USER MANUAL

SEM-Filter

ARTURIA©

YOUR EXPERIENCE • YOUR SOUND

Special Thanks

DIRECTION

Frédéric Brun
Kevin Molcard

DEVELOPMENT

Corentin Comte
Baptiste Aubry
Matthieu Courouble
Raynald Dantigny
Pierre-Lin Laneyrie
Samuel Limier
Germain Marzin
Mathieu Nocenti
Pierre Pfister
Benjamin Renard

DESIGN

Baptiste Le Goff
Shaun Elwood

SOUND DESIGN

Jean-Baptiste Arthus
Gustavo Bravetti
Victor Morello

MANUAL

Fernando Rodrigues
Florian Marin

BETA TESTERS

Gustavo Bravetti
Andrew Capon
Chuck Capsis
Jeffrey M Cecil
Marco Correia
Dwight Davies
Luca Lefèvre
Terry Marsden
Fernando Rodrigues
Peter Tomlinson
George Ware

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11 Chemin de la Dhuy
38240 Meylan
FRANCE
www.arturia.com

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Thank you for purchasing SEM-Filter!

This manual covers the features and operation of Arturia's SEM-Filter.

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

Special Messages

Specifications Subject to Change:

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

IMPORTANT:

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

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

Introduction

Congratulations on your purchase of Arturia's SEM-Filter

Since the late 1990s, Arturia has received acclaim from players and reviewers alike for designing state-of-the art software emulations of the venerable analog synthesizers from the 1960s to the 1980s. From the Modular V, back in 2004, to Origin, a modular system that was introduced in 2010, to the Matrix 12 V (2015), the Synclavier V (2016), and most recently the Buchla Easel V, the DX7 V and the CMI V, Arturia's passion for synthesizers and sonic purity has given demanding musicians the best software instruments for professional audio production.

Arturia also has a growing expertise in the audio field, and in 2017 launched the AudioFuse, a pro studio quality audio interface that features two proprietary DiscretePRO® microphone preamplifiers and a set of top-notch AD/DA converters.

The Arturia SEM-Filter is another piece of software that results from over a decade of experience in recreating the most iconic tools of the past.

Arturia has a passion for excellence and accuracy. This led us to conduct an extensive analysis of every aspect of the SEM hardware and its electrical circuits, even modeling the changes in behavior over the course of time. Not only have we faithfully modeled the sound and behavior of this unique instrument, we have added some features that were unimaginable in the days the original Synthesizer Expansion Module (SEM) was being manufactured.

SEM V remains one of our most praised synths. So, it's no surprise that, when thinking of launching a new filter plug-in, the SEM-Filter came up as a natural choice.

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

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The Arturia team

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1. WELCOME

The Oberheim SEM (short for Synthesizer Expansion Module) was exactly what was implied in its name – a small synthesizer-in-a-box meant to be played by a sequencer or to expand the sound possibilities of other synthesizers of the time. Therefore, the SEM included all the main components of a synthesizer: two oscillators, a filter, an amplifier, an LFO and two envelopes.

Despite its deceptively simple appearance, the SEM was able to produce a great sound, which is, after all the most important quality. This was especially evident when Tom Oberheim started to build his polyphonic synthesizer systems, by combining several SEMs together. Even today, the big EVS (Eight-Voice System, which combined eight SEMs plus some special modules to allow to manage all of them) is revered as one of the most powerful sounding synthesizers ever built.

No other synth of the time (except maybe some custom modular systems) was able to produce such earth-shaking unison sounds as the ones that came out of the EVS. Yet, it was also capable of fantastic sounding, almost ethereal, synth pads.

Some years ago, Arturia launched the SEM V, a software replica of the sought-after SEM synthesizer, and now presents the filter part of that emulation, under the form of an audio FX plug-in.

The original SEM filter is very different from the filters that equipped synthesizers like the ones from the Moogs, the ARPs, or the Prophets. It's a state variable filter, which means that it can produce several filter responses (Low Pass, High Pass, Band Pass and Band Reject or Notch), simultaneously from the same filter design. The famous Bob Moog's filter, for example, only produced Low Pass filtering.

The advantage of this filter design is that not only can it produce several types of filtering, it does that continuously (we don't have to switch the filter from low pass to high pass, for example). In fact, we simply rotate the knob and it smoothly changes from low pass to high pass (with notch at the center point). Only exception to this behavior is the Band Pass filter, which is only reachable through a switching fixed point in the lower left.

