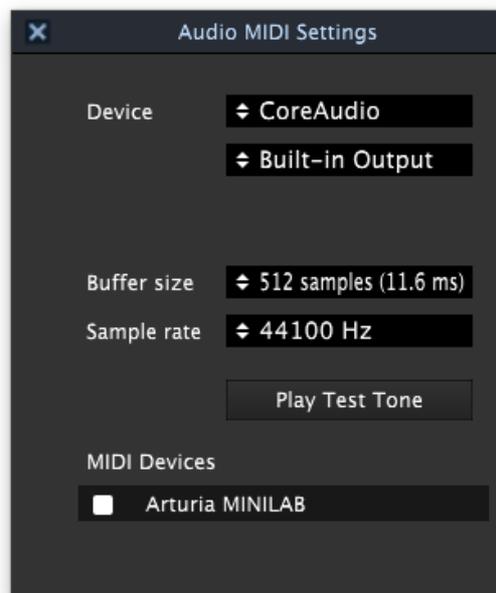
2.2.3 Using Matrix-12 V in plug-in mode

Matrix-12 V 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 instrument and its interface and settings work the same way as in standalone mode, with a couple of differences.

- You can automate numerous parameters using your DAW's automation system.

- You can use more than one instance of Matrix-12 V in a DAW project. In standalone mode you can only use one at once.

- You can route Matrix-12 V's audio outputs more creatively inside your DAW using the DAW's own audio routing system.

3 USER INTERFACE

In this chapter we will give an overview of the features available to you with Matrix-12 V. As with every Arturia product, we have gone to great lengths to make the use of this software instrument as simple and as much fun as possible, while also striving to make sure you never run out of new things to do with it as your knowledge expands. After reading this chapter you should be ready to delve as deeply into the workings of Matrix-12 V as you would like.

3.1 The virtual keyboard

The virtual keyboard lets you play a sound without connecting an external MIDI device; just click a key to hear the active Voice. Drag the cursor across the keys to hear a glissando.

The Matrix-12 V virtual keyboard and Levers

Note also the Levers to the left of the virtual keys. Lever 1 is available as a modulation source, while spring-loaded Lever 2 is usually dedicated to pitch bend purposes. Either can be routed to other destinations on the Modulation Page, though.

3.2 Toolbar

The toolbar that runs along the top edge of the instrument both in standalone and plug-in mode provides access to many useful features. Let's look at them in detail. The first seven of these options can be found by clicking on the Matrix-12 V section at the very top left hand corner of the instrument window.

3.2.1 Save Preset

The first option lets you save a preset. If you select this, 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 select some tags that describe the sound. 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.
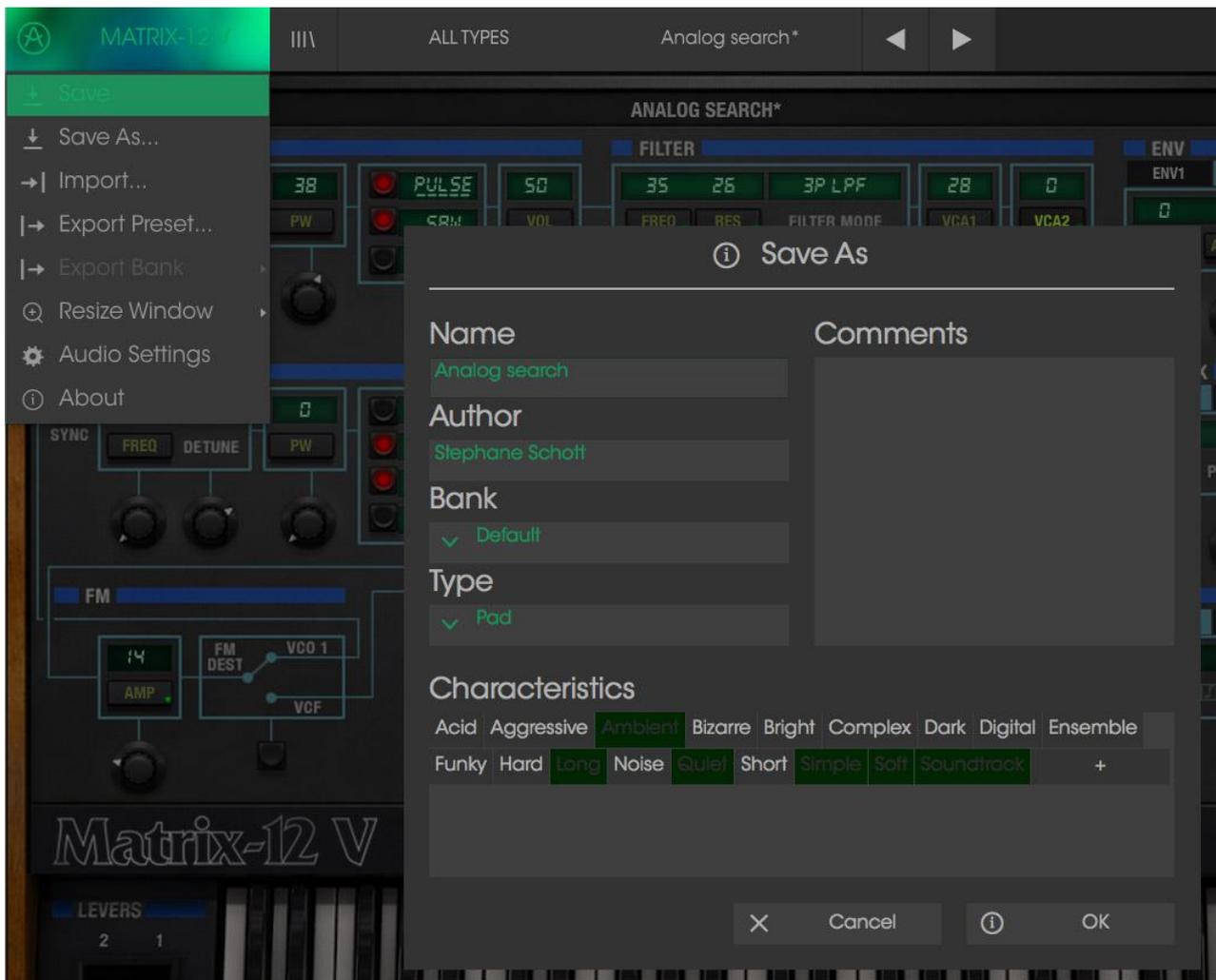
The Save Preset window

3.2.2 Save Preset As…

This works in the same way as the Save command, but lets you save a copy of the preset instead of saving over the original. It's useful for creating variations on patches but still keeping individual copies of each one.

3.2.3 Import preset

This command lets you import a preset file, which can be either a single preset or an entire bank of presets. Both types are stored in the .matrix format.

After selecting this option, the default path to these files will appear in the window, but you can navigate to whichever folder you are using.

3.2.4   Export preset

You can export and share 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.

3.2.5   Resize window options

The Matrix-12 V 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 might 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

3.2.6   Audio settings

Here you manage the way the instrument transmits sound and receives MIDI. See section 2.2 of the manual for full details on this.

3.2.7   Preset browser overview

The Preset browser is invoked by clicking the toolbar button that has four vertical lines. See section 3.3 of the manual for full details on this. The Filter, name field and left / right arrows in the toolbar all assist with preset selection.
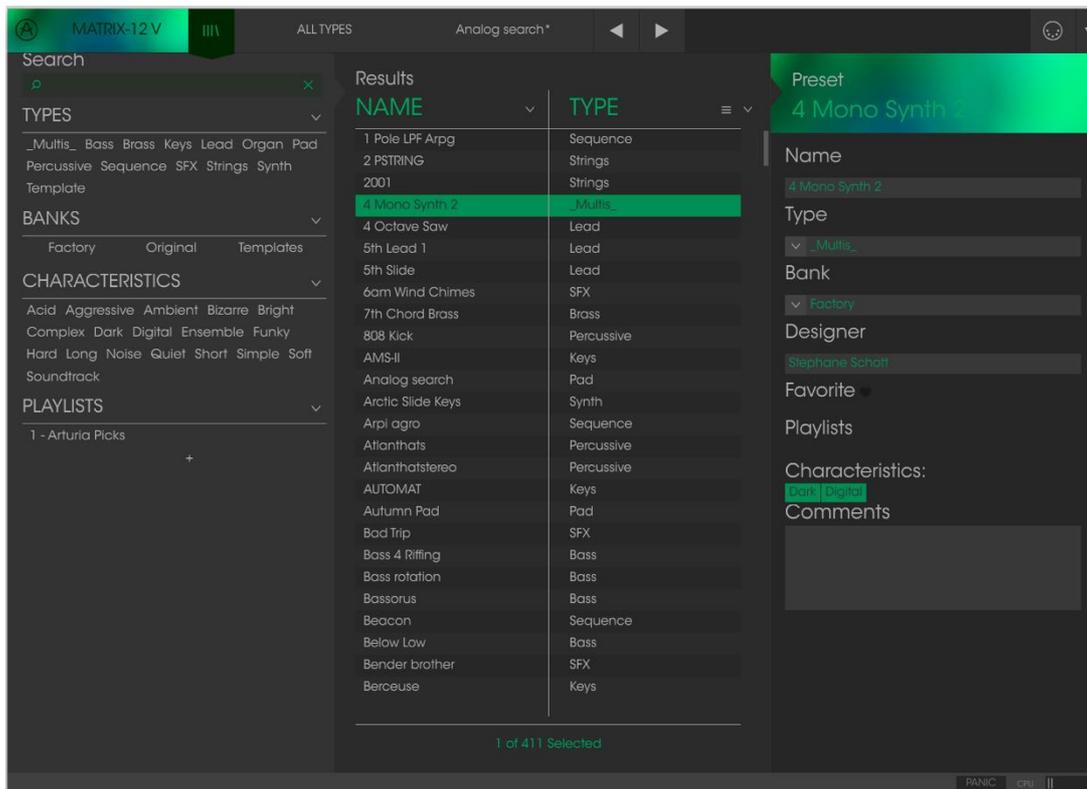
The Preset Browser

3.2.8 MIDI Learn assignment

The MIDI plug icon at the far right side of the toolbar places the instrument into MIDI learn mode. Parameters that can be assigned to MIDI controls will be shown in purple, and the idea is that you map physical buttons, knobs, faders or pedals from hardware MIDI controllers to specific destinations inside the instrument. A typical example might be to map a real expression pedal to the virtual volume pedal, or buttons on a controller to the effect switches so you can change the sound from your hardware keyboard.
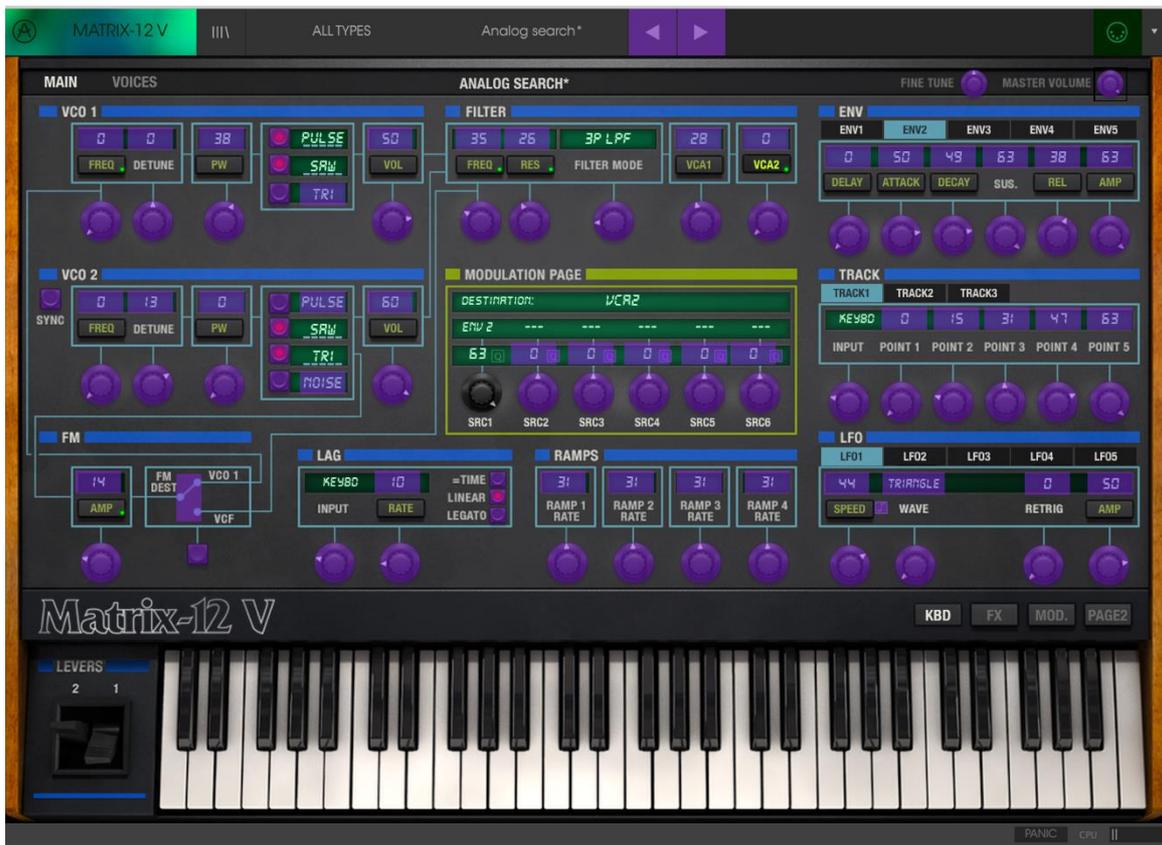
MIDI Learn mode

