USER MANUAL

SEM V

ARTURIA
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1 INTRODUCTION

Arturia would like to thank you for purchasing our synthesizer model: the SEM 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. SEM 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.

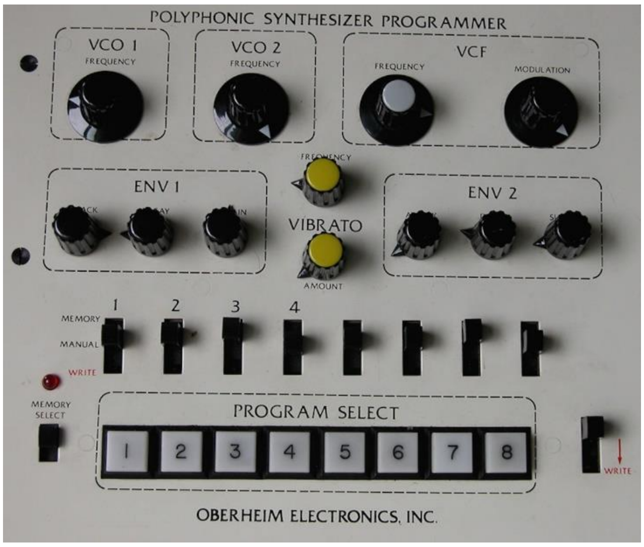
And again, the DS-2a paved the way for sequencers with more memory and polyphonic capabilities, including the Oberheim DSX and products from other manufacturers.

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

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.

1.3 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.

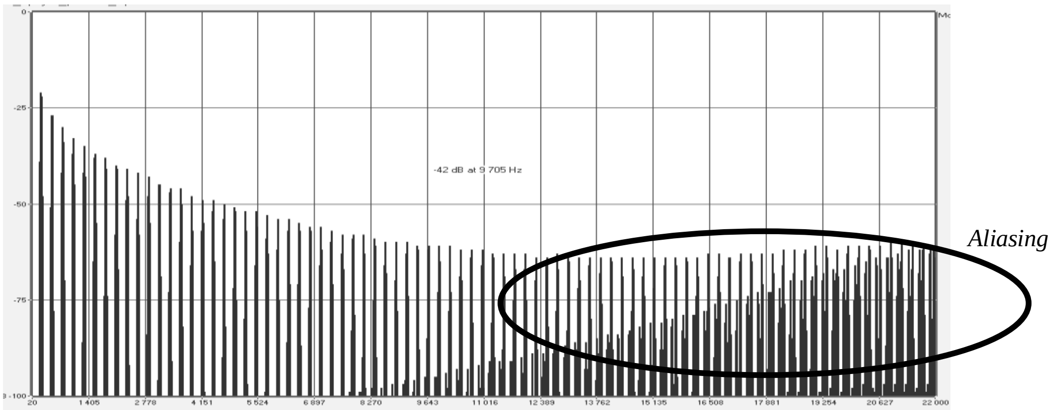
TAE®'s software algorithms result in spot-on emulation of analog hardware. This is why SEM V offers an unparalleled quality of sound, as do all of Arturia's virtual synthesizers.

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

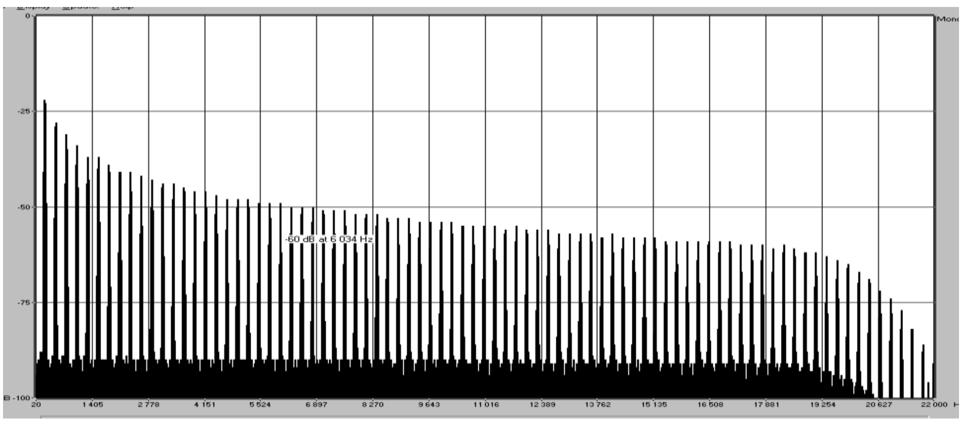
1.4.1 Aliasing-free oscillators

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

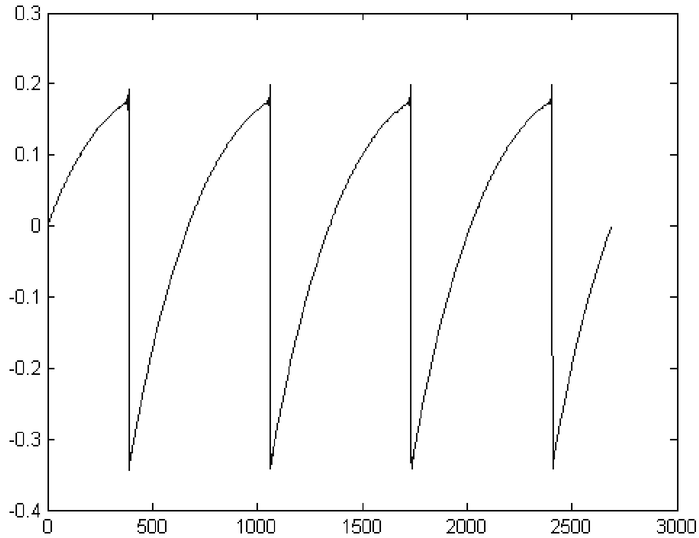


Linear frequency spectrum of an oscillator modeled with TAE®

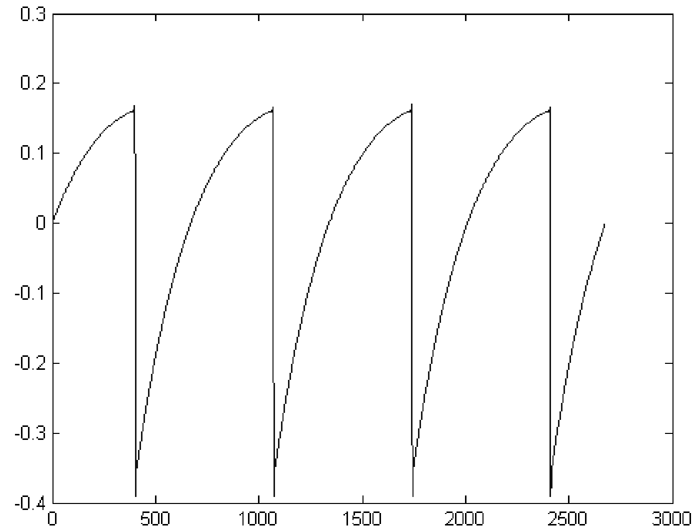
1.4.2 A better reproduction of analog oscillator waveforms

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.

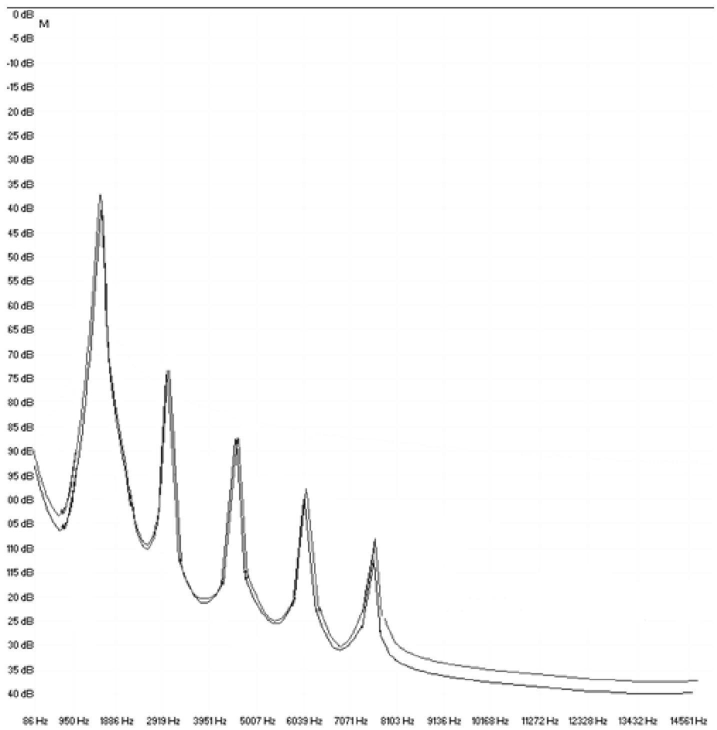
TAE® reproduces the instability of oscillators, resulting in a fatter and "bigger" sound.

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 SEM 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

SEM 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 SEM V as an Audio Units, AAX, VST2 or VST3 instrument.

Once SEM 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

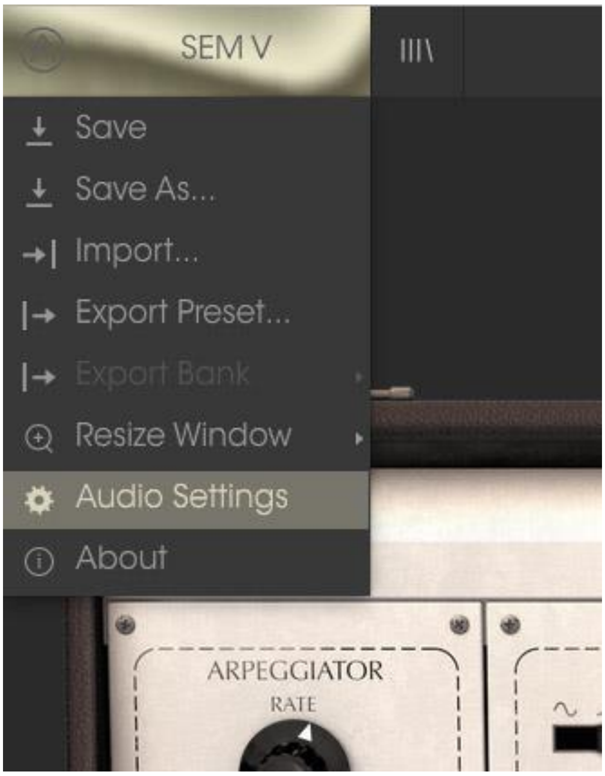
Note: If you don't have an Arturia account yet, you will need to create one. The process is quick, but it does require that you can access your email address during the registration process.