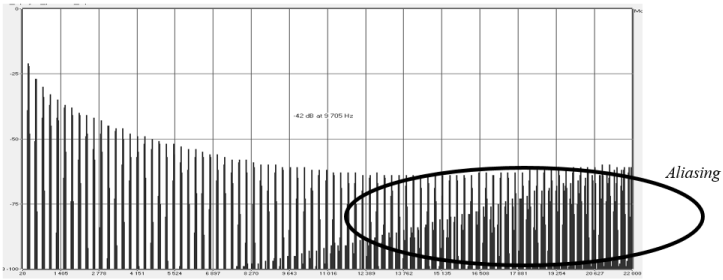
The filter also had another particular characteristic. Instead of a slope of -24 dB (meaning that the frequencies are attenuated by -24 dB per octave beyond the cutoff point) like the famous ladder filter, this one only had a slope of -12 dB. This means that the frequency cutoff is much softer. This was great especially for pads, strings and other sustained sounds, some of the sounds the Oberheims have been famous for.

It's this filter, with all its smoothness and originality, that Arturia is now releasing as an independent FX plug-in, to be used in audio, as it was before in many Oberheim synthesizers, like the mammoth Eight-Voice System or the OB-X.

Of course, the filter is complemented with the usual (and not so usual) modulation sources. So, we have the obligatory envelope and LFO, but we also have a gate sequencer, which is not so common. This feature set allows the user to be adventurous and creative about the sound.

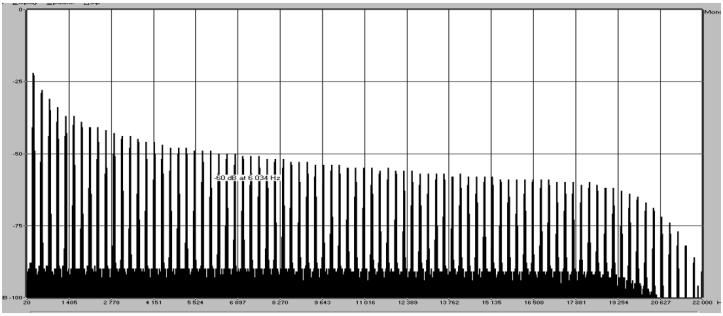
1.1. Arturia's secret ingredient: TAE®

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



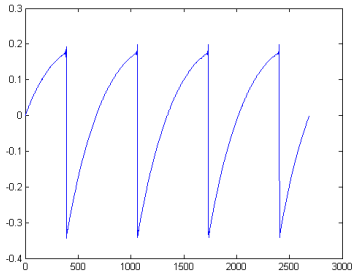
Linear Frequency spectrum of a well known software synthesizer

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

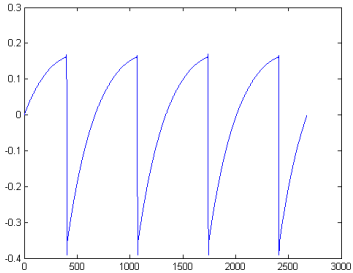


Linear frequency spectrum of an oscillator modeled with TAE®

TAE® combines major advances in the domain of synthesis:



Temporal representation of the "sawtooth" waveform of a hardware synthesizer



Temporal representation of a "sawtooth" waveform reproduced by TAE®

1.2. Arturia's SEM-Filter

So, what is and what do we get with the SEM-Filter plug-in?

The Arturia's SEM-Filter is a software version of the original SEM filter in the form of an audio plug-in, intended for use as an audio processor, as an insert effect or as a bus effect, in any plug-in host.

The plug-in includes the filter section of the SEM, complete with the sweepable filter mode between Low Pass and High Pass, with Notch in the center, and a switch for Band Pass in the lower left.

The filter is complemented with some modulators, to allow more variety and liveliness. Besides the Envelope section and the LFO section, we also have a Gate Sequencer section with sixteen steps, which can be used to modulate both the Envelope and the LFO, and a Modulation Grid where we can route the LFO and the Envelope to several modulation destinations, like Cutoff, Resonance, filter Mode, Noise source, LFO rate and amplitude, Envelope amplitude, and Filter Output.

2. ACTIVATION AND FIRST START

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

The SEM-Filter with all controls in their Init position

2.1. Activate the SEM-Filter license

Once the software has been installed, the next step should be to activate your license, so that you can use it without limitations.

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

2.1.1. The Arturia Software Center (