3.2.8.1 Assigning / unassigning controls

If you click on a purple area you'll put that control into learning mode. Move a physical knob or fader 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 a button to unassign the two from each other.

Filter cutoff frequency selected and assigned

3.2.8.2 Min / Max value sliders

There are also minimum and maximum value sliders that you can use to restrict the parameter change range to something other than 0%-100%. For example, you might want the filter cut-off be controllable via hardware from 30% to 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 other control if you like.

3.2.8.3 Relative control option

The final option in this window is a button labelled "Is Relative". It is optimized 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 (0-127, 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 massively out of their current setting as soon as you start to modify them.

3.2.8.4 Reserved MIDI CC numbers

Certain MIDI Continuous Controller (MIDI CC) numbers are reserved and cannot be reassigned to other controls. These are:

- PitchBend
- AfterTouch
- Ctrl Mod Wheel (CC #1)
- Ctrl Sustain On/Off (CC #64)
- Ctrl Expression (CC #11)
- Ctrl All Notes Off (CC #123)

All other MIDI CC numbers may be used to control any assignable parameter in Matrix-12 V.

3.2.9 MIDI controller configuration

There's a small arrow at the far right hand side of the toolbar that deals with MIDI controller configurations. 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 Matrix-12 V without having to build all the assignments from scratch each time you swap hardware.
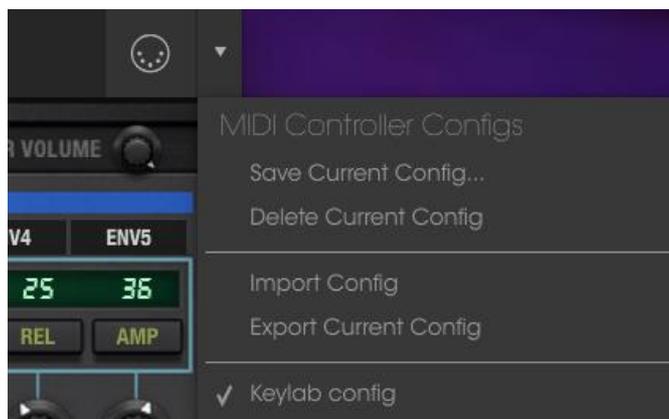
3.2.10 The lower toolbar

At the left hand side of the lower toolbar 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.
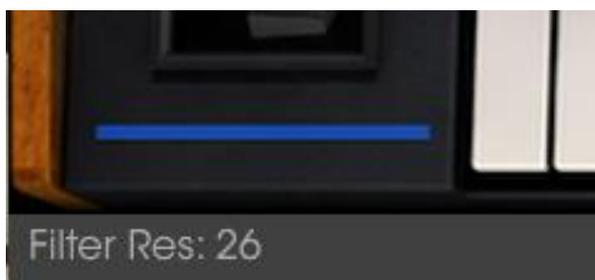
The Panic button can be pressed to reset all MIDI signals in the event of stuck notes or other issues. The Panic button is also MIDI-assignable.

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

3.3   The Preset Browser

The preset browser is how you search, load and manage sounds in Matrix-12 V. It has a couple of different views but they all access the same banks of presets.

To access the search view, click on the browser button (the icon looks a bit like books on a library shelf).
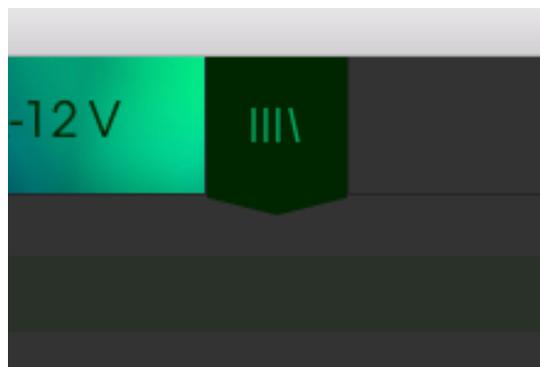
The Preset Browser button

3.3.1   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 X button in the search field to clear the search.
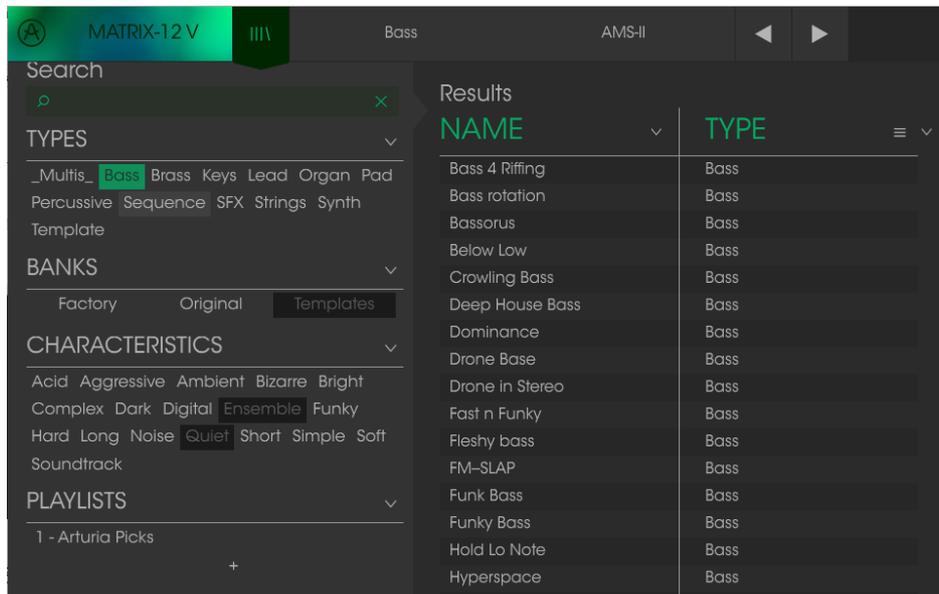
3.3.2 Using tags as a filter

You can also search using different tags. Clicking on a Type field shows only presets that match that tag. The tag fields can be shown or hidden by using the small down arrow buttons in their title fields. Results columns can be sorted by clicking the same arrow button in their own section.

You can use multiple search fields to perform narrower searches. So by entering a text search and also specifying type, bank and characteristics options you could 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. Using "Ctrl + click" (Windows) or "Cmd + click" (Mac) will allow you to select multiple elements in the same area.
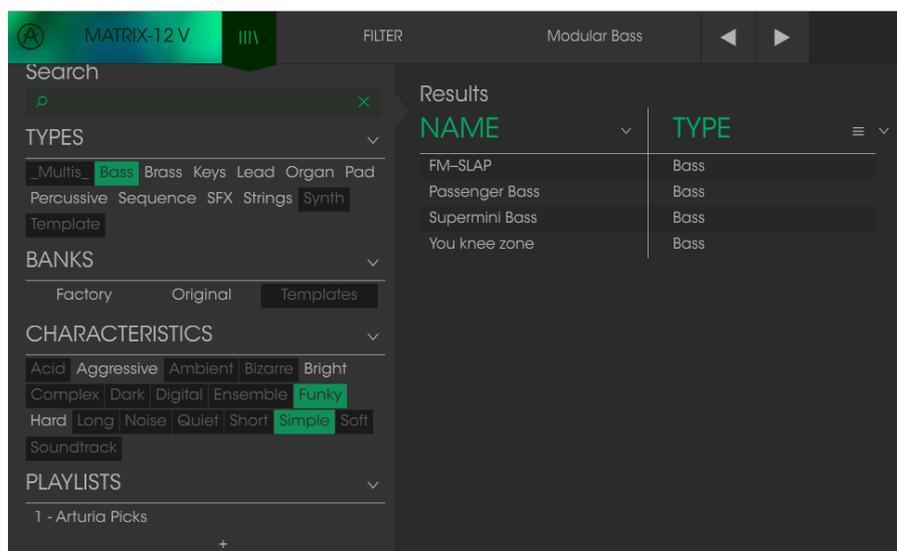
The second Results column can be switched to show Type, Sound Designer, Favorite or Bank tags depending on how you like to search. Click on its options menu button just next to its sort arrow.
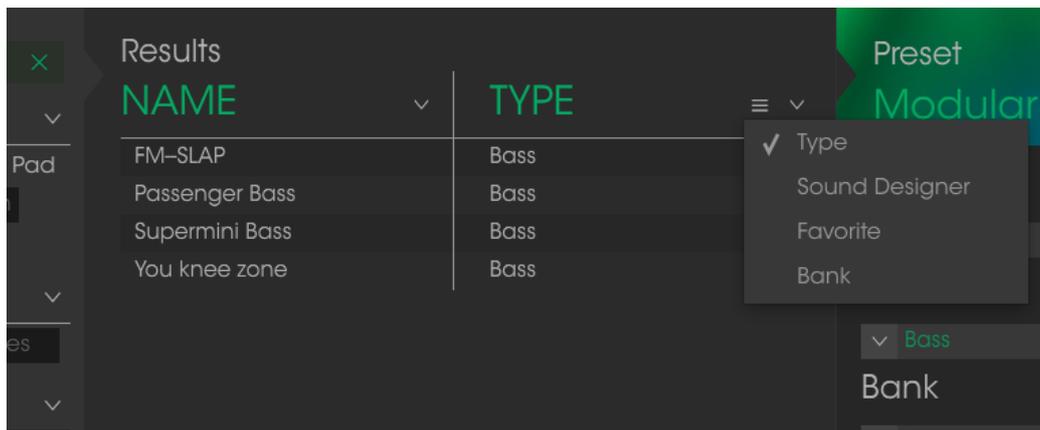
3.3.3 The Preset Info section

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

However, if you want to alter the information for a Factory preset you must first use the Save As command to re-save it as a User preset. After this the Info section will gain Edit and Delete buttons at the bottom of the window.