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

2.2 Initial setup

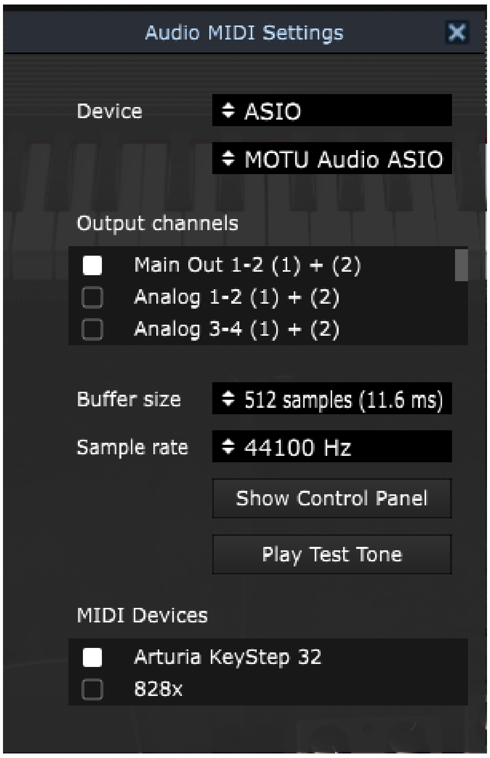
2.2.1 Audio and MIDI settings: Windows

At the top left of the SEM 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.



SEM 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

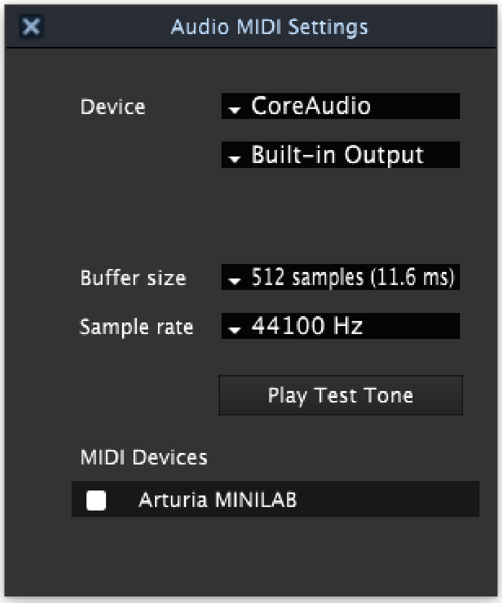
Starting from the top you have the following options:

- **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, SEM 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 SEM V in plug-in mode

SEM 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 SEM V in a DAW project. In standalone mode you can only use one at once.
- You can route SEM 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 SEM 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 SEM 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 SEM V virtual keyboard and Levers

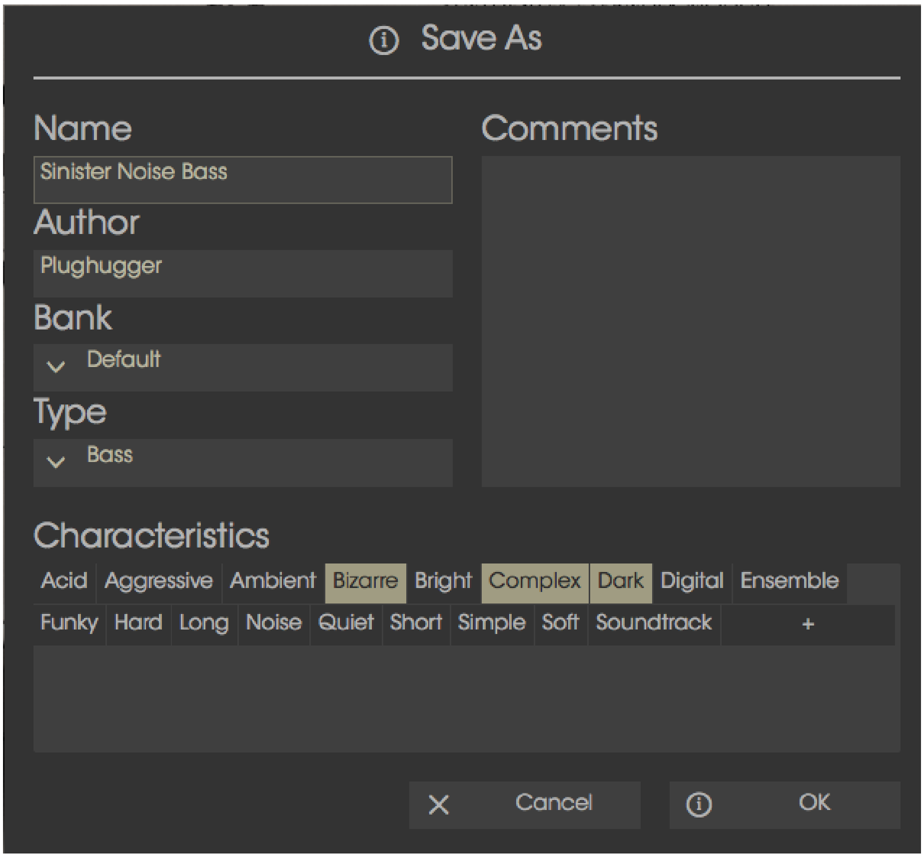
Note also the Levers to the left of the virtual keys. “Mod” lever is available as a modulation source, while spring-loaded “Pitch” lever is usually dedicated to pitch bend purposes. Either can be routed to other destinations on the Modulation Page, though.

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 SEM 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

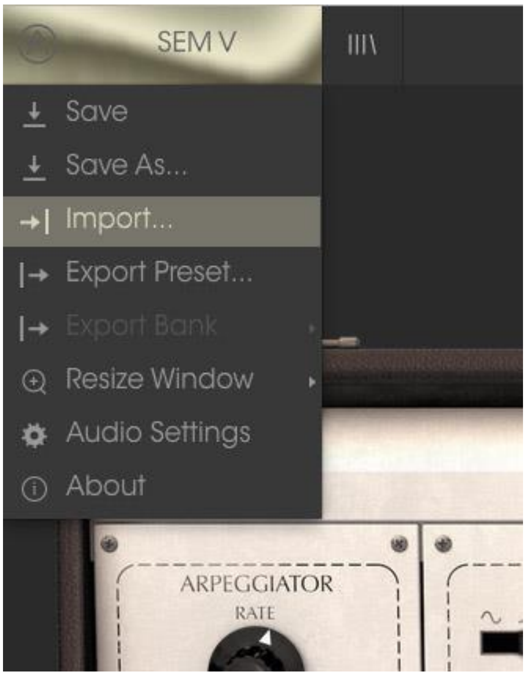
3.2.2 Save Preset As…

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 .semx 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.

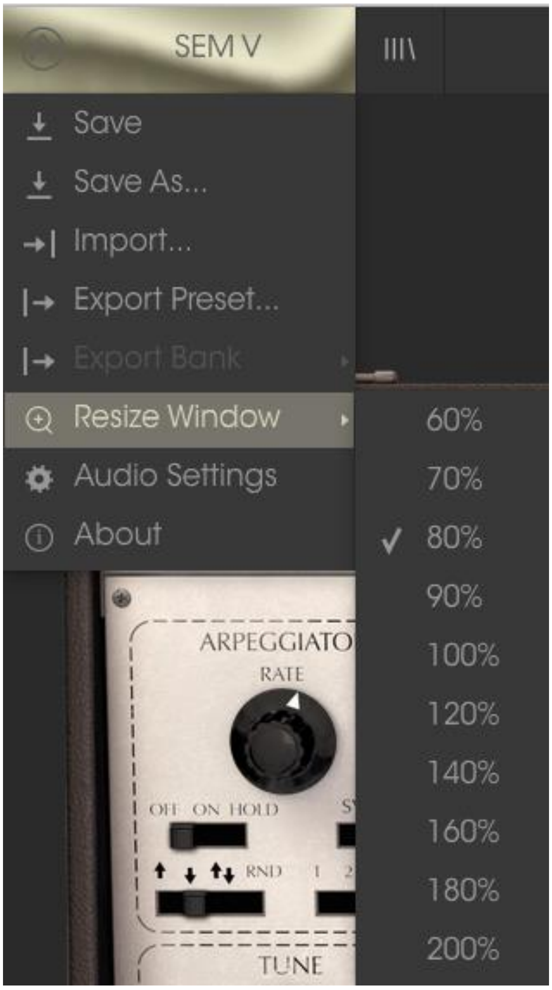


3.2.4 Export preset

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 SEM 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.



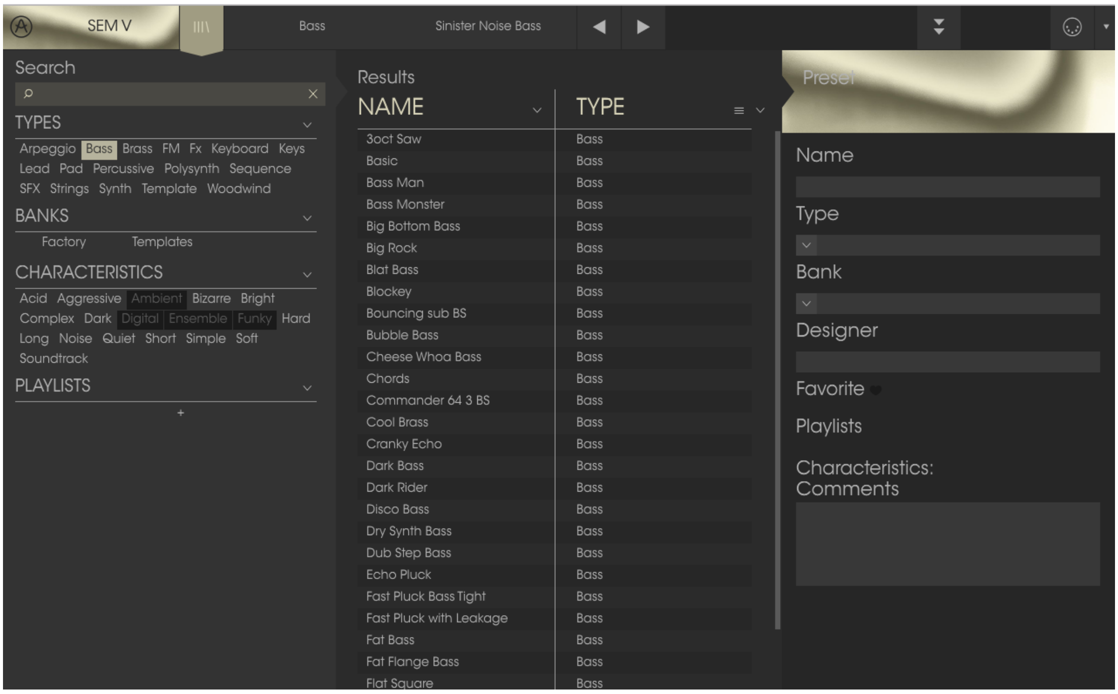
The Resize Window menu

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 Open and Close Advanced section

The Advanced section can be revealed by clicking on the button with the two downward arrows at the right of the toolbar. This lets you access the more advanced features of the instrument like. Click this button once to reveal the advanced section of the instrument and again to hide it. You can also click on the frame of the instrument to open and close it.

3.2.9 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

3.2.9.1 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.

VCF cutoff frequency selected and assigned

3.2.9.2 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.

3.2.9.3 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.

3.2.9.4 Reserved MIDI CC numbers

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

- PitchBend
- Ctrl Mod Wheel (CC #1)
- AfterTouch
- Ctrl Sustain On/Off (CC #64)
- Ctrl All Notes Off (CC #123)
- Ctrl Omni Mode Off (CC #124)
- Ctrl Omni Mode On (CC #125)
- Ctrl Poly Mode Off (CC #126)
- Ctrl Poly Mode On (CC #127)

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

3.2.10 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 SEM V without having to build all the assignments from scratch each time you swap hardware.

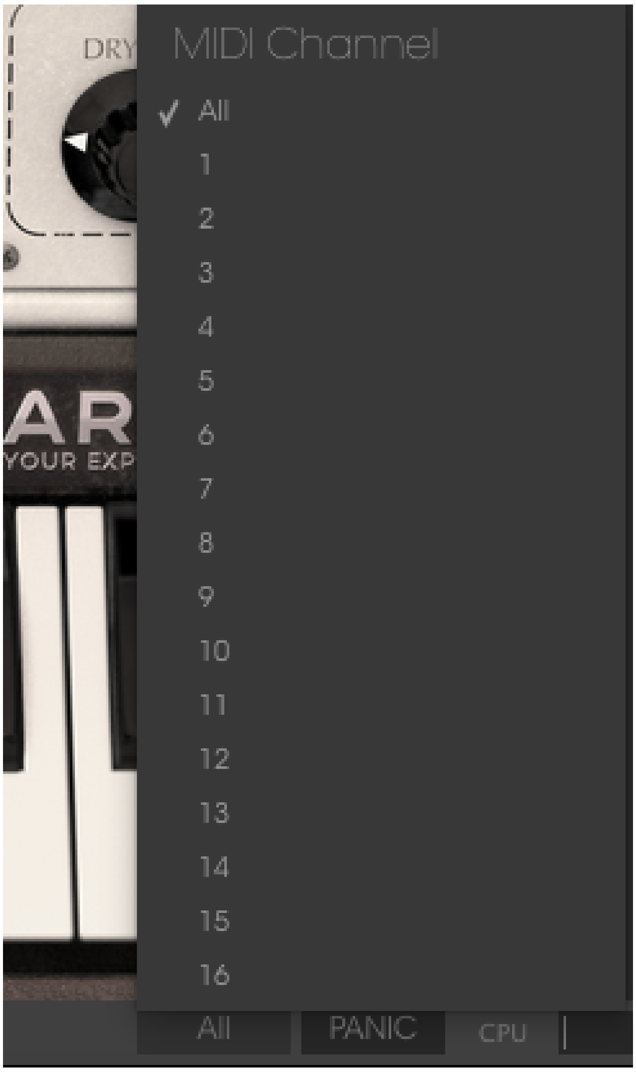
3.2.11 The lower toolbar

3.2.11.1 Current control value

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.

3.2.11.2 Midi Channel Setting

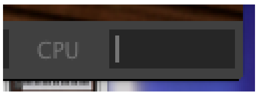
At the right hand side of the lower toolbar are three small windows. The first one on the left indicates the current MIDI Channel setting. Click on it and it will expand to show the full range of values you can select (All, 1-16).



3.2.11.3 Panic button and CPU meter

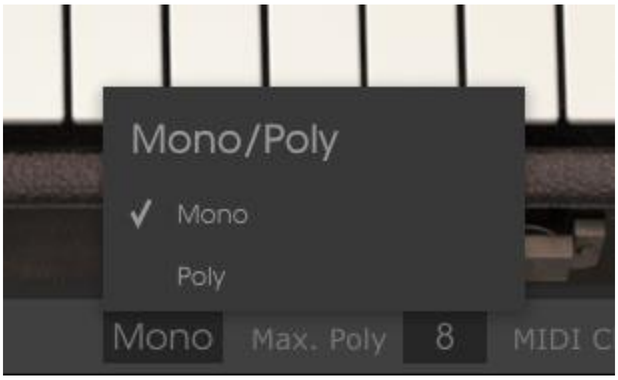
The Panic button can be pressed to reset all MIDI signals in the event of stuck notes or other issues. The Panic button is also MIDI-assignable.

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



3.2.11.4 Mono/Poly selector

This selector activates or deactivates the polyphonic capability of the SEM V, in other words the capability to produce several notes simultaneously.



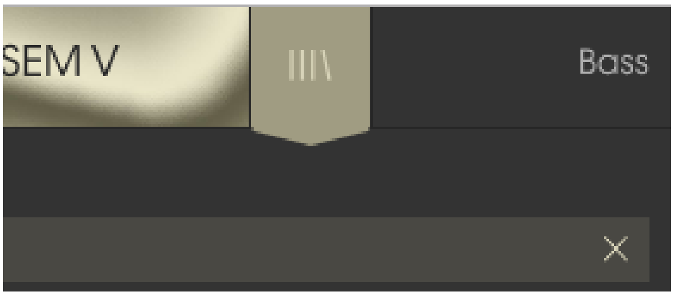
3.2.11.5 Maximum Polyphony

By clicking this button, you will be able to adjust the upper limit for the number of voices played the SEM V. It can be set from 1 to 32. Having lower setting will result in less CPU being used. Setting the number too low can create situation where the voices cut off and create unnatural sustains. The key is to find a balance that you and your computer can both live with.

3.3 The Preset Browser

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

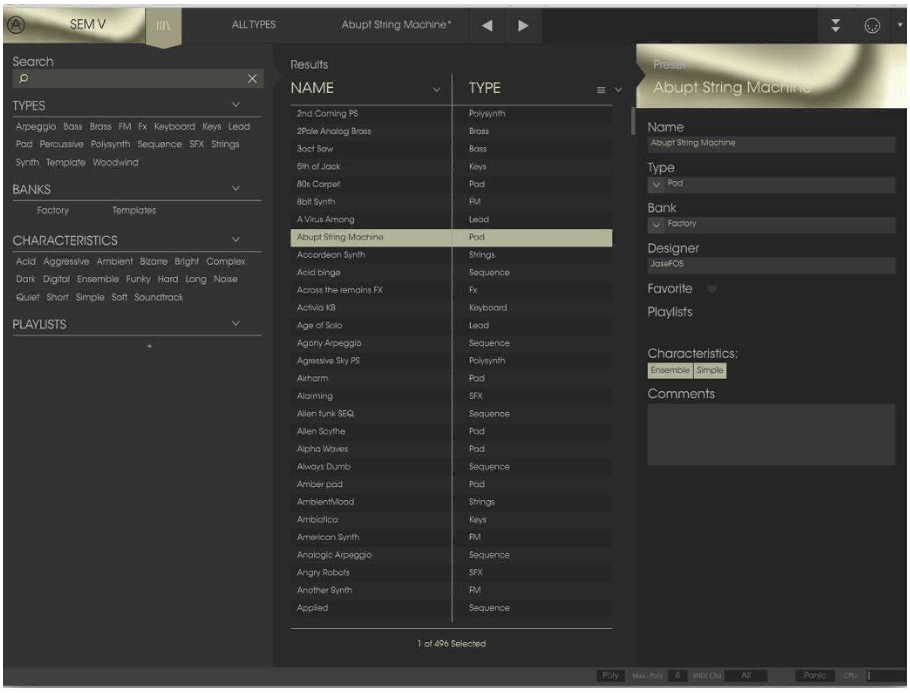
To access the search view, click on the browser button (the icon looks a bit like books on a library shelf).