Click Edit and then make the desired changes, either by typing in one of the fields or by using a pull-down menu to change the Bank or Type. You can even add new Characteristics by clicking the + sign at the end of that list. Click Save when you are done.
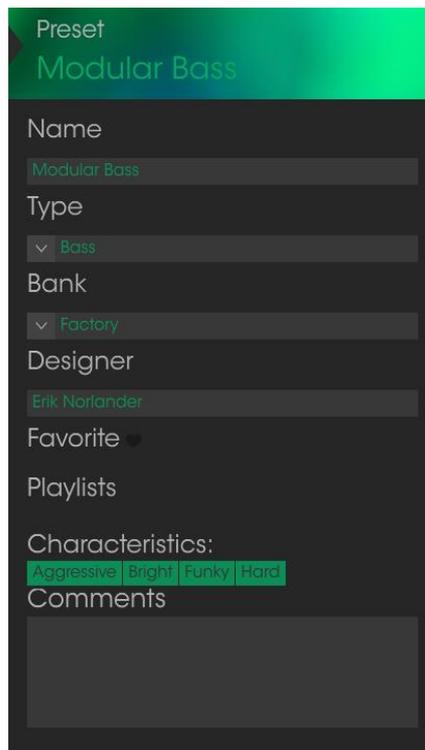
3.3.4   Preset selection: other methods

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 "Love" in the main search area, the results of that search will appear here.

Similarly, if you previously selected a Type in the Search field you would see the results of that search in this area instead.
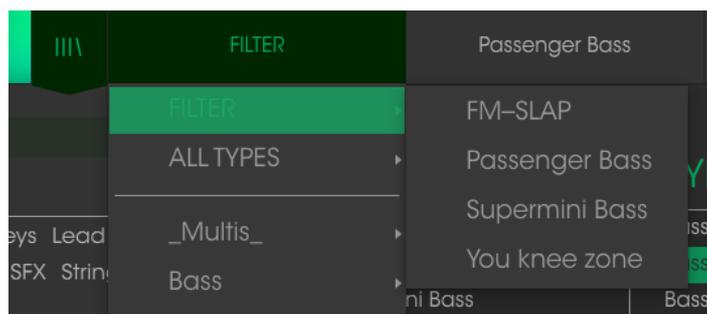
Filter results may differ based on Search criteria

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.

3.3.4.1 Selecting a preset by its Type

Clicking on the name field in the center of the toolbar 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 Characteristic such as "Funky" this shortcut menu will only show you presets that match that tag.

The left and right arrows in the toolbar 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.

3.3.5 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.

3.3.5.1 Add a playlist

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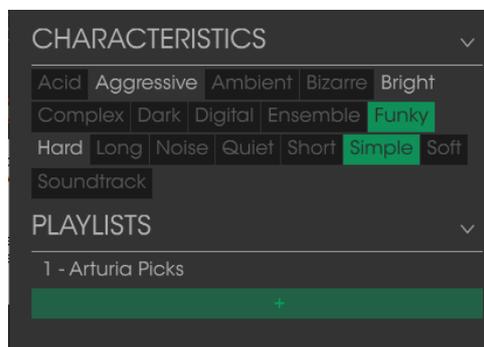
Give the playlist a name and it will appear in the Playlists menu. You can rename the playlist at any time; just click the pencil icon at the end of its row.

3.3.5.2 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.
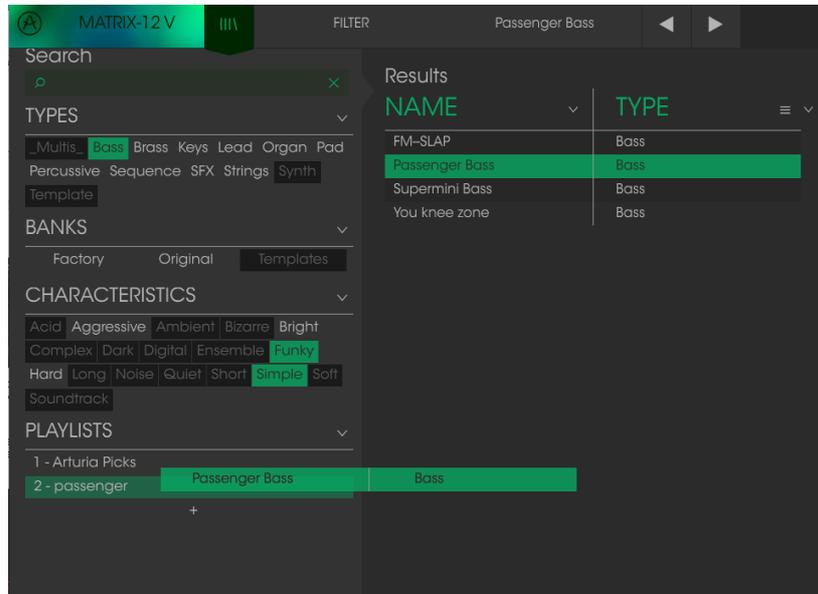
Click and drag from the Search Results list onto one of the playlists

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

3.3.5.3 Re-order the presets

Presets may be reorganized within a playlist. For example, to move a preset from slot 2 to slot 4, drag and drop the preset to the desired location.

This will move the preset into the new location.

3.3.5.4 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

3.3.5.5 Delete a playlist

To delete a playlist, click the x directly to the right of the playlist name.

Click the X to delete a playlist.

3.4 Overview of the main window components

Matrix-12 V has many features and several modes of operation. But it is helpful to know that each mode has things in common with the others. So let's take a look at the broad picture first.

3.4.1 Page bar

Beneath the tool bar is another strip called the Page bar. The Main, Voices and Program Chain pages are accessed through their respective labels. Note that two of these pages (Main and Voices) have slightly different capabilities depending on the mode of the selected preset.

The page bar will look a bit different in each mode, too, but these are the basic items you will always see.

Also constant to this region are the Master Volume and Fine Tune knobs, which are global controls affecting all Voices and modes equally.

An example of the Page bar in Single mode

An example of the Page bar in Multi mode

And of course the name of the current preset is always in the middle of the Page bar. If you see an asterisk after the name it means the preset has been edited. In this case, be sure to save your changes before selecting a new preset.

3.4.2   Parameter window

The center section in each mode is called the Parameter window. It has a very different appearance depending on the page and mode you have selected:

3.4.2.1 The Main page

Chances are high that you'll spend a majority of your Matrix-12 V time on the Main page, both in Single and Multi modes. The other pages are very useful and have lots of great features, but the Main page is at the center of it all:

The parameter window of the Main page

3.4.2.2 The Voices page

As with the Main page, the Voices page exists in both Single and Multi modes. But what it does in each mode is quite different.

In Single mode its parameters affect what happens with a single Voice. But in Multi mode its parameters can be applied to up to twelve Voices at once, each with its own settings.
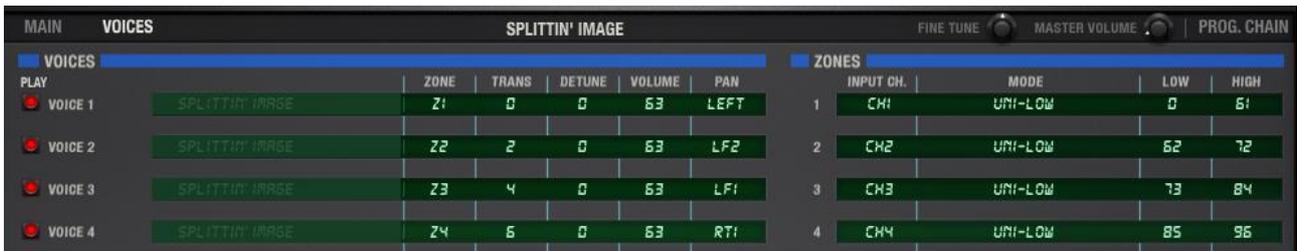
An example of the Voices page in Single mode
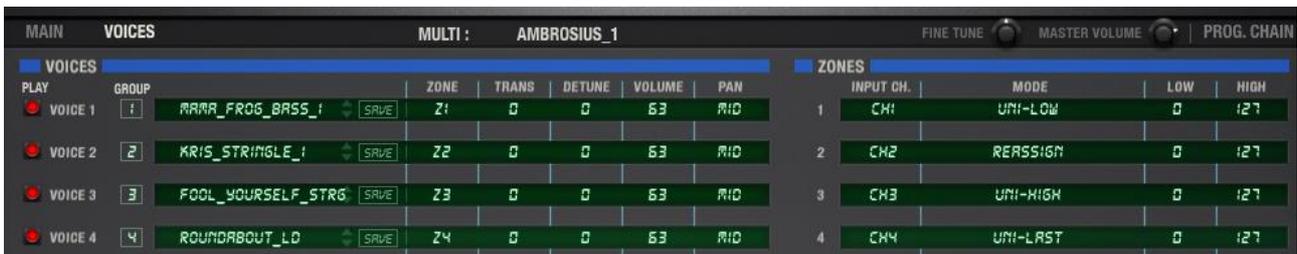
An example of the Voices page in Multi mode

3.4.3   Nameplate

The nameplate is the springboard for bottom panel functions when Matrix-12 V is on the Main page.

Cutaway view of the nameplate

Each button on the right side of the nameplate will change what is seen in the bottom panel.

3.4.4   Bottom panel

The nameplate has four buttons which alter the contents of the bottom panel (Main page only):

KBD: Keyboard. Click it and the virtual keyboard and levers show up.

FX: Effects..

MOD.: The Mod lists, i.e., lists of modulation routings.

PAGE2: The Page 2 parameters. These are not often changed when programming a sound, but their functions are important.

3.5   The tool bar: Managing presets

Presets can remember the settings for any Single or Multi you create with Matrix-12 V. When saving a preset you have the opportunity to designate the "bank" and "type" to which it belongs, so make as many banks as you need and fill each one with as many types and presets as you like. The "type" designation can help you categorize your presets into groups such as "Leads", "Pads", etc. There's no limit on the number of presets within a type, so use these features in the way that works for you.

Matrix-12 V ships with several banks of factory presets, each containing a number of types and presets. These cannot be overwritten, but they are great starting points for creating new presets. Then you can use the **SAVE AS** feature to save the changes as your own presets.

4 SINGLE MODE

Single mode is where it all begins, and it's really important to understand the available features. So now we're going to take a look at each parameter group and how they interact with one another.

4.1 Front Panel Functions

At first glance Matrix-12 V seems deceptively simple compared to modular synthesizers. There are no patch cables, for one thing.

However, beneath the straightforward front panel over 1,000 parameters are available for sculpting the sound and behavior of this marvelous instrument.

4.1.1 Overview of the parameter organization

When you first move to a new city its buildings, streets, lights and signs present themselves all at once, which can seem bewildering. But eventually the sights will become organized in your mind, often with the aid of a map.

And at first, Matrix-12 V may resemble an aerial view of a small city at night:

But the following sections should help you familiarize yourself quickly with the layout of Matrix-12 V.

4.1.1.1 Parameter groups

Built into the layout of the Parameter window is a "top down" user interface that starts with **parameter groups**, which are delineated by thick blue lines such as those for VCO 1 and FILTER, shown below:

The VCO1 and Filter parameter groups

In the pictures above the Frequency, Detune, Pulse Width, waveform and Volume parameters are all part of the VCO1 parameter group. Likewise, the Filter frequency, Resonance, Mode, and both VCA settings are part of the Filter parameter group.

Here's a list of all the parameter groups in Single mode:

Name	*Parameters*
VCO 1	Freq, detune, PW, waveform, volume
VCO 2	Sync, freq, detune, PW, waveform, volume
FM	Amplitude, destination
Filter	Freq, res, mode, VCA 1 level, VCA 2 level
Lag	Input, rate, mode switches
Ramps	Rate control (x4)
Env	Delay, attack, decay, sus, rel, amp (x5)
Track	Input, track point levels 1-5 (x3)
LFO	Speed, wave, input, retrig, amp (x5)

Many of those parameter groups have additional "members" that are available when you press the **Page 2** button.

4.1.1.2 Tabs

The next level of organization are the **tabs** inside certain parameter groups. For example, you can only see one envelope, tracking generator, and LFO at a time, but the tabs beneath the blue bars alert you to the fact that there are more than one of these parameter groups available.