The Preset Browser button

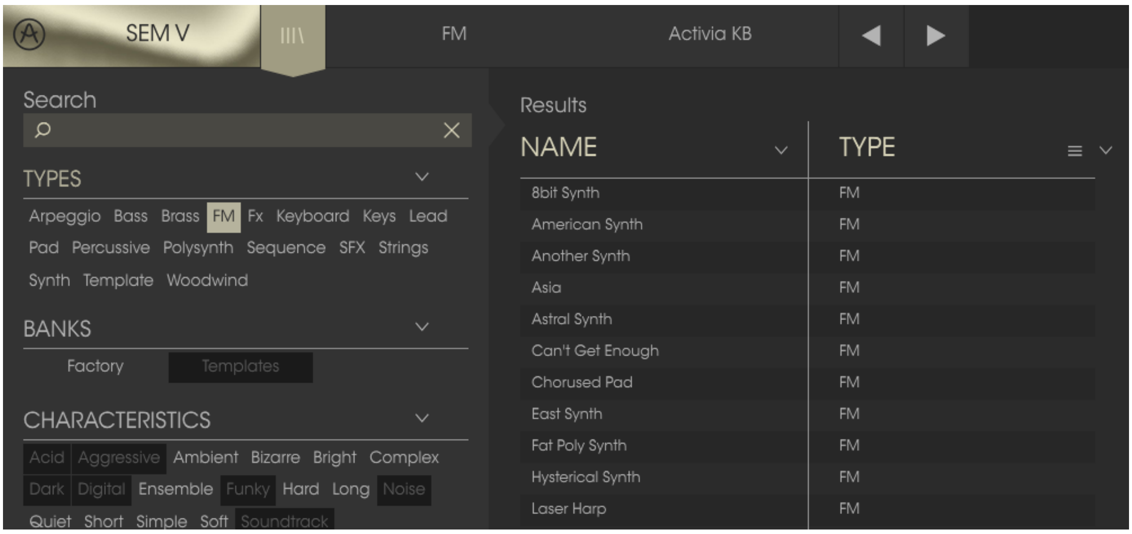
3.3.1 Searching presets

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.

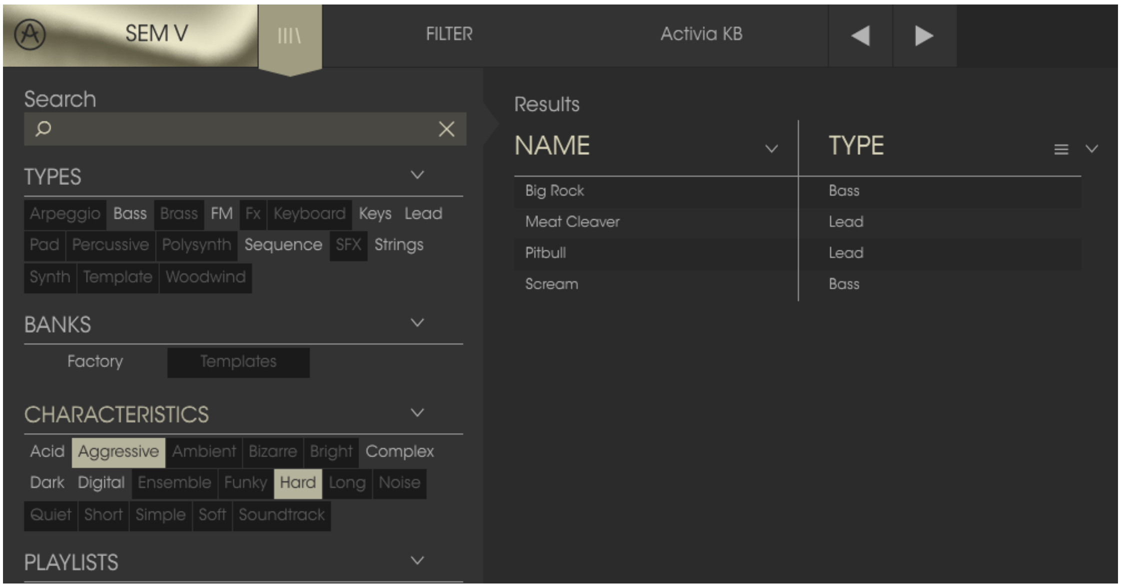


3.3.2 Using tags as a filter

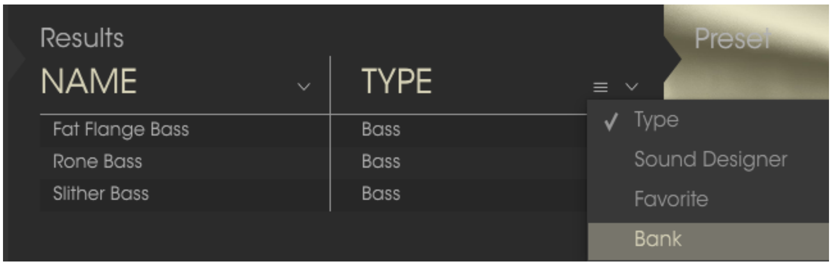
You can also search using different tags. Clicking on a Type field shows only presets that match that tag. The tag fields can be shown or hidden by using the small down arrow buttons in their title fields. Results columns can be sorted by clicking the same arrow button in their own section.



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.

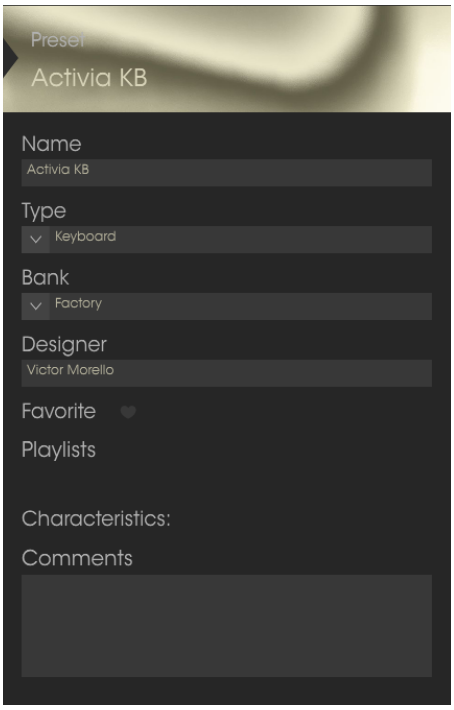


3.3.3 The Preset Info section

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.

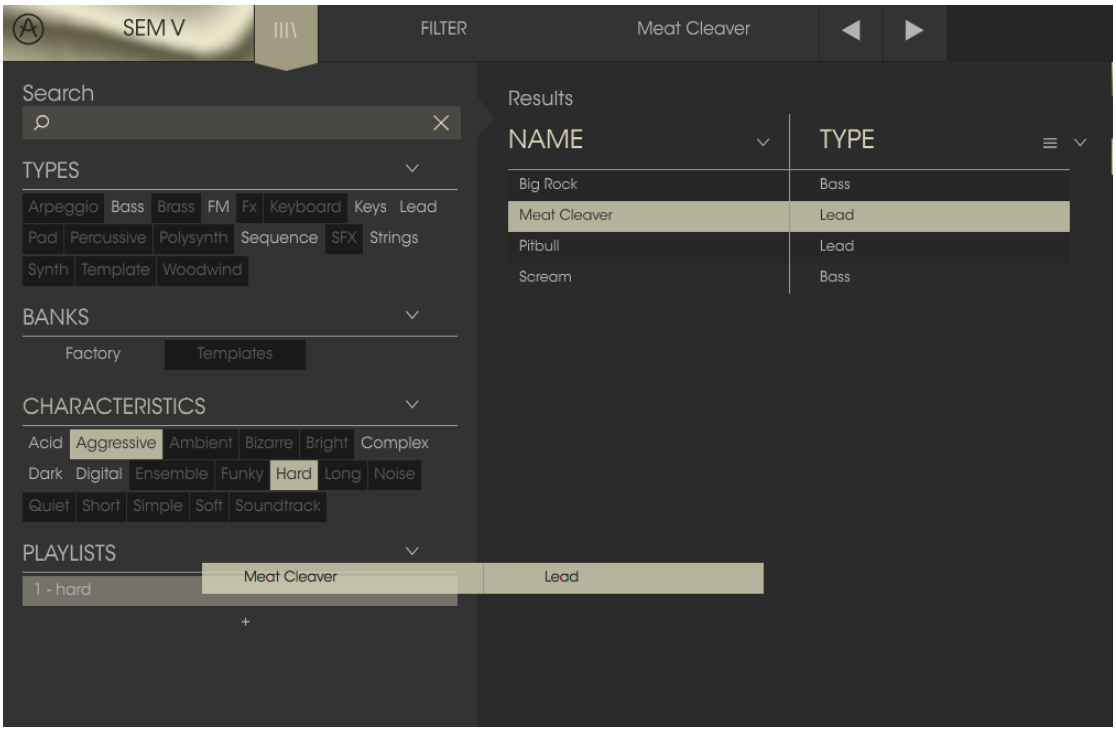
3.3.5.1 Add a playlist

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

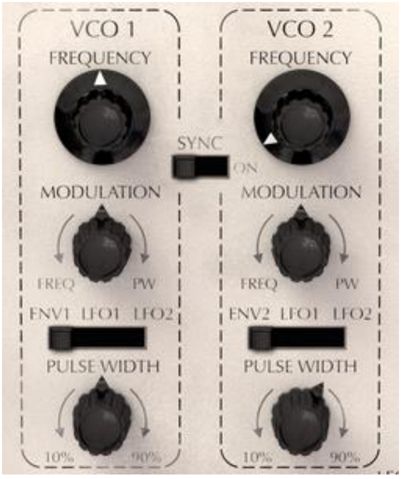
To delete a playlist, click the x directly to the right of the playlist name.

Click the X to delete a playlist.

3.4 Main panel: original SEM features

This part is the core of the sound generation and almost everything that makes the SEM V what it is can be find here. You may notice that SEM stands for Synthesizer Expander Module.

3.4.1 VCO



VCO is an oscillator which generates the foundamental audio signal to be processed. There are two VCO sections in the SEM V, named VCO1 and VCO2.

In this section, you can set the pitch by using the frequency knob, modulation depth for oscillator pitch modulation or pulse width amount. These knobs are found below the frequency knob.

Their waveforms, however, are choosen in the VCF section on the module.

The bottom-most knob in the VCO sections controls the width of the pulse wave. When this knob set at center position, the OSC generates a symmetrical square wave, and when it moves to clockwise or counter clockwise, the waveform progressively changes to an asymmetrical shape and the tone also changes with it.

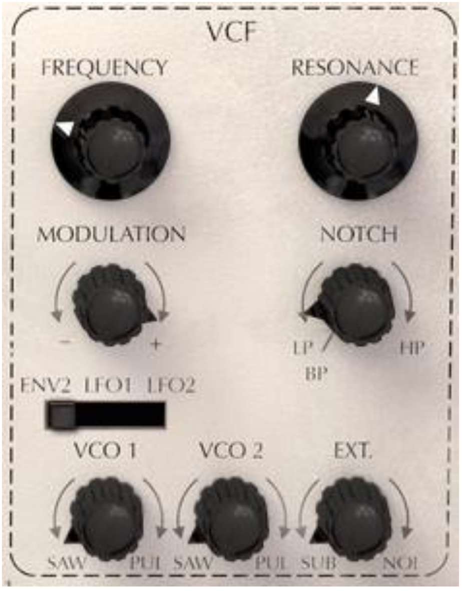
The switch in between OSC 1 and 2 is an on/off switch for oscillator pitch synchronization. When this switch is on the pitch of OSC 2 will be forced to synchronize with the pitch of OSC1.

For more details of OSC sync, please have a look at the next chapter.

The modulation knob can have two different functions: if turned to the left side, it will control how much of the the selected modulation source affects the oscillator frequency. But if turned on the right side, it will affect the modulation of the pulse width. In the center position, which can be reached by double-clicking for ease of use, there is no modulation of frequency or of pulse width.

The switch below the modulation knob determines the VCO's modulation source: either ENV, LFO1 or LFO 2.

3.4.2 VCF



This section determines the tone character of your sound.

FREQUENCY: This parameter indicates the frequency to be filtered which is dependant on the filter type.

RESONANCE: It normaly enhance the frequency determined by the FREQUENCY knob except when the filter type is set to notch.

For more details about filter type, please have a look at the next chapter.

MODULATION: The depth of filter frequency modulation. +(clockwise) means positive value and -(counter clockwise) means negative value. A double click will reset the value to zero.

The filter type selector is the knob to the right of the MODULATION knob. You can choose the type of filter you'd like to use, among the 4 kinds of filter types available: low pass, notch, high pass or band pass.

Modulation Source selector: This switch determines the modulation source affecting the filter frequency: ENV2, LFO1 or LFO2.

On the very bottom of the VCF section are three additional 'mix' knobs.

VCO 1: This knob works as waveform selector between sawtooth (if turned towards the left) and pulse (if turned towards the right) and also works as volume for each waveform, since the center point is no volume for either waveform. You can double-click on it to reset it to the center.

VCO 2: This knob works for VCO2, and the function is the same as the VCO1 knob. You can also double-click on it to reset it to the center.

EXT: This knob works as source selector between sub oscillator (if turned to the left) and the noise (if moved to the right) and also works as a volume control for each audio signal. You can double-click on it to reset it to the center.

3.4.3 ENV 1

ENV is an abreviation of Envelope. It creates a modulation shape that is used to control the VCA amplitude, or volume, of each SEM V voice. This shape is generated every time a note is played, and the three knobs present in the ENV1 section (Attack, Decay and Sustain) help determine the shape of the envelope.

In the case of the SEM V, the ENV1 can also be sent as a modulation source to VCO1's pitch or pluse width.

3.4.4 ENV 2

ENV 2 is not directly sent to a SEM V parameter like ENV 1 (sent to the VCA). It can be viewed as an external Modulation source, and can be sent to the filter frequency in the VCF or else to the frequency or pulse width modulation of VCO2.

On SEM V, the Decay knob controls the release time as well. Please have a look next chapter for more details.

3.4.5 LFO 1

LFO 1 generates a sinewave for modulation purposes. The frequency knob sets how "fast" the LFO is running. If you turn the switch to the "on" position, the frequency will be synchronized with the host sequencer's tempo and you can set the ratio between 1/32 to 16 times the host tempo.

3.5 Main panel: New SEM V features

The following parameters include brand-new, original modules of Arturia's SEM V that the real SEM didn't have.

3.5.1 Sub Osc

The upper switch determins the wave form of the Sub Oscillator, either sine, sawtooth or pulse. You can chose the pitch to be either of 1 or 2 octaves lower than VCO 1's pitch. The knob on the right side controls the width of pulse wave.

Sub Osc is not a independent oscillator, the pitch is always identical of Osc 1 and is either 1 or 2 octaves lower. For the detail, please read the chapter 5.

3.5.2 LFO 2

On this LFO, you can chose the modulation waveform by using the upper switch: either sine, sawtooth or square.

If you turn the "SYNC" switch to the "ON" position, the frequency will be synchronized with the host sequencer's tempo and you can choose the ratio between 1/32 to x16 the tempo.

If the "RETRIG" is set to "ON", the LFO phase will be reset when a key is played, if there are no other keys pressed. Otherwise, the LFO will be free-running.

The "FADE IN" knob allows the LFO amplitude to rise continuously from the moment a key is pressed, with a duration which is set by this knob.

The LFO 2 is a monophonic LFO: when playing the Oberheim SEM V in a polyphonic way, it will modulate all voices identically. On the other hand, LFO1 is part of the SEM module, so it is replicated in each voice.

3.5.3 Effects

SEM V has 3 kinds of effects (Distortion, Chorus and Delay). You can turn them on or off by using the switches on the right side and control the dry/wet balance by adjusting the knobs to the left.

For the details of the function of these effects, please read the chapter 6.

3.5.4 Output

The Level or Master Volume knob adjusts the final output level of the audio signal. It can range from -80dB to +24dB, so be careful, because it may cause audio clipping if set too high.

The Soft-clip feature allows to introduce a very subtle distortion to the signal which brings some additional warmth, like on analog output stages. At high volume levels, this may cause unwanted distortion.

3.5.5 Arpeggiator

When the upper left switch is on, the SEM automatically plays an arpeggio according to the notes which you play. When the switch is in the "HOLD" position, the Arpeggiator continues to play the note or chord which you played after keys are off- and until next note or chord is played.

The lower left switch determines the type of arpeggio when more than 1 octave is played: upward, downward, upward and downward, or random notes.

The rate knob controlls the speed of arpeggio. If the upper right switch is on (Sync), the speed is synchronized with the host sequencer and the knob works as tempo ratio that is set according to the host sequencer's tempo.

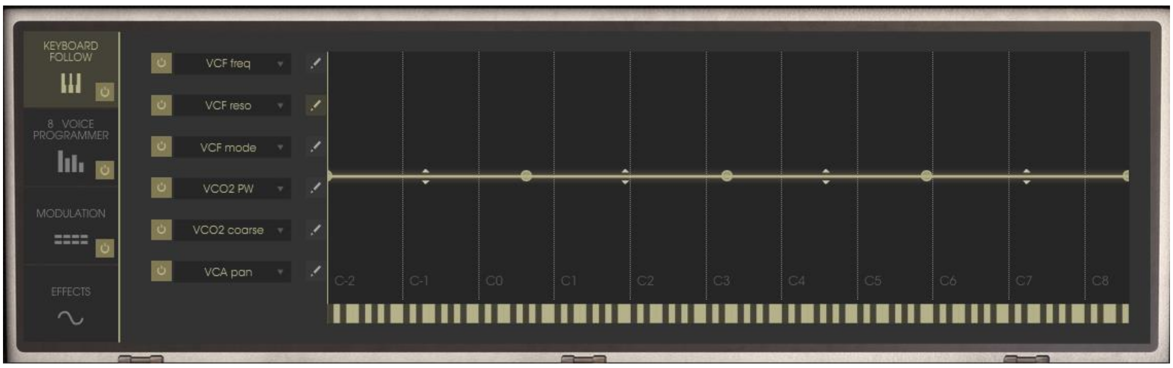
The lower right switch selects the octave range of the arpeggio. (1 to 4 octaves)

3.5.6 Tune and Portamento

TUNE controls the master tuning of SEM V within A=420Hz to 460Hz.

The PORTAMENTO section has 2 parameters. The ON-OFF switch and the knob that adjusts portamento time in between 0 millisecond to 2000 milliseconds. Portamento creates a glide in pitch between two notes that are played one ofter the other.

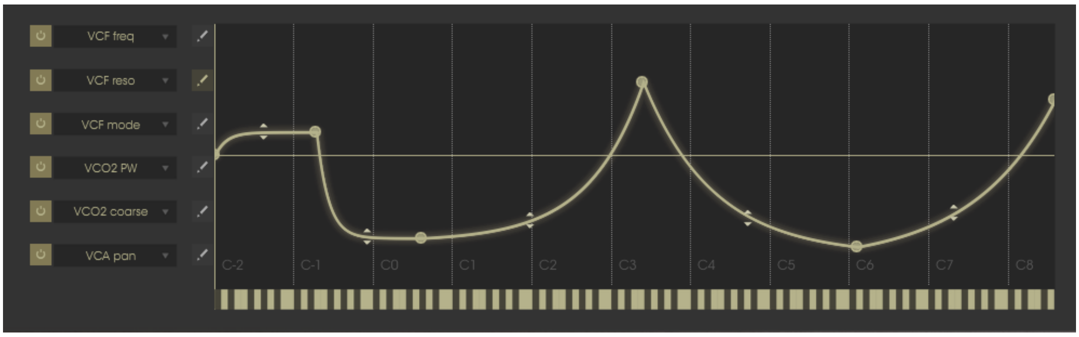
3.6 Open Mode



The top panel (which can be opened by clicking on the button on the upper toolbar with the two downward arrows) gives access to 3 special functions especially made for the SEM V. These advanced modulation functions are not available on the original analog SEM and highly enhance the sound making capacities of the SEM V: the “Keyboard Follow”, the “8 Voice programmer”, and the “Modulation Matrix”.

Keyboard follow is a fairly common parameter for many kinds of synthesizers; however, this "keyboard follow" has a several new features which are sure to make it stand out from the rest.

3.6.1 Keyboard Follow



The Keyboard Follow module of the SEM V allows you to modify the value of up to 6 parameters according to the note that is played on the keyboard. It does this by letting you draw the curve of 'reactivity', so to speak, between the pitch of the note and the parameter modulated.

3.6.1.1 Activate/Deactivate Keyboard follow

You can activate or deactivate keyboard follow function globally (for all 6 parameters) and individually for each parameter.

To activate or deactivate the entire keyboard follow function, click on the top left button (next to the words "KEYBOARD FOLLOW"): it will toggle the function on and off.

To activate or deactivate individual keyboard follow parameters, click on the corresponding button to the right of each individual parameter name (in the parameters column on the left).

3.6.1.2 Multi break points

Each keyboard follow parameter has 5 default break points: 2 fixed position points (for low and high end) and 3 movable ones in between.