Case in point: look at the parameter group ENV, where there are 5 tabs labeled ENV1, ENV2, etc. This means there are 5 envelopes available for programming a Voice.
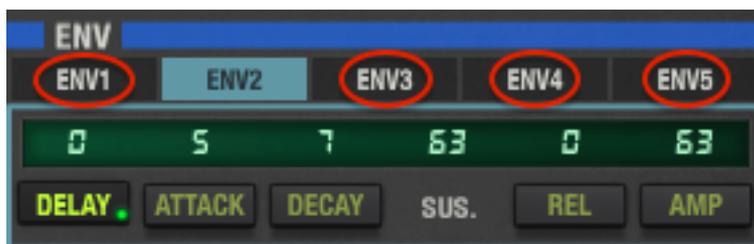
The Envelope parameter group: Tab 2 is selected; the other tabs are circled

A quick glance around the Matrix-12 V parameter window will show that tabs are also available for the LFOs (5 per Voice) and the Tracking generators (3 per Voice). There is no shortage of programming options here!

4.1.1.3 Parameters with and without mods

The foundations of each parameter group are the **parameters** themselves. But you'll notice some parameters have buttons containing their names and some do not.

The ones with buttons are available for further tweaking in the Modulation Page. And while the ones without those buttons can't be modulated, they can of course be edited right where they are by turning one of the knobs or toggling a switch, depending on the parameter.

One more thing: There are some lesser-used but very important parameters available on a sort of "sub-page", too. These are found by pressing the Page 2 button on the right side of the nameplate:
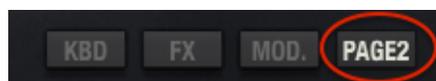
4.1.2 The nameplate

As mentioned previously, the function of the nameplate is to allow easy access to the four different aspects of the bottom panel. All four figure heavily into what a Voice can do, so the buttons to select them are conveniently located.

4.1.3 The bottom panel

Selecting the various buttons in the nameplate reveals different features of the Main page. These buttons are not available on the other pages; only the virtual keyboard will be seen there.

4.1.3.1 Virtual keyboard

The virtual keyboard lets you play a sound without connecting an external MIDI device; just click a key to hear the active Voice. Drag the cursor across the keys to hear a glissando.

The Matrix-12 V virtual keyboard and Levers

Note also the Levers to the left of the virtual keys. Lever 1 is available as a modulation source, while spring-loaded Lever 2 is usually dedicated to pitch bend purposes. Either can be routed to other destinations on the Modulation Page, though.

4.1.3.2 FX

When the FX button is pressed the bottom panel will reveal the two FX modules associated with the current preset, if any.

If none are present, it's a simple thing to add them.

Keep in mind that these are master effects: only two are available at one time in any of the modes. This means that all the Voices in a Multi will share two FX modules, not twenty-four!

4.1.3.3 Mod

This is the very heart of the power of Matrix-12 V. Here you can route any one of 27 sources to any one of 47 destinations, and do it 40 times over. The refinements and spur-of-the-moment decisions the Modulation Page puts at your disposal are listed here in the bottom panel, almost spreadsheet-style, when the Mod button is clicked.

As powerful as this section is, it is quite easy to use.

4.1.3.4 Page 2

The Page 2 button reveals a second page of Voice parameters. These actually play a support role to their respective parameter groups (VCO1, Filter, etc.).

There are tabs here too, meaning the envelope modes can be set differently for each of the five envelopes in the Voice, for example.

4.2   Parameter groups

Let's start by focusing on one of Matrix-12 V's parameter groups and get a feel for how Single mode works.

Note: For the examples in the rest of chapter 4 we will assume you have already selected a Voice in Single mode, not from the Main page in Multi mode.

4.2.1   VCOs

"VCO" stands for "Voltage Controlled Oscillator", the most basic building block of sound in an analog synthesizer. The quickest way to make a large change in the tone of a Voice is through its waveforms. In this section we'll cover some of the simpler ways to do that.

In the upper left-hand area of the Parameter window is a blue bar labeled VCO 1. All of the parameters under that bar are part of the VCO 1 parameter group.

You can see from the picture below that the parameters are almost identical for VCOs 1 and 2, so we'll describe them all in this section.

The VCO 1 & 2 parameter groups

The Page 2 VCO parameters are exactly the same for both oscillators:

The Page 2 parameters for VCO 1 & 2

We'll cover these at the same time as the other VCO parameters. Here's what they do:

4.2.1.1 VCO Frequency (Freq)

The fundamental pitch of the oscillator is set here, with a range of 0-63 semitones (chromatic steps). Click the button just below the value and the VCO frequency parameter will come into focus on the Modulation Page. Any one of 27 mod sources can then modulate the oscillator frequency, a lot or a little, in a positive or negative direction.

4.2.1.2 VCO Detune

This is literally a fine tune control, able to gently nudge the oscillator frequency flat or sharp in smaller increments. Its range is -31 to +31, and it is capable of even more precision by holding the control key while adjusting the value.

4.2.1.3 VCO Pulse Width (PW)

This control affects only the Pulse wave form, changing it from "thin" to full and then to a sort of "inverted thin" in a range from 0 to 63, with 32 as the midpoint. The value of 32 is a full square wave, with a 50% duty cycle and the requisite "hollow" sound caused by the presence of only the odd-numbered harmonics.

Note again the presence of a Mod button, meaning the pulse width can be modulated by a source such as an LFO. This is a great way to bring a sound "alive" by causing its harmonic content to shift constantly. (As long as we're giving out programming tips, you may want to use an additional, static mod such as the keyboard to tame the PW modulation in the lower frequency ranges. Or not.)

4.2.1.4 VCO waveform

There are three choices here for each VCO (pulse, saw, triangle) and a fourth for VCO 2 (noise). Here's a brief description of each:

Pulse: A square wave. See the previous section 4.2.1.3 for its description.

Saw i.e., Sawtooth: When unfiltered it is sort of brash or buzzy, and great for leads that cut through.

Tri a.k.a. Triangle: It's a simple, pure waveform akin to a sine wave.

Noise (VCO 2 only): Random frequencies all at once; great for wind effects, electronic drums, etc.

4.2.1.5 VCO Volume

This governs the output level of the oscillator. And it's equipped with a Mod button so the VCO can fade in and out as desired. This effect can cause a subtle tremolo or an intermittent chorusing, for example, depending on how the two VCOs are interacting. It can also cause subtle or radical timbral shifts if VCO 2 is also synced to the frequency of VCO 1. More about that in the next section.

4.2.1.6 Sync (VCO 2 only)

Sync stands for "synchronize", and when this button is pressed the relationship of the two VCOs changes drastically. Rather than being completely independent, the Sync button slaves the waveform(s) of VCO 2 to VCO 1 and forces them to output a single, complex waveform.

Lower the frequency of VCO 1 to zero and experiment with changing the frequency of VCO 2, either manually or by means of a mod routing (envelopes can be loads of fun here). The resultant timbral shifts are unlike anything else a synthesizer can produce, ranging from a controlled scream or an aggressive, biting tone to a subtle shimmer, depending on the mod source and amount.

4.2.1.7 VCO Page 2

There are other, less "glamorous" parameters available for the VCO parameter groups. To access these, press the Page 2 button in the nameplate. The bottom panel will show the Page 2 parameters for six different parameter groups, two of which are identical (VCO 1 and 2). We'll cover only those parameters in this section; the rest will be covered with their own parameter groups.

Keyboard: This decides whether the VCO frequency will change when different keys are played.

Lag: When enabled, the VCO frequency will respond to the settings of the Lag generator. This is what enables "glide", or portamento. It can be cool to have one VCO "scoop" while the other is static, so try enabling Lag for one and disabling it for the other.

Pitch bend: If you want the VCO frequency to ignore lever 1, disable this switch.

Vibrato: Same thing: if you want a VCO to ignore lever 2, switch off this parameter.

These seem like simple parameters, being only on/off switches. But as you can see, their settings can have a huge impact on the overall sound and behavior of a Voice.

4.2.2   FM

FM is an abbreviation for "Frequency Modulation." Did you know analog synthesizers were doing this form of synthesis long before those famous Yamaha synths? The concepts are very similar: the FM parameters in Matrix-12 V can put VCO 1 and VCO 2 into a carrier/modulator relationship, or do a similar thing to VCO 2 and the Filter.

The FM parameter group contains the following parameters:

Amp: The amount of the FM effect being sent to VCO 1 or the Filter

FM Dest: Selects the destination for the FM effect: VCO 1 or the Filter

This is where things become a bit more complicated to describe. But it will help to take another look at the Parameter window, specifically the thin blue lines connecting VCO 1 and VCO 2 to the FM parameter group.

We're going to take a close look at what is meant by **Oscillator FM**.

In the picture below the red arrow traces one of the paths for the output of VCO 2, thanks to a hardwired connection it has to the FM input. To hear this effect most clearly, set the VCO 2 VOL parameter to zero:

VCO 2 being fed into the FM input

As you trace the blue line past the red arrow and through the FM AMP box, note the connection from FM DEST to OSC 1 in the second blue box. This establishes the connection to VCO 1 courtesy of the FM DEST switch, which we'll describe later.

There's a problem, though: the input Amp amount is set to 0. So you won't hear an effect on the sound unless one of its active mod routings is pushing the FM input level in a positive direction.

Here's a larger picture that traces the path of the FM signal to VCO 1 after raising the Amp amount:

FM output being fed into VCO 1

That, ladies and gentlemen, is how Oscillator FM works. The frequency of VCO 2 affects the VCO 1 waveform in ways that are impossible with normal additive synthesis or even Oscillator sync. And using a mod routing to change the VCO 2 frequency with an envelope, for example, will cause the harmonic content of VCO 1 to change over time in ways an analog filter would not be able to produce, either.

4.2.2.1 Filter FM

A similar path exists between the FM output and the Filter section. Simply press the FM DEST toggle button at the bottom of the FM parameter group and the blue line will point downward, making the connection to VCF. You can trace the connection to the Filter section by following those little blue lines in the Parameter window, as we did with the connection to VCO 1.

In a nutshell, Filter FM in Matrix-12 V uses the VCO 2 Triangle waveform to turn the filter into an oscillator. This effect is best heard with a high Filter Resonance value, and you may want to try different filter types and filter frequencies to get a feel for what is happening. You may enjoy hearing the direct output of VCO 2 at the same time, but for now leave it set to zero. There's a particular "grunge" caused by Filter FM that you'll want to hear clearly.

And of course, your friendly neighbourhood Modulation Page stands ready to control the VCO 2 frequency in any number of ways.

4.2.3 Filter

This parameter group contains one of the most distinctive features of the Matrix-12 V sound. Rather than merely offering the requisite 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 total!

As sound-sculpting tools, the filter parameters rival the oscillators for flexibility. Take a look at the parameters:

4.2.3.1 Filter Frequency (Freq)

This is probably the first knob anyone tries on any synth: "How do the filters sound". The amount and type of change you hear depends on a number of factors, but rest assured the Matrix-12 V filters provide 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.

4.2.3.2 Filter Resonance (Res)

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. Self-oscillation is possible when the resonance is at maximum, so be careful not to destroy any audio components (like your ears!).

4.2.3.3 Filter Mode

Sometimes it helps to have a visualization of something that affects the audio realm. The following graphic actually opens up as a window inside Matrix-12 V when you click in the Filter Mode value field. It shows all 15 of the Matrix-12 V filter modes:
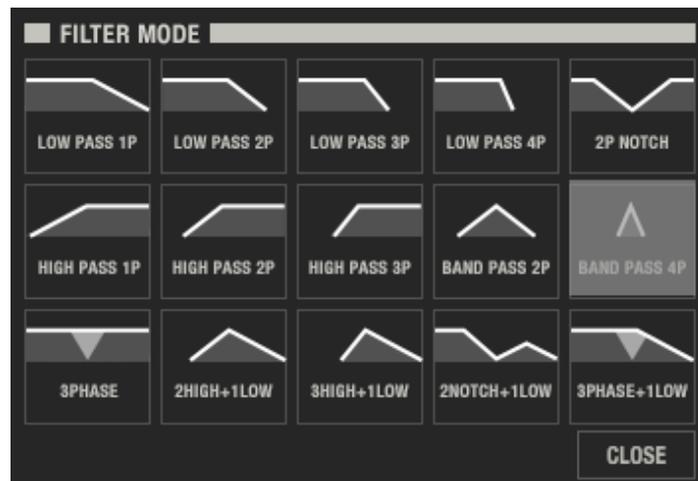
The Matrix-12 V Filter mode selection window, with Band Pass 4P selected

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, simply turn the filter mode knob to dial up the one you want.