You can easily add or remove the number of break points. Just click anywhere on the screen to add a new point. Right-click on the circle to remove the breakpoint. To move a breakpoint's position, simply click and hold on the circle and move it by dragging.

The number of the breakpoints is limited to 32.

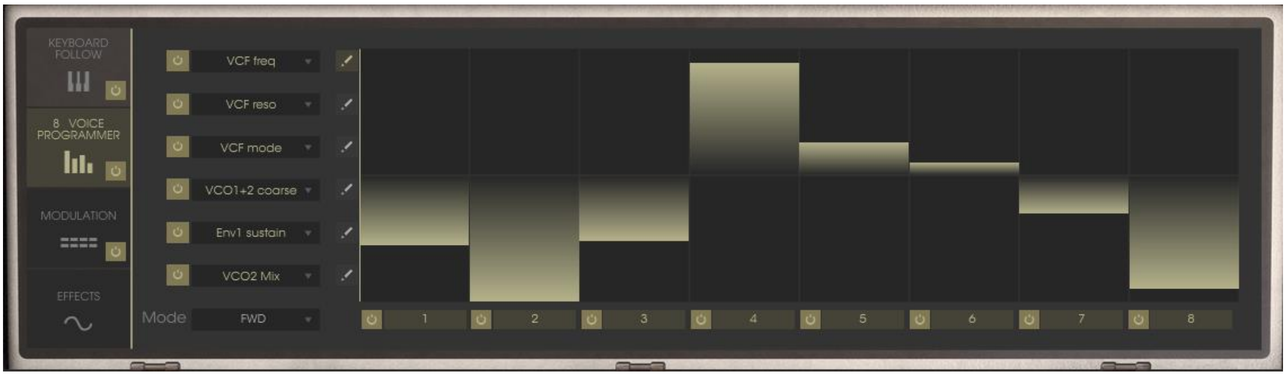
3.6.1.3 Linear and exponential slope

The slope between each breakpoint can be between linear or exponential. You can change the curve between two points by dragging the up and down arrows that are in the middle of each line segment between two breakpoints.

3.6.1.4 Changeable destination and activation switch

The keyboard follow destinations are changeable. Just right click on the columns on the left side: a drop down menu will appear showing you the parameters that can be assigned as destinations of the keyboard follow. You will also have the option of clearing the current curve for the selected parameter.

3.6.2 Voice Programmer



The 8 Voice Programmer module of the SEM V allows you to change the value of up to 6 parameters according to the current state defined by the "board" (see 2.2). The word board references the Circuit Boards used in the original polyphonic SEM-based synthesizers to give it 2, 4 or 8 separate multi-timbral voices. With this module, you can think of the SEM V as an 8-Voice synthesizer, where each voice can play a different sound, opening up an entire world of multi-timbrality.

3.6.2.1 Activate/Deactivate Voice Programmer

You can activate or deactivate *Voice Programmer* function globally and individually. To activate or deactivate entire function of the voice programmer, click on the top left button (next to "Voice Programmer"): it will toggle the function on and off.

To activate or deactivate individual Voice Programmer settings, click on the buttons to the right of the parameter names.

3.6.2.2 Polyphony

The polyphony of SEM V is, in theory, limited to 32 voices (depends on CPU power). However, in order to reproduce the multi-timbral function like a real 4 voice or 8 voice model, the SEM V is equipped with eight sound modules (which are called "boards").

To set the number of multi-timbral 'boards' (when in poly mode), click on buttons 1 to 8 (visible on the bottom of the window) to turn the corresponding board on or off.

3.6.2.3 The Barlines

The barlines of the window allow one to offset the desired parameter value from its original value defined on the main panel (using the normal GUI interface knobs). For each barline, the center position represents the same parameter value as the value defined on the main panel: there is no modulation of the parameter. Moving the barline upwards means giving the parameter a positive offset whereas moving the barline downwards means giving the corresponding parameter a negative offset.

3.6.2.4 Changeable destination and activation switch.

The Voice Programmer destination is changeable. Just right click on the columns on the left side: a drop down menu will appear showing you the parameters that can be assigned as destinations of the Voice Programmer.

3.6.2.5 Allocation modes

The bottom left black box determines the mode of voice allocation; in essence, the direction. On FWD (forward), the board goes from left to right. It goes the other way on BWD (backward), and can also be played back and forth on FWD BWD (Forward and backward). Finally, it can make the boards play randomly with the RANDOM setting.

3.6.3 Modulation Matrix

Thanks to this feature, you can modulate many parameter values by several control sources such as Pitch bend, Modulation wheel, Velocity, Aftertouch, LFOs and Envelopes.

To operate the modulation matrix, just click on the 'Source' columns and select the control source. Then click on the destination column and select one of the destination parameters.

The Amount knob, in between the two 'source' and 'destination' menus, sets how much modulation is sent from the source to the destination. The center position is the zero value, meaning no modulation is sent. Turning this knob to the left means giving the destination parameter a negative amount of modulation; turning it to the right means a positive amount.

You can also make a “layered assignment”, meaning you can select multiple sources (in the left column) to be the same modulation source, then assign them to different destinations. For example, with the modulation wheel, you can control the LFO depth and VCF cut off at the same time.

Conversely, you can control same destination with different sources, for example, control LFO depth by both the Modulation wheel and by Aftertouch.

3.7 Effects

SEM V has 3 kinds of effects which are distortion (overdrive), chorus and delay.

3.7.1 Overdrive

DRIVE: Adjusts the degree of distortion.

DAMPING: Reduces the high-frequency content of the output.

3.7.2 Chorus

SHAPE: Selects the chorus modulation waveform. You can choose either sine wave or noise.

RATE: Modulation speed.

DEPTH: Modulation depth/amount.

FEEDBACK: Value of the modulation feedback.

SPREAD: Stereo width of the chorus effect.

DELAY: Delay time of modulated signal (wet signal).

TEMPO SYNC: When this button is highlighted, the modulation rate varies in proportion to the host sequencer's tempo.

3.7.3 Delay

LINK: When this button is highlighted, left and right channel's delay times will be identical.

TIME: Delay time.

FEEDBACK: Delay feedback level.

PING PONG: When this button is highlighted, the delayed signal will be repeated with stereo panning.

DAMPING: Tone control of the delay signal. Moving this knob to the right will reduce high frequencies.

TEMPO SYNC: When this button is highlighted, the delay times are proportional to the host sequencer's tempo.

4 THE BASICS OF SUBTRACTIVE SYNTHESIS

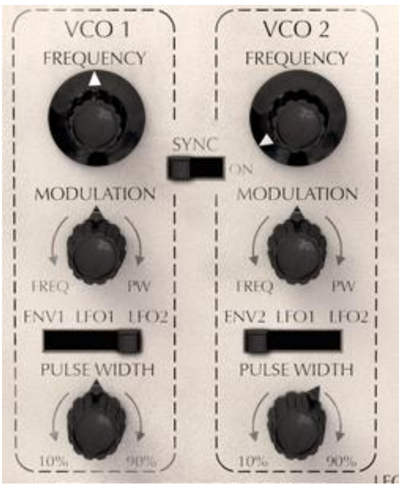
Of all forms of sound synthesis, subtractive synthesis is one of the oldest and still certainly one of the most employed today. It is this method that was developed on analog synthesizers like the ones from Bob Moog, ARP, Yamaha, Buchla, Oberheim, Sequential Circuits (Prophet series), Roland, Korg and many others towards the end of the 70's. This concept of subtractive synthesis is still used on most current digital synthesizers, complementing sample reading or wave tables, which progressively replaced the analog oscillators of the first synthesizers in the 80's. The original SEM, or even your own SEM V, is among the best illustrations of the enormous possibilities of subtractive synthesis.

4.1 The three main elements

4.1.1 Oscillator, or VCO

Oscillator (Voltage Controlled Oscillator) is the starting module (with the noise generator which is often classed among the oscillators) for the creation of a sound on an analog system.

It will generate the initial sound signal. We can think of the oscillator like a violin string that once stroked or plucked, vibrates to create its sound.



The oscillators

The main oscillator settings are:

- The **pitch**, determined by the oscillation frequency. You can set the frequency of the oscillator with controller knob.

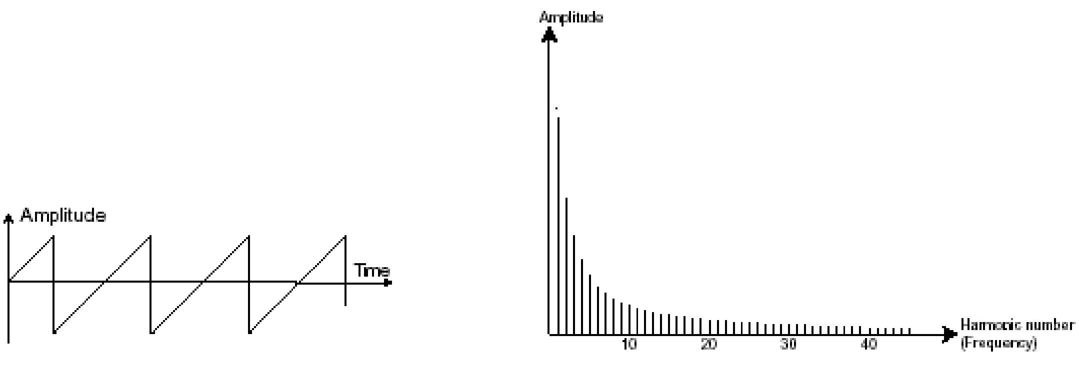
On the SEM V, pitch changes are made by semitone steps (Coarse) when you move the knob while using the left click. If moving while using the right click, the knob works as a "fine" tuning knob (changes the frequency in Cents).

- The **waveforms**, which determine the harmonic richness of the audio signal. On the SEM V, the waveforms shown below are available:
    - Sawtooth
    - Pulse
    - Sub Oscillator (1 or 2 octaves lower than OSC1, selectable as a saw or square)
    - Noise

SEM V waveform selector

4.1.1.1 Sawtooth

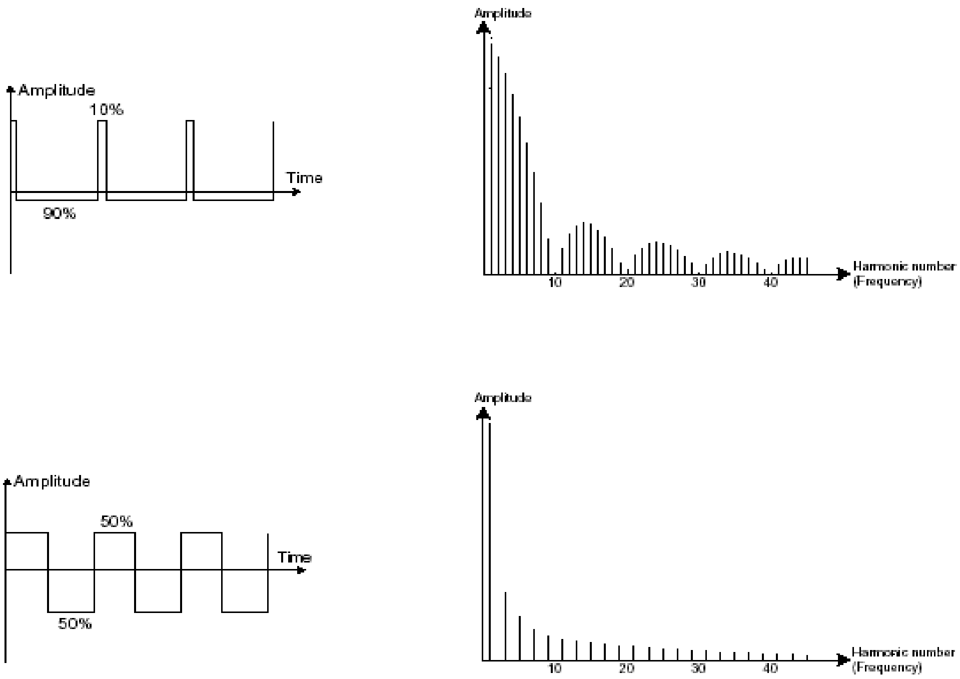
The **sawtooth wave** presents a rich audio signal (it contains all of the harmonics at decreasing volume levels in high frequencies). Its sound is ideal for brass, string, percussive bass, or rich accompaniment sounds.



*Time and spectral representations of the **sawtooth** waveform*

4.1.1.2 Pulse

Pulse wave possesses a more "hollow" sound that the sawtooth (it contains only odd harmonics). It is often used for woodwind sounds like clarinet or oboe (depending on the percentage of Pulse Width), and decaying sounds such as piano or guitar.



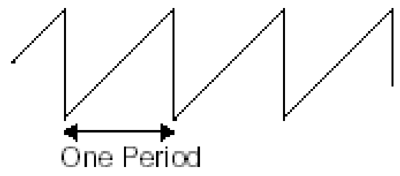
*Time and spectral representations of the **pulse** (top) and **square** (bottom) waveforms*

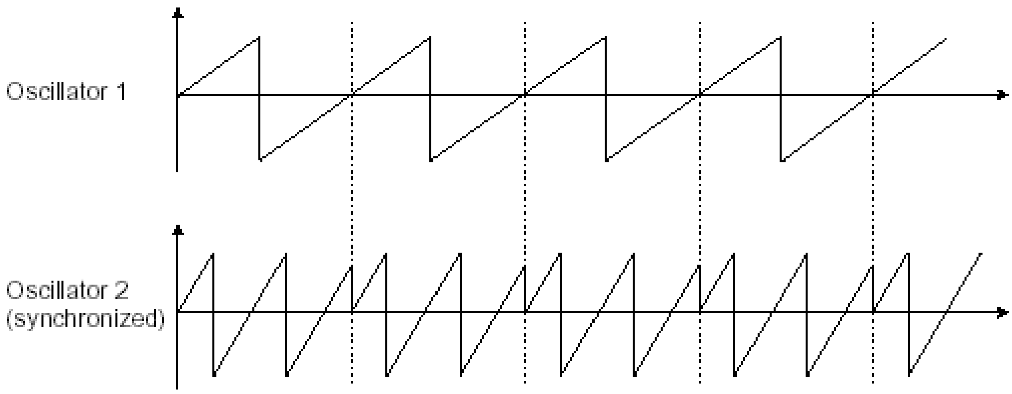
4.1.1.3 Pulse width Modulation

PWM (Pulse Width Modulation) is a setting that allows you to modify the percentage of the asymmetric diversity of pulse wave by using an LFO or the Envelope generator. This pulse width variation translates to a spectrum modification, resembling a waveform change.

4.1.1.4 Synchronization

The synchronization of an oscillator with another creates more complex waveforms. If for example, you synchronize oscillator2 with oscillator1, oscillator2 will restart a new period every time the first oscillator completes a period, even if oscillator2 has not completed a full period (this signifies that it is not tuned to the same tonality!) The more you tune oscillator2 upwards, the more you will encounter composite waveforms.





In the above image, oscillator2 is synchronized with the first oscillator and tuned to double the tonality. The resulting waveform is unique in that it cannot be created by standard synthesis techniques such as layering or filtering.

4.1.1.5 Sub Oscillator

The Sub Oscillator is not an independent oscillator module. Its pitch is derived by taking the pitch from Oscillator 1 and using a frequency divider to drop the pitch by either 1 or 2 octaves.

It is often used in single OSC synthesizers to make the sound richer and fatter, like an octave unison bass sound.

4.1.1.6 Noise

The noise signal spectrum has all frequencies at an equal volume level, often referred to as "white noise". For this reason, the noise is used to create special sound effects, like the imitation of the wind, jet planes, helicopters, steam locomotives and much more.

On pre-cabled (patched) synthesizers, the noise is either integrated into the oscillator (its audio output being placed to compliment the waveform outputs), or within the mixer directing the signals towards the filter.

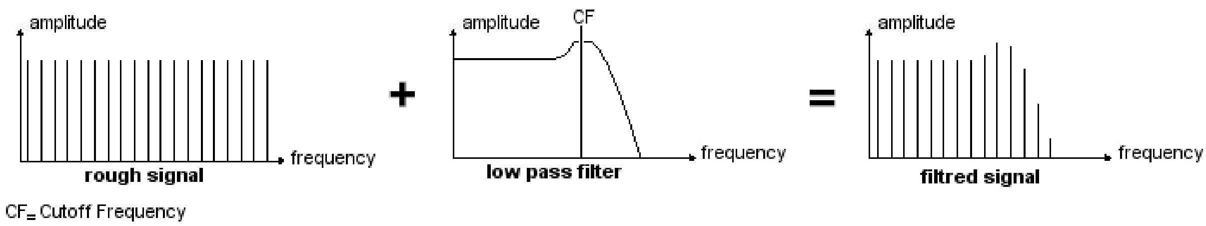
4.1.2 Filter or VCF

The audio signal generated by an oscillator (the waveform) is then usually directed to a filter module (**Voltage Controlled Filter**). It is this module that we use to control the sound by filtering (by subtraction, which explains the name given to this type of synthesis) the harmonics situated around a cut-off frequency. It can be seen as a sophisticated equalizer that reduces, depending on the case, the high or low frequencies of a sound.

4.1.2.1 Cut-off frequency

The removal of undesirable frequencies at the cut-off frequency is not done suddenly but progressively, depending on the filtering slope. This filtering slope is expressed in decibels per octave (or dB/Oct). The filters used in classic analog synthesizers have 24 dB/Oct or 12 dB/Oct slopes.

SEM V offers one type of slope: 12 dB/Oct slope.

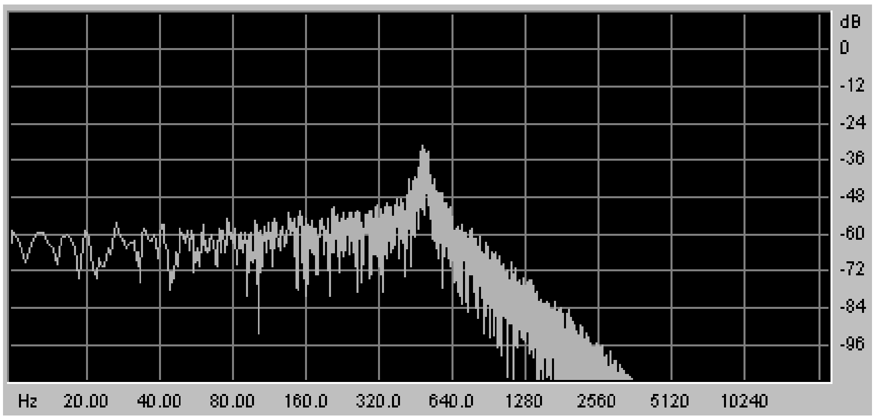


*Spectral representation of **12DB LP filtered** signal (rough)*

With the SEM V, you have access to three types of filtering.

The Low-pass filter (LPF) progressively eliminates high frequencies above the assigned frequency limit (cut-off frequency) and allows the frequency of the sound below the cut-off to pass through. We will hear the sound becoming more "bright" as we augment the cut-off frequency, or more "muffled" as we lower the cut-off.

This is the type of filtering that you will find most often on synthesizers that use subtractive synthesis. It can be found on most of the recent analog and digital synthesizers.



Spectrum of a noise signal proceed with a low-pass filter

The Notch Filter eliminates frequencies around the cut-off frequency and allows all other frequencies to pass through.

The High-pass filter progressively eliminates high frequencies below the cut-off frequency and allows the frequencies above the cut-off to pass through. We will hear the sound becoming more "thin" as the cut-off frequency is augmented, or more "thick" as the cut-off is reduced.

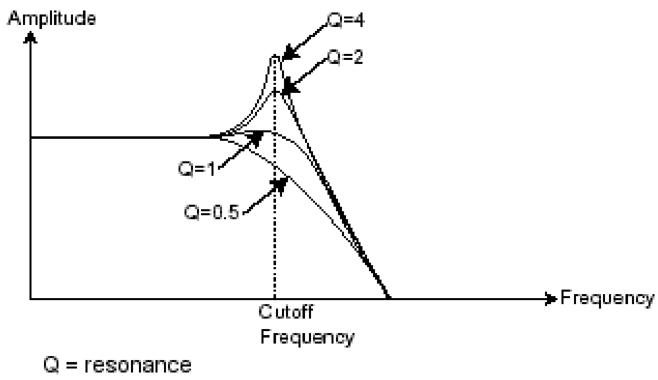
Band-pass filter eliminates the frequencies above and below the cut-off, and allows the band around the cut-off frequency to pass through.