Click the Close button to exit without making a change, or select one of the graphics and the window will close itself.

4.2.3.4 Filter Page 2

As with the VCOs, there are other parameters available for the Filter parameter group. To access these, press the Page 2 button in the nameplate. The bottom panel will show the Page 2 parameters for six different parameter groups, including those related to the Filter. We'll cover only those parameters in this section; the rest will be covered with their respective parameter groups.

Keyboard: Tells the Filter frequency to "track" the keyboard, raising the corner point for higher notes and lowering it for lower notes.

Lag: When this is enabled the Filter frequency will be affected by the settings of the Lag generator. Note: when the Lag section's Legato setting is off you may hear the filter sweep in a jagged, "stair step" manner. But if that sounds interesting, try it!

Pitch bend: If you want Lever 1 to sweep the Filter frequency, enable this switch.

Vibrato: Likewise, switch on this parameter to allow Lever 2 to sweep the Filter frequency.

Though these are basic on/off controls, you can see that their settings can have a significant impact on the overall sound and responsiveness of a Voice.

4.2.3.5 VCAs

As with the VCOs, VCA 1 and VCA 2 are basically identical so we'll cover them both in this section. The main difference between them is that they are "wired" in series; in other words, VCA 1 comes first in the signal path and VCA 2 comes last.

The VCAs are the final stage in getting the signal out of Matrix-12 V and into the world, a journey that began with the VCOs, traveled through the VCF, and now needs to be allowed to pass through these two parameters.

Assuming no mods are active, if both VCAs are set to zero then no sound will be heard. But simply raising their levels isn't always a good idea: when the output is wide open and not being shaped in some way, it means any note that has been played will come on and stay on.

So the thing to do when starting out is to set VCA 1 to a value higher than 0 (63 is max) and use at least one mod routing to control the VCA 2 level. This job is usually handled by an envelope being triggered by the keyboard, which is the most predictable way of controlling how quickly the sound begins and how long it sticks around.

So why are there two VCAs? Wouldn't it be simpler just to control one?

Simpler, yes, but also much less flexible. For example, velocity, aftertouch, a tracking generator and/or a couple of LFOs can be routed to VCA 1 for certain types of level control while still leaving the envelope controlling the overarching output characteristics of VCA 2.

In fact, between the two VCAs a combined total of 12 mod routings can be set up for the final output stage (6 per VCA). With the VCAs serving as the gatekeepers for the output of a Voice, that much control can be a good thing.

4.2.4 Lag

Simply put, a lag generator affects the rate of change between two data points. In synthesis terms this is often thought of in relationship to pitch, as in "this is what controls the speed and type of the glide between two notes" (i.e., the "portamento speed" or "glide time").

But the output of the Matrix-12 V lag generator can be applied to any number of targets, not just pitch, and use a source such as an LFO or a Lever, not just the keyboard. Generally the mod routes are handled through the Modulation Page, but the Page 2 button reveals hardwired routings for the VCOs, the Filter, and the LFOs.

These are the Lag parameters:

Input: Select the source with the knob or click the value field and select one that way

Rate: Sets the rate of the Lag effect. The mod button allows a source to vary the Lag rate.

= **Time**: Only functions when Linear response is selected. It works like this: "Equal time" is selected with keyboard as the input. When middle C is played and then the C one octave up is played, let's say the time it takes the pitch to reach the upper C is 1 second. If you then played two notes 5 octaves apart, the glide time from one to the other would also be 1 second, even though the distance is bigger.

Linear: Choose the lag curve here: when this is lit, the response is Linear; when unlit, the response follows an Exponential curve.

Legato: Legato provides a smooth, gradual response. When deselected, the result is a "stepped" response. In terms of pitch, the result would be a chromatic scale between two notes, not a "pitch bend."

Again, you are free to insert the Lag generator between any mod source and any mod destination.

4.2.5   Ramps

What a ramp generator does is provide a simple mod source that can be routed positively or negatively to affect a mod destination, such as VCO Frequency or the rate of an LFO. The source rises from 0 to maximum in a linear fashion, then remains at its maximal value.

There are four independent Ramp sources, each with its own rate control and Page 2 settings. You can specify whether the Ramp will rise only once after it is triggered by the keyboard and then wait for another trigger, or be triggered repeatedly by an LFO.

Suggested experiment: Try routing these to the rate or amplitude of an LFO with the LFO routed to the VCOs. This is tons of fun, and at one extreme can provide a beautiful, subtle shift in vibrato; the other extreme will set new standards for aural insanity.

You'll only see the Rate parameter for each Ramp until you click the Page 2 button. Use the Tabs to select between Ramps 1-4 on Page 2:

Trigger: Selects single or multiple trigger response

LFO Trigger: Allows a Ramp to be triggered by a specific LFO. When clicked it reveals the other parameters.

Gated: Instructs the Ramp to be triggered by an LFO only when a key is being pressed.

LFO select: Offers a choice of Ramp triggers: LFO 1-5 or VIB.

The VIB parameters are found on the Voices Page.

4.2.6 Envelopes (Env)

If it weren't for envelopes it would be very difficult to achieve any subtlety when programming; every sound would simply turn on and off. (There's a way to do that, too, when needed: that's what a Gate does.)

Every sound in the universe has an envelope. Some are more complex than others, but most have at least four of the five stages offered by each of the five Matrix-12 V envelopes:

Delay: Specifies the time that will elapse before the envelope begins

Attack: Controls the rapidity of the onset of the envelope once it starts: instant, lazy, slow, etc.

Decay: Once the envelope reaches its peak, this tells it how quickly to fall to the Sustain level

Sustain: The envelope will remain at this plateau until the note is released

Amp: The overall depth of the envelope

The Page 2 button provides some very important parameters which define the basic characteristics of each envelope:

Reset: Reset will cause the envelope to start again each time a new trigger event occurs

Freerun: Forces the envelope through all of its stages when triggered, regardless of whether the source gate has closed (for example, even after the key has been released)

DADR: Further simplifies the envelope by removing the Sustain stage

Trigger: Selects single or multiple trigger response

LFO Trigger: Allows an Env to be triggered by a specific LFO. When clicked it reveals the other parameters.

Gated: Instructs the Env to be triggered by an LFO only when a key is being pressed. In this case once the Release stage begins the LFO input will cease, which is a cool option.

LFO select: Offers a choice of Env triggers: LFO 1-5 or VIB.

Remember: the Tabs allow you to make a selection between Envelopes 1-5, and the mod buttons will bring their parameters into focus in the Modulation Page for the creation of mod routes.

4.2.7 Tracking Generators (Track)

There are three Track parameter groups per Voice, and what they do is similar to what the Lag processors can do: process an incoming signal and reshape it according to the Track settings.

One way to visualize their function is to think of them like an FX insert: when configured correctly they will exist between the source and destination of a mod routing. Let's say you have set up a simple mod route with LFO 1 as the source and VCO 1 Frequency as the destination. Now instead of this direct connection, choose LFO 1 as the Track 1 input and change the mod source from LFO 1 to Track 1. Now the LFO will pass through the Tracking generator on its way to VCO 1 Frequency.

The Track points are evenly spaced, so in the case of an LFO, Point 1 affects the initial stage of the LFO; Point 2 affects the value at ¼ of the way through the LFO cycle; Point 3 affects the value at the halfway point; and so on until Point 5, which affects the final stage of the LFO cycle. Then the process begins again. You can shape some interesting peaks and valleys in the modulation routing, even with a simple triangle wave as the Track input source.

The input can be any one of the 27 modulation sources, and the Point values are from 0-63. Negative modulation directions are achieved through the Modulation Page.

4.2.8   LFOs

"LFO" is the easy way to say "Low Frequency Oscillator." Matrix-12 V has five of them available per Voice, which can make for some very interesting modulation routings. Each LFO has seven waveform choices that range from rounded (Triangle) to potentially abrupt (Square, Saw Up/Down) to nearly unpredictable (Random, Noise, Sample).

Let's look at the waveforms before we talk about the other parameters in the LFO group.

Triangle: On a graph this would look like a ripple on the surface of a pond after tossing in a pebble, with a smooth rise and a smooth fall.

Square: More like the teeth of the gears within a clock: sudden rise, sudden fall, with a steady state between transitions.

Saw up: Gradual, linear transition from zero to maximum with a sudden return to zero.

Saw down: Gradual, linear transition from maximum to zero with a sudden return to maximum.

Random: Capable of selecting any value within its range at regular intervals.

Noise: Completely arbitrary data points presented nearly simultaneously at unpredictable intervals. "Rate" does not apply to the Noise selection, but Amp controls the level.

Sample: Short for "Sample & Hold". This option reveals the Input parameter, where you can choose any one of the 27 mod sources. When selected, the LFO looks to the input source for its next value at regular intervals. It's more predictable than Random but still can be pretty crazy.

Let's go through all of the other LFO group parameters now:

Speed: Controls the rate of the LFO waveform, except for Noise.

Input: Only available when LFO Wave is set to Sample, with 27 source options.

Retrig: Allows you to specify the starting point of the LFO for the first five waveforms. A setting of 0 means the LFO will start from the beginning of its cycle, while values of 1 or higher will cause the LFO to be triggered from somewhere in the middle of its cycle.

Amp: Determines the overall LFO output that will be sent to the target.

Press the Page 2 button for the final two parameters in the LFO group:

Lag: The LFO Lag parameter can be used to round off the edges of the LFO waveform. This will be most noticeable when using a Square wave LFO and least noticeable when using the Triangle waveform. It is actually not related to the Lag processor and has a predetermined impact. If a greater Lag effect is desired, the LFO can be routed through the Lag processor directly and then routed to the VCO through the Modulation Page.

Retrig Mode: Selects single or multiple trigger response. "Off" provides the LFO with a "freerun" setting.

Remember to use the Tabs to select LFOs 1-5, and the the mod buttons will bring their parameters into focus in the Modulation Page so you can create one or more mod routes.

4.3   The Effects

To add one of the effects to a Matrix-12 V sound in Main mode, click the FX button in the nameplate:
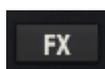
The bottom panel will reveal two drop-down menus, either of which will show you the available Effects choices when you click the words "SELECT EFFECT":
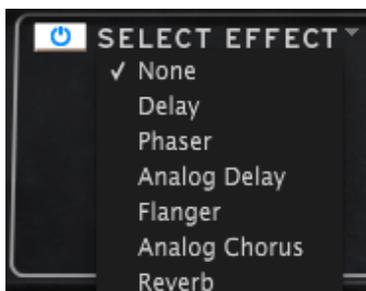
Make a selection and its FX module will appear in the bottom panel, either on the left half or the right, depending on which one you chose.

We'll take each of the Effect menu choices one at a time, starting from the top.

The first one is labeled **None**. Selecting this option will close the current FX module and the Effect menu, removing that FX device from the signal path.

But **None** isn't the option to use when you want to hear the sound without the FX unit temporarily. You'll notice every FX device also has a bypass button in its upper-left hand corner, which is the quickest way to remove the FX module from the mix.

There is also a Wet/Dry slider which controls the percentage of the original signal that passes through to the output. Moving this all the way to Dry will remove that effect from the output.

One more thing: All FX parameters are MIDI-assignable, which means they "learn" the controllers on your external USB MIDI device.

What follows is a description of each available FX device.

4.3.1   Delay

Sync: Locks the delay to the current tempo of the DAW and/or the rate of the Phaser. (These are the only two FX modules with a Sync button.)

Link: Makes the delay mono, after which the top row of Time and Feedback controls are used to adjust the effect

Time: A clockwise turn increases the delay time; a turn in the opposite direction shortens it

Feedback: Adjusts the Feedback amount. Larger values cause the delay to be heard longer

Pingpong: Hard-pans the effected signals so they "bounce" from left to right

Damping: Higher settings will roll off the high-frequency content of the delayed signal more quickly

Wet/Dry: Sets the ratio between the original (dry) and modified (wet) signals

This digital delay repeats the input signal and creates an "echo", giving it more space and depth. The **Time** knob offers a range of settings from 4.5 milliseconds to 1 full second.

4.3.2 Dual-stage Phaser

The following parameters are available independently for both Stage 1 and Stage 2:

Rate: Sets the speed of the phaser

Depth: Sets the depth of the phaser activity

Feedback: Controls the amount of phaser resonance

Shape: Sine vs. noise

Stages 1 and 2 share these parameters:

Sync: Locks both stages to the current tempo of the DAW and/or the rate of the delay. (These are the only two FX modules with a Sync button.)

Mode: "Single" means Stage 1 is on the left side and Stage 2 is on the right. With "Dual" both stages process both sides; the Phaser output is mono.

Mix: Alters wet/dry mix

The phase shifter was one of the most popular effects used with electric instruments in the 1970s. It works by splitting the incoming signal, changing the phase of one side, and recombining it with the unaffected signal. This creates a notch-comb filter which can be swept through the frequency spectrum, causing the signature "whooshing" sound of the phase shifter.

The sweep is caused when the phase of the affected half is modulated by an oscillator, with the frequency determined by the **Rate** control. The **Depth** knob sets the amplitude for the action of the filtering, while **Feedback** amplifies certain harmonics.

4.3.3   Analog Delay

Delay: Sets the delay time. (The delay is mono for authenticity.)

FB Tone: Changes feedback filtering

FB Amount: Adjusts Feedback

LFO Depth: Sets the amount of delay time modulation (which follows a sine wave)

LFO rate: Controls the modulation rate

Mix: Alters wet/dry mix

This analog-style delay reproduces the sound of the old solid state units that used analog bucket brigade circuits. The **Delay** knob allows you to select a time between 4.5ms and 1000ms. The **Feedback Amount** knob sets the feedback level. The **Tone** knob controls a feedback filtering effect: low-pass to the left, high-pass to the right. You can set the delay modulation by changing the **LFO rate** and **LFO depth** values. The **Wet/Dry** slider sets the ratio between the original and modified signals.

4.3.4   Flanger

Delay: Sets the delay time, which changes the harmonic content.

Depth: Sets the modulation depth. This is set to "max out" at less than 100% to limit runaway feedback.

Rate: Sets the modulation rate for the delay time.

Feedback: Adds positive or negative feedback for a harsher or "ringing" sound. Double-click this control or set it to the 12:00 position for zero feedback.

A flanging effect is created by combining two identical signals, delaying one of the signals by a small amount, and then modulating the delay time. The recombined output produces a sound that sweeps up through the harmonics of the original signal and back down.

Flanging can create both subtle and extreme effects, depending on the **Rate** and **Depth** of the modulation. With higher Depth settings you will begin to hear changes to the pitch of the sound. This is how the circuits in an analog flanger work, and we have taken care to recreate these conditions.

4.3.5   Analog Chorus

Type: Select one of three chorus types

Stereo Width: Controls the width of the stereo effect

Stereo Rate: Sets the speed of the stereo effect

Chorus Rate: Adjusts the speed of the chorus

Amount: Controls the depth of the chorus

Delay: Sets the amount of delay applied to the input signal

Wet/Dry: Changes the balance between the input signal and processed signal

A chorus module recreates the sound of multiple takes of an instrument being combined in a mix. Even with the best of performances there are always differences in tuning or timing, and when taken to either extreme the results can range from slow and lush to a wobbling, quick vibrato.

The speed of the effect is set by the **Chorus Rate** knob, while its depth and width are controlled by the **Amount** and **Delay** knobs, respectively. The resulting "frequency blur" is different for the left and right halves of the signal, which allows us to derive a stereo signal from a mono signal. The difference between the two halves then can be set with the **Stereo width,** with the speed of the left-right rotation under the control of the **Stereo rate** knob.

The **Wet/Dry** control sets the ratio between the input signal and the treated signal, while the **Type** switch selects between three different chorus models: simple, medium, and complex.

4.3.6 Reverb

Gain: Sets the input level of the signal going into the Reverb

HF Damping: Rolls off high frequencies gradually in the reverb

Brightness: Controls the high frequency content of the input signal

Diffusion: Masks or reveals individual reflections within the reverb

Decay Time: Determines the length of time the reverb takes to fade out

Feedback: Adjusts the number of reiterations of the reverb effect that occur after the pre-delay time

Pre-Delay: Controls the onset of the initial burst of reverb by delaying it for up to 2 seconds

Wet/Dry: Allows you to decide the balance between the input signal (dry) and the effect (wet).

In summary, we think you'll find that these six FX modules provide some extremely useful effects for building great analog synth sounds. Have fun!

4.4   The Modulation Page

You'll notice many items in the Parameter window have buttons containing their names. These are found just beneath the parameter value.

When one of these buttons is clicked its name will appear in the top line of the Modulation Page, and any existing modulation routes and amounts will be automatically displayed in the next two lines:

In the above picture the VCO 1 Frequency mod button has been selected, bringing that parameter into focus in the Modulation Page. We can see that both LFO 1 and Tracking Generator 1 are selected as modulation sources, although the mod level for the Track 1 mod is set to 0 so it is having no impact on VCO 1.

These assignments and values can be altered or removed, and more can be added if mod routings are available. In the example above there are four unused mod routes.

As for the "buttonless" parameters, the ones that can't be used as modulation destinations can still be edited by their respective controls, and also can be controlled by incoming MIDI data.

4.4.1   The MOD list

Another way to view the modulation routings is to press the MOD button in the nameplate. This will change the bottom panel to display the Mod list:


Modulation routings 1-20

Note that there are two tabs on the far right side of the Mod list: 1-20 and 21-40. This means you have 40 modulation routings available for each Single mode preset, and also means that you will probably never run out of them.
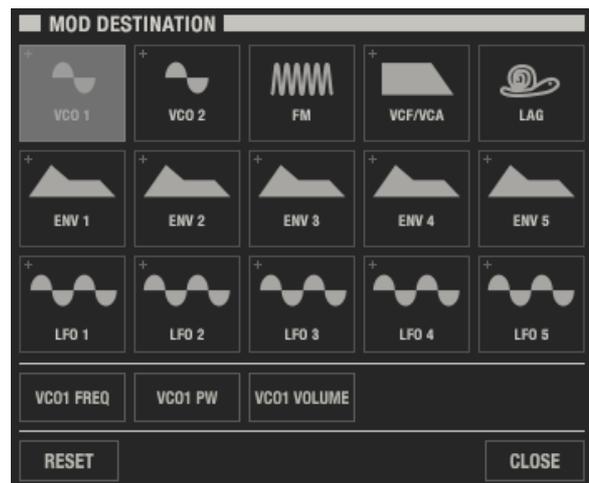
Modulation routings 21-40

The controls are arranged in groups of three: Source, Amount and Destination. There are a lot of Sources and Destinations, so when you click one of them a window will open to show you the options:
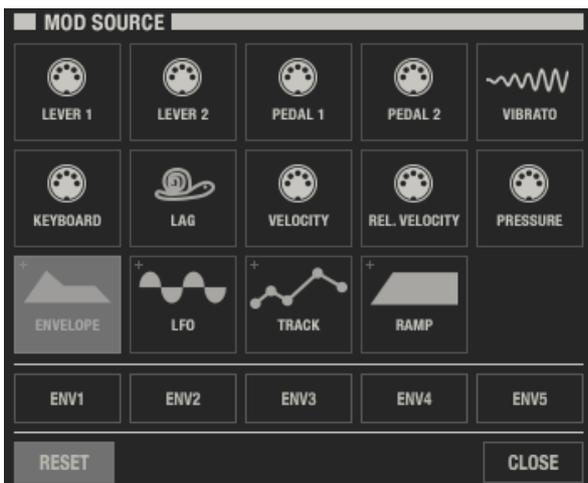
The Mod Source list

The Mod Destination list

Some sources like Envelope have more than one option. So do some destinations, such as VCO 1. For these you need to be more specific about which source or destination to use, so you'll have to click a second item, like ENV1 or VCO1 FREQ. Once the selection is made the Mod list will close.

After making your selections, use the associated SRC knob at the bottom of the Modulation Page to dial in the modulation level. Or if the Mod page is open on the bottom panel, you can use the appropriate AMT knob to set the value instead.

4.4.1.1 Resetting part of a mod

When you're inside one of the lists pictured above you can clear the selection you've made by clicking the RESET button. This will remove that selection from the mod routing and close the list. The name of the item you cleared will be replaced by three dashes in the display.

To reset the modulation amount, not the source or destination, double-click one of the following:

- the value field on the Modulation Page
- the related AMT knob in the bottom panel's Mod field (pictured in the next section)
- the SRC knob immediately below the value on the Modulation Page.

4.4.1.2 Clearing a mod

If you decide you want to remove the entire modulation route (source, destination and amount), there's a very quick way to do that:

- Find the mod in the bottom panel's Mod display
- Click the CLEAR button on the right side of that mod, as pictured below.

The bottom panel Clear button

The SRC and DEST names will be replaced by three dashes, and the AMT value will return to 0 (12:00).

4.4.1.3 Quantizing a mod

The Quantize feature can be used to force an otherwise smooth modulation transition to happen in **steps**. Imagine: a sweeping LFO, an envelope, a Lever or any other mod source can be made to move in discreet increments rather than "gliding" through the values.

Here's a cool example: When a mod to the VCO frequency or the filter cutoff is quantized, the modulation output can be heard in semi-tones. You may find an immediate use for that application by setting up a sort of "mini-sequence" that will transpose itself when you play the keys. This is interactivity at its best, and one gift among many from us to you.

To quantize a mod, select the Modulation Page and then toggle the button with the letter 'Q' next to the clear button.

4.5   Page 2

The Page 2 nameplate button reveals important settings such as VCO keyboard tracking, LFO trigger modes and pitch bend/vibrato assignments for the levers, to name a few features.

Most of what Page 2 offers are on/off parameters, with buttons that light when activated. When a parameter is "on" its name will be underlined in the display; if it is not underlined the parameter is "off".

The Page 2 window in the bottom panel

As with the Parameter window, blue bars identify parameter groups and tabs are used to access parameter sets such as LFO 1-5 and Ramp 1-4. None of the Page 2 parameters are available as destinations in the Modulation Page, so they do not have additional buttons beneath their names.

All Page 2 parameters are MIDI-assignable.

Page 2 will remain in the bottom panel until one of the other nameplate buttons is selected (KBD, FX, or MOD), or until one of the other pages is selected (Voices or Program Chain).

4.6   Single mode: the Voices page

4.6.1   First of all: What's a Voice?

There are two uses of the word "voice" related to Matrix-12 V. To keep them straight, we capitalize one ("Voice") and don't capitalize the other ("voice"). Here's a definition of each:

voice: A unit of polyphony; i.e., the number of notes that can be played simultaneously.

> Example: "The Xpander is a six-voice synthesizer."

Voice: The collection of synthesizer parameters that make up a preset in Single mode, not counting the FX settings.

> Example: "This Multi was constructed by layering two Voices."

4.6.2 The Voice page

The Voices page has fewer features in Single mode than it does in Multi mode, which is to be expected since you can only work with one Voice at a time. But it still is capable of enhancing that Voice quite a bit! Options include independent panning and transposition for each of the 12 voices, the ability to have two mono voices stacked on the right side of the keyboard with the other 10 voices layered and detuned on the left side, etc.

Plus, the Voices page houses the VIB parameter group. So there are a lot of features to discuss here.