4.1.2.2 Resonance

A second setting to compliment the cut-off frequency is the resonance. You will also find it called "Emphasis" or "Q" – for Quality of filtering on certain synths.

Resonance amplifies the band around the cut-off frequency. Depending on the type of filter, resonance reduces the frequencies above or below the cut-off frequency more or less drastically. However, resonance exceptionally works as the "band width" when used on the notch filter.

When you increase the resonance a lot, the filter range becomes narrower and the cut-off frequency is strongly amplified: the sound begins to ring.

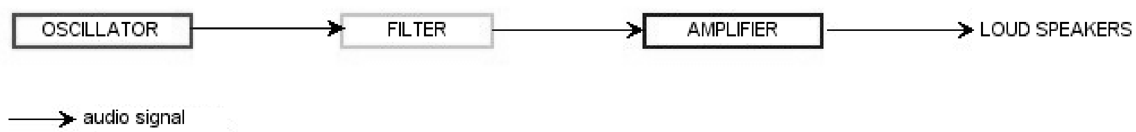


With a very high resonance level, the filter will begin to self-oscillate, creating sound which is close to a sine wave. At this stage, using a key follow on the filter frequency is very important, since it will allow you to play a melody with correct tuning.

4.1.3 Amplifier or VCA

The Amplifier (Voltage Controlled Amplifier) receives the audio signal coming from the filter and adjusts the volume before it is sent to the speakers.

In conclusion, here is a diagram that should help you to understand the composition of a basic sound:



Basic audio path in analog synthesis

4.2 Other modules

4.2.1 The keyboard

If we stop here, the sound that you will obtain will be uniform, without life and without an end!! The oscillator delivers a sound signal (the audio output of a waveform) of a fixed pitch in a continuous manner. In the diagram above, the only way to stop this quickly unpleasant sound is to lower the filter cut-off frequency so that it becomes more and more damp until it finally disappears; or simpler yet, lower the volume of the amplifier!

To start and stop the sound, at the pitch that we require, we use a keyboard that is connected both to the VCA through a gate and to the frequency of the oscillators. This will “play” the sound as soon as a key is pressed, and mute it when released. Of course, this connection is made through MIDI (it replaces the “gate” type of connections on analog synthesizers, which trigger the note when a key is pressed and stop it when released).

The key position provides a control voltage that tells the oscillator what pitch level to play when the gate opens.

If you don’t have a MIDI keyboard, you can also play on the virtual keyboard of the SEM V to hear sounds while editing voices on your computer.

4.2.2 The envelope generator

The envelope generator, connected to the amplifier, is used to “sculpt” the volume of the sound when we press a key on the keyboard. It ends (volume returns to zero) after the note is released.

The most common modules developed use 4 settings that we can vary:

Attack is the time that the sound will take to reach its maximum volume once we have pressed a key on the keyboard.

Decay (fall) is the time that the sound will take to diminish after the attack portion is complete.

Sustain (hold) is the maximum volume level that the sound will reach after the decay is complete. It stays at this level as long as the key is held.

Release is the time that the sound will take to fade to silence once the key has been released.

The two envelopes of the SEM V contain only 3 parameters: **A**ttack, **D**ecay and **S**ustain. The release time is thus identical to the decay time of the envelope.

The SEM V ADS(R) envelope

4.2.3 Low frequency oscillator

The LFO (*Low Frequency Oscillator*) possesses more or less the same characteristics of the classic oscillator but it only produces frequencies lower than 20 Hz. In other words, you won’t hear the sound from this oscillator.

It will create a cyclic modulation of the parameter to which it is connected.

For example:

- If the sine waveform of an LFO modulates the volume of an amplifier, the sound will increase in volume then disappear in a varying manner following the speed (the frequency) of this LFO. This will produce a **tremolo** effect.
- A sine waveform on an LFO modulating the frequency of an oscillator would produce a **vibrato** effect.
- With an LFO sine wave modulating the cut-off frequency of a slightly resonant low-pass filter, you will obtain a "**wah-wah**" effect.

The original SEM doesn't have LFO modulation for the VCA, so the SEM V doesn't either.

The LFO modules of the SEM V

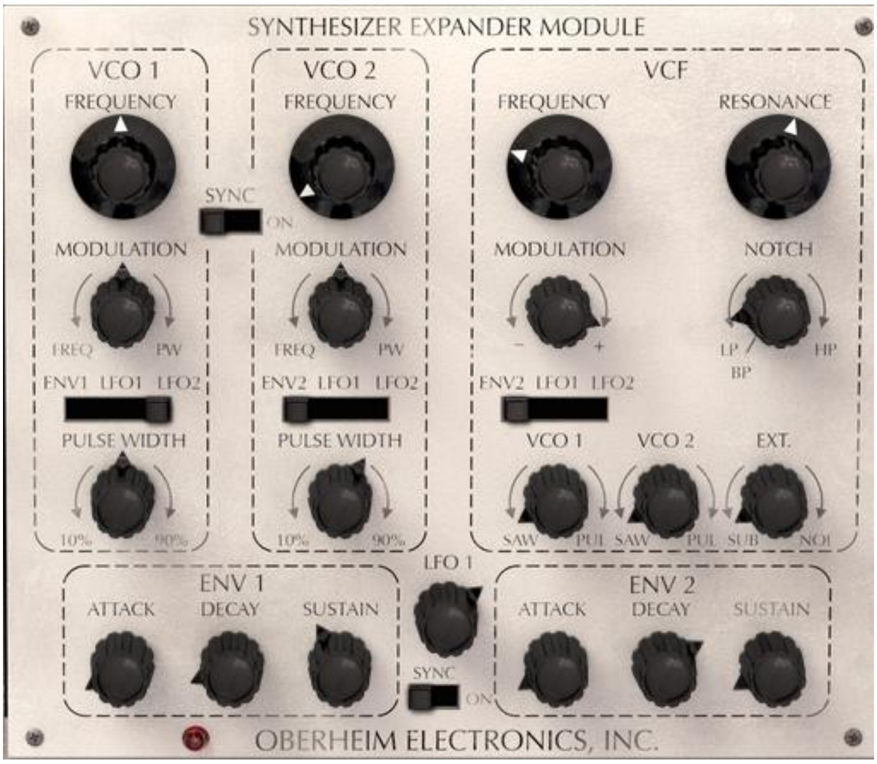
5 Elements Of Sound Design

Here are some examples that will guide you to make your own original sounds with SEM V.

5.1 Simple Synth Brass

One of the typical SEM sounds is simple but powerful synth brass sound.

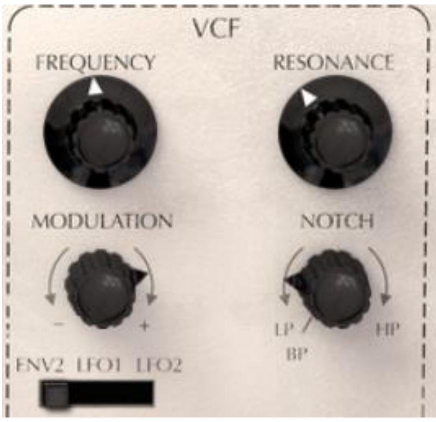
Let's try to make it from scratch (Templates / Init Voice 1).



First of all turn VCO 2 knob to the left in the VCF section. Now both of the oscillators are sounding.

Secondly, in the VCO2 section (to the left of the VCF section), turn the VCO 2 FREQUENCY knob slightly to the right with control + click (or right-click with the mouse) in order to get a comfortably detuned sound.

Next, returning to the VCF section, set the modulation source to ENV 2 then make the amount of modulation maximum (positive, towards the right). Turn FREQUENCY knob to the left to get the effect of ENV 2's modulation: the frequency here sets the point where the modulation returns to. Becasue the modulation is set to positive, when a note is played the frequency of the filter will increase (depending on the ENV2 settings that we'll look at in the next paragraph), brigthening momentarily the sound, then return to the frequency set here. Turn RESONANCE to the right to get sound to ring a bit more (enhance the filter frequency): the filter type should be set to LP (low pass filter)



Go to ENV2 and adjust ATTACK between 20 to 100ms (recommended to emulate the fast attack of brass instruments), then adjust DECAY to about 100ms.

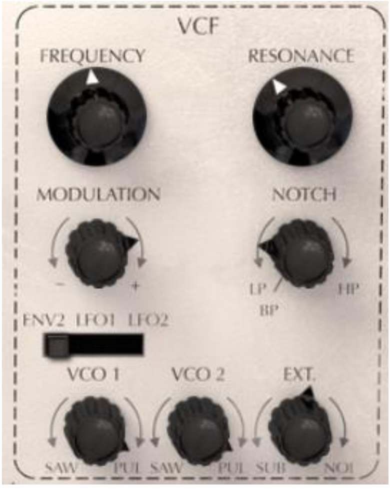
Now the basic Brass sound is completed.

5.2 Clavinet-like decaying sound with keyboard follow

This is another typical SEM sound.

Stay on the Brass sound that you have just made in the previous section. In the VCF zone, adjust the VCO 1 knob to fully right and VCO 2 to center. This means using VCO 1 only with pulse wave: VCO 2's volume is set to zero.

Leave FREQUENCY and MODULATION as they are, then adjust RESONANCE value to what ever you want (between 0% and 50% is recommended).



Go to ENV 2 and set the SUSTAIN knob to left-most setting (-60.00 dB) and adjust decay time to somewhere around 2000 ms: this gives our filter envelope no sustain but a very long decay and release time.

The next steps involve adjusting VCO 1 and ENV 1. In the VCO 1 section, turn the PULSE WIDTH knob to left-most setting in order to set the PW value to 10.00%, then go to ENV 1 and adjust the decay time to about 100ms and the sustain level to about -7 to -8dB.

This should give you a clavinet like sound with a long decay!

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