So now let's take a look at the Parameter groups on the Voices page and see what they can do in Single mode.
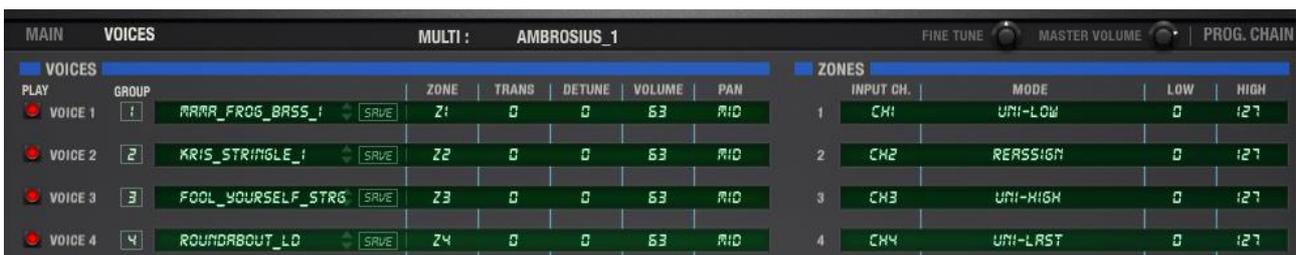
The first step is to select a preset that's a Single, not a Multi. If you're not sure, then click the word "Voices" to access the Voices page. You should see a page that looks like this:
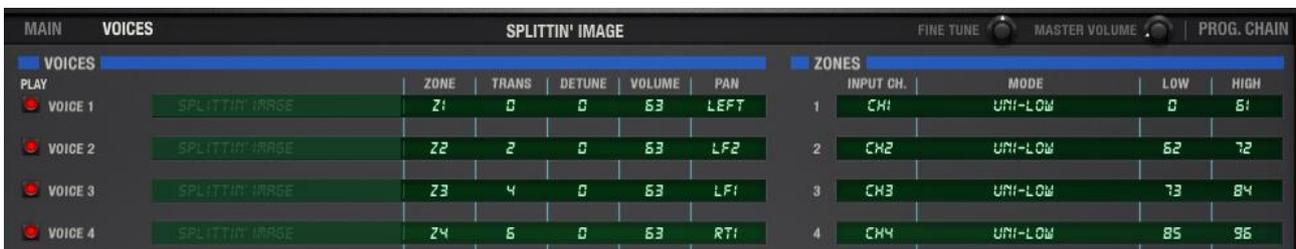
A portion of the Voices page in Single mode

…with the name of the Single preset in the Page bar and also "greened out" in the display column to the left.

But if what you see looks like this instead:

An example of the Voices page in Multi mode

…then Matrix-12 V is in Multi mode, not Single mode. Select one of the Single presets from the Factory banks so what we're describing and what you're seeing are the same thing.

When you're ready, see the next section. We'll start with the parameter group labeled "Voices" and work our way across the screen.

4.6.3　Voices
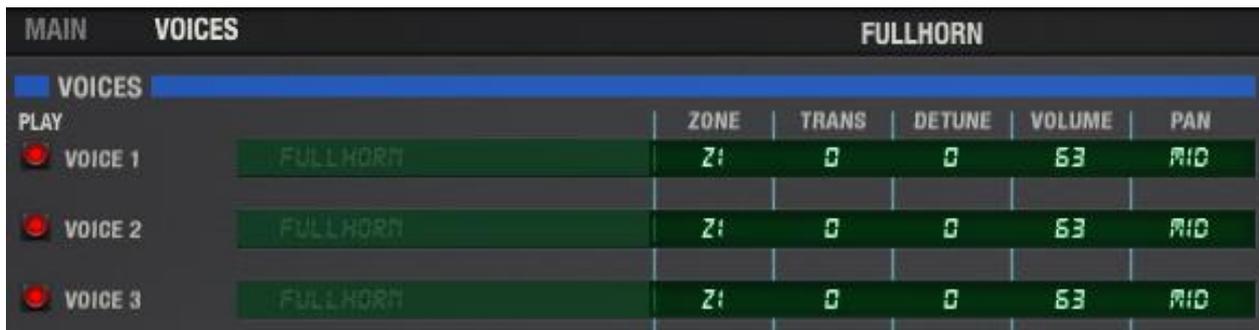
The Voices parameter group in Single mode

Note: unless otherwise indicated, click and drag to edit a parameter value; double-click to reset.

Play: Normally on (red) by default, these turn green when a particular voice has been triggered. You can also click them to disable a voice so it will not play.

Zone: This field is used to assign a voice to one of the six Zones, where its note range, keyboard mode, and other settings can be defined. You can also select one of the 16 MIDI channels for a voice and bypass the Zone parameters.

Transpose: Allows you to change the pitch of a voice in chromatic increments within a range of + 2/-3 octaves. You could use this for semi-random octave jumps, for example.

Detune: With this you could put some voices slightly flat and others slightly sharp, or make them way out of tune. The range is -31 to +31, and in combination with the Transpose parameter you could set the pitch of a voice to pretty much whatever you want.

Volume: This parameter sets different output levels for the voices, which could be handy when layering transposed voices so you can get the balance just right.

Pan: Choose one of seven stereo positions per voice to give the Voice a bit of motion. This could be extra cool with percussion or a sound effect. Click the field and a window will open to display the options.
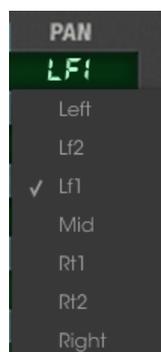
The per-voice pan positions

4.6.4 Zones

*The Zones parameter group in Single mode*

The first thing to notice are the Zone numbers in the left-most column (1-6), indicating which Zone will be edited by the parameters to the right. The per-voice Zone assignments themselves are handled in the Voices parameter group over to the left.

Here's what's possible with the Zone parameter group:

Input Ch.: Assign the Zone to MIDI Channel 1-16 or Omni.

Mode: Determines how voices will be triggered. The first three modes are **polyphonic** (Rotate, Reassign, Reset), and the last three are **monophonic** (Uni-Low, Uni-High, Uni-Last).

Click the field and a window will open with the choices. A check mark indicates the current mode:
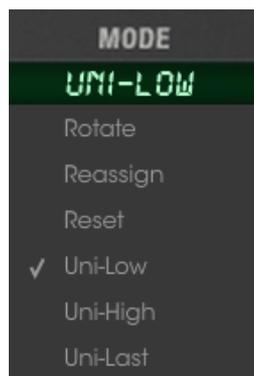
Note assignment modes

Below is a description of how each mode behaves:

Rotate: Each new note will play the voices assigned to that Zone one after the other, and then they'll loop (1-2-3, 1-2-3, etc.).

Reassign: Checks each new note to see if any voice in that Zone is already playing it. If so, the note will be retriggered using the same voice. If not, it'll grab the next higher voice if it's available.

Reset: Plays the voices assigned to the Zone in numerical order, always starting with the first voice. If consecutive staccato notes are played it'll use the first voice each time; if two legato notes are played it will use the first voice, then the second, and so on. But when either of those notes ends it'll start over with the lowest-numbered available voice.

Uni-Low: The voice(s) will be triggered by the lowest note played in the Zone. If the note is held and a higher note is played, the voice(s) will stay on the current note; if a lower note is played, the voice(s) will change to that note. Also known as "low note priority".

Uni-High: The voice(s) will be triggered by the highest note played in the Zone. If the note is held and a lower note is played, the voice(s) will stay on the current note; if a higher note is played, the voice(s) will change to that note. Also known as "high note priority".

Uni-Last: A combination of high and low note priority: The voice(s) will be triggered by the last note played in the Zone, whether it is above or below the previous note.

V. steal: Voice stealing helps maximize the voice allocation within a Zone. For example, if five voices have been assigned to a Zone and all five notes are playing, what happens if a sixth note is played?

Ordinarily, nothing: the sixth note is ignored. But with V. steal enabled, the first note that was played is "stolen" and made available to the sixth note. So on the plus side, V. steal means every note played within the Zone will be heard. However, the downside is that if the Voice has a long release time, then after a note is released it is at risk of being cut off abruptly if a new note arrives.

Low: Click and drag the value to set the lower MIDI note range for the Zone.

High: Click and drag the value to set the higher MIDI note range for the Zone.

4.6.5   Vibrato (VIB)

The VIB section provides a modulation source that is sort of like having an "LFO #6". However, it doesn't have as many options available as LFOs 1-5.

Here are the LFO features that are missing from VIB:

- One less waveform: Sample & Hold is not available
- No Retrig setting: the VIB waveform always starts from the beginning when triggered
- Single trigger only: no Multiple trigger setting or "freerun" capability
- No Mod buttons: VIB parameters are not available as destinations on the Modulation Page.

Now let's focus on the positive! Here's what the VIB section does provide:

Speed: A rate control with a range of 0 to 63; click and drag to edit the value. A simple mod routing Is provided.

Source: Modulate VIB speed from one of two sources: Mod Wheel or Pedal. Click the field to reveal the choices:
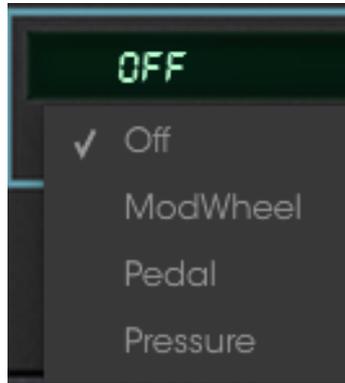
Mod sources for VIB rate

Mod Amt: The amount of control the Source will have on the VIB speed. Click and drag to edit the value. The modulation can be positive or negative, with a range +/- 63.

Wave: Six options: Triangle, Up/Down Saw, Square, Random and Noise (no Sample & Hold). Click the field to reveal the choices:

VIB waveform options

Lag: Click the button to use the Lag generator as an additional mod for the VIB effect.

Amp: The overall output of the VIB section. A simple mod is provided

Source: Modulate VIB output from one of two sources: Mod Wheel or Pedal

Mod sources for VIB output

Mod Amt: The amount of control the Source will have on the VIB output. Click and drag to edit the value. The modulation can be positive or negative, with a range +/- 63.

One more thing: as with almost every Matrix-12 V parameter, everything in the VIB parameter group is MIDI-assignable.

5   MULTI MODE

So you want to play synth bass with your left hand and some funky riffs with the other? How about multi-timbral sequences, or layering two Voices to create The Pad To End All Pads?

Whatever your musical requirements may be, Multi mode is the place to go when your plans call for Matrix-12 V to have two or more Single mode presets active at the same time.

5.1   How to switch modes

There are two main modes in Matrix-12 V: Single and Multi. However, what you see in the Page bar are the words "Main" and "Voices", so those aren't what you use to switch between the modes.

Here's why: even though the Main page and the Voices page look pretty much the same in each mode, they actually do different things depending on the mode of the selected preset.

However, to enter Single or Multi mode you must select a preset that is known to be one or the other type. And once you are in Multi mode, for example, selecting a Single mode preset will take you back to Single mode.

5.2   Single mode vs. Multi mode

Visually Single mode and Multi mode seem very similar, but functionally they are quite different. The focus of Single mode is the Main page, where practically every synthesizer parameter known to man resides, for the express purpose of creating a Voice. The Voice can stand on its own or become the basis around which a Multi is constructed.

The focus of Multi mode, on the other hand, is the Voices page. As the plural of "Voice", the name "Voices" implies that Multi mode enables you to access more than one "Voice" at a time.

The potentially confusing part is that there is a Main page in Multi mode and a Voices page in Single mode. The advantages to this system, though, is that there are certain parameters on both pages that are useful in either mode, whether you're dealing with a Single preset or a Multi.

But how do you tell which mode Matrix-12 V is in without loading a new preset that you know is one or the other?

Here's the fastest way we've found: If you're on the Main page, click Voices in the Page bar and look at the name field for the voices. If the voice names are "greened out" and they're all the same, Matrix-12 V is in Single mode.

The Voices page in Single mode

In Multi mode the Voices page has several more options, so it should be easy to tell in that case.

But eventually you'll know at a glance that Matrix-12 V is in Multi mode even when you're on the Main page. The only difference is in the Page bar:

The Main page view of the Page bar in Multi mode

Note the word "MULTI" and the name of a Multi in the middle of the Page bar, plus the term "G1" and the name of a Single preset surrounded by blue arrows. We'll describe those things later, but those factors indicate that Matrix-12 V is in Multi mode.

But in Single mode the Page bar only has the name of a Single preset in the middle, like so:

The Main page view of the Page bar in Single mode

All other parameters on the Main page are the same in either mode, so it is always possible to switch to the Main page and edit the parameters of a Voice.

And since the focus of Multi mode is the Voices page, that's where we'll begin.

5.3   Multi mode: the Voices page

When you load a Multi preset it will default to the Voices page. The Zones and VIB sections are identical to the ones you see on the Voices page in Single mode; the only noticeable difference on the Voices page in Multi mode are on the left side of the window:

An example of the Voices page in Multi mode

Notice the Group column and the green "Save" button, neither of which are there in Single mode. All of the other features are the same in each mode, except for the ability to select a new Single preset for each Voice: Multi mode has this and Single mode does not. We'll go over those features carefully in this section of the manual.

The Voices page is divided into three parameter groups (Voices, Zones, VIB), with each section offset by a thick blue line. Let's focus on each parameter group one at a time.

5.3.1   Voices

In Multi mode each of the Voices in Matrix-12 V can be assigned its own Single preset. But it doesn't stop there: each Voice can be transposed, detuned, and panned independently, and then placed into a Group and Zone which may or may not contain any of the other Voices.

Most of the features in the following screen shot are already familiar:

The Voices parameter group in Multi mode

The focus here will be mainly on the features that are unique to Multi mode. Follow the links to the sections where these are covered in greater detail.

Note: unless otherwise indicated, click and drag to edit a parameter value; double-click to reset.

Play: The buttons enable/disable Voices and indicate Voice activity.

Group: Only available in Multi mode, this function links multiple Voices for quicker edits.

Voice name: Show the names of the Single mode Presets used for the Voice(s).

Select arrows: These will move up and down through the presets within the current type selected for the Voice.

**Save**: Provides a quick way to save an edited Voice.

Zone: Used to assign a Voice to one of the six Zones, where its note range, keyboard mode, and other settings can be defined. Or select one of the 16 MIDI channels and bypass the Zone parameters.

Trans: Allows you to change the pitch of a Voice in chromatic increments within a range of + 2/-3 octaves.

Detune: Tune a Voice flat or sharp. The range is -31 to +31, and can be used in combination with the Transpose parameter.

Volume: Use this to adjust the output levels for each Voice.

Pan: Each Voice can be placed at one of seven positions in the stereo field. Click the field and a window will open to display the options.
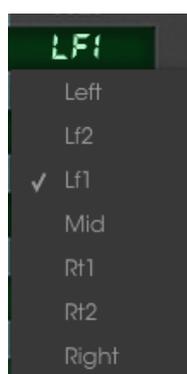

The Voice pan positions

5.3.2   Groups

A group is a combination of Voices that simplifies the process of assigning a new Single preset to multiple Voices that share that preset. For example, if Voices 1 through 4 are assigned to the same group and you select a Brass preset for Voice 1, then Voices 2, 3, and 4 will also change to the Brass preset,

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They retain their individual Transposition and other settings, but they move together when a new preset is selected.

The default Multi has all 12 Voices assigned to the same group, so it is necessary to reassign them. To do this, click on the Group field next to the Voice you want to change:
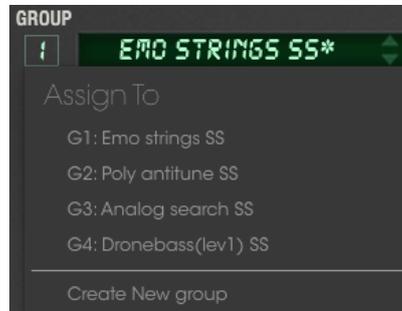

The Group menu in Multi mode

There's only one group available so far, so click "Create New group" at the bottom of the list. After this there will be two groups available and you can assign any one of the Voices to either group.

As soon as one of the Voices has been assigned to group 2 it is possible to select a different Single mode preset for it without affecting the first Voice. In the picture below this has been done, and Voices 1 and 2 are now independent.

However, Voice 3 is still assigned to group 1. You can assign it to group 2 if you like by following the same process.
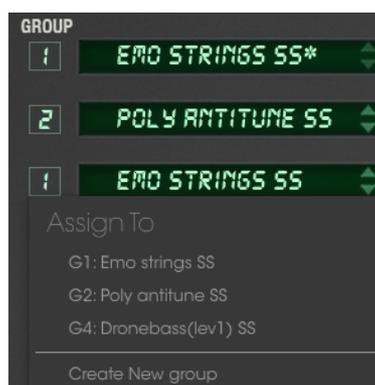
Selecting a different group for Voice 3

Using this same technique repeatedly it is possible to set up several groups and use them to combine Voices into a single polyphonic synth. Let's take another look at this graphic from the previous section and analyze what's happening:

The Voices parameter group in Multi mode

- Voices 1 and 2 are in group 1, so they share the same Single mode preset. But Voice 1 is an octave lower than Voice 2 and slightly quieter. They've been assigned to Zone 1 also.
- Voices 3-7 (5 Voices) are in group 2 and sharing one Single mode preset. But each Voice is slightly detuned from the others and panned to a different location in the stereo field. All five Voices are assigned to Zone 2.
- Voices 8-12 (5 Voices) are in group 3, sharing a Single mode preset. But in addition to different Transpose, Detune, Volume and Pan settings they are also assigned to MIDI channel 3 (not a Zone), so they will respond to incoming MIDI data on that channel without being additionally constrained by the settings of a Zone.

You can create up to 12 groups inside the Multi, so there are lots of other possibilities, too.

5.3.3 Selecting a single

If you'd like to select a new Single mode preset for a Voice inside the Multi, all you have to do is click inside the name field for that Voice.
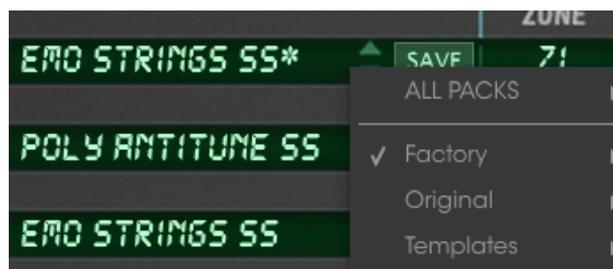
Selecting a new preset for a Voice

After that it's very similar to selecting one from the tool bar: just navigate to the bank and type you want and make a selection from the presets inside that type.

You can also use the small vertical arrows next to the Voice name to select a new preset inside the current bank and type.

5.3.4 Zones

The Zones parameter group allows you to specify MIDI channel response, MIDI note range, monophonic and polyphonic keyboard modes, and enable voice stealing. These settings will be in effect for any Voices that have been assigned to a given Zone.

5.3.5 VIB

The VIB section is identical to the one in Single mode. It provides a modulation source very similar to LFOs 1-5, although it doesn't have as many options available.

Saving an edited Multi

_____

As soon as you change one of the Multi parameters, an asterisk will appear next to the name of the Multi. This is to remind you to save the Multi before selecting another one, or else you will lose the edits you have made.

The process is basically the same as that described previously "Saving an edited preset": click either the single-disc icon to overwrite the original file, or click the double-disc icon to keep the original preset the way it was and save a renamed copy of your latest edits.

If you started with a Factory preset then the "Save Preset As…" window will open automatically no matter which icon you click.

After the window opens the first thing you might want to do is select a new bank or type for the preset. There's a drop-down menu in each field for this purpose:
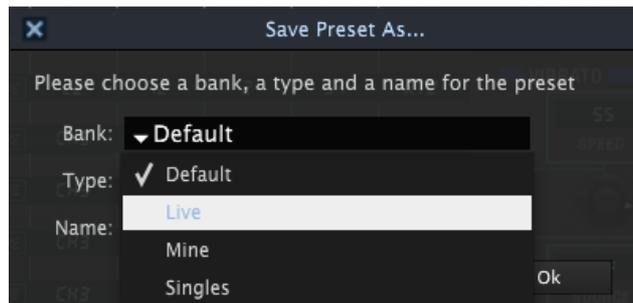
*Drop-down menus to select a new bank or type for the preset*

Or if you would like to create a new bank or type, just highlight the appropriate field and type in a new name.

Rename the preset and click OK to complete the process.

5.3.6   Saving a Voice into a Single Preset

If at some point while programming a Multi you went over to the Main page and made a few tweaks to one of the Voices, an asterisk will appear next to its name on the Voices page. This means you can extract the single assigned on a voice to save it into a single preset.

Just click the small green "SAVE" button next to the Voice name...
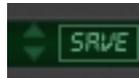
...and the Save window you've seen in Single mode will appear:
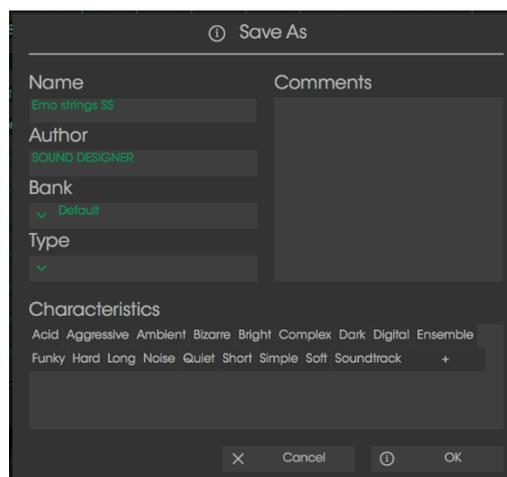
Saving an edited Single mode preset from within a Multi

Once you've selected the Bank and Type and given the preset a new name (if desired), click OK and the preset will be saved. The new preset will show up in the Voice name field and the asterisk will disappear.

After this, be sure to save the Multi itself!

5.3.7 Saving a Multi
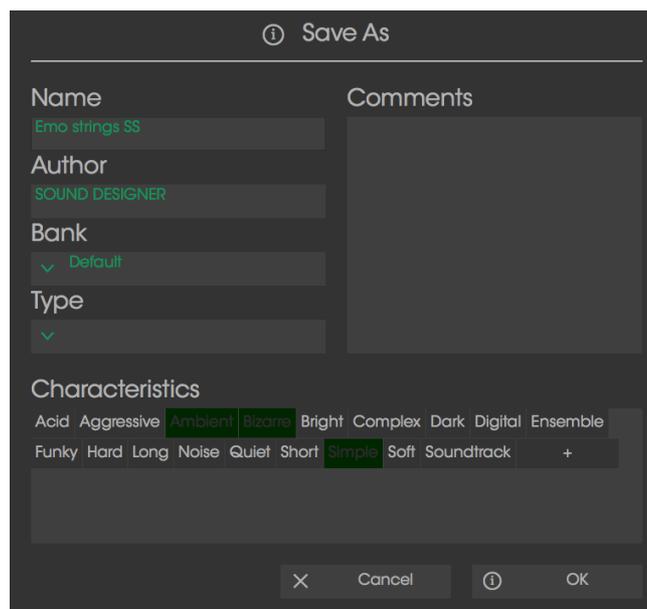
The Multi is embedding all the informations of the voices into itself. It means that modifying a single preset affected to a voice will not impact the multi.

Just open the Save As dialog as usual to define your preset.

Save the edited Single Preset also!

Click the OK button and your work will be preserved.

5.4   Import / Export

The process of exporting and importing banks or presets is fairly straightforward: click the appropriate button and then either 1) navigate to the folder where the items are, or 2) create a folder where they will be.

5.5   Multi Mode: the Main page

The Main page in Multi mode is almost identical to the Main page in Single mode, with only two exceptions. Both exceptions are found in the Page bar, not in the Parameter groups.

The differences are in the very middle of the Page bar:

The center of the Page bar on the Main page in Multi mode

Difference #1: The word "MULTI" is followed by the name of the current Multi preset.

Difference #2: The name of the Voice being edited is between a pair of blue "Previous/Next" arrows. The "G1" stands for "Group 1", the group to which the Voice has been assigned (see section 5.2.2 for information about this).

The cool thing is that those blue arrows will switch the selection from Group 1 to Group 2, etc., which calls up the Voice that has been assigned to that Group. Then the preset can be edited immediately from the Main page, and you didn't have to go back to the Voices page to select the Voice you want to edit. Definitely a time-saver there.

For the full description of each Main page parameter, see chapter 4.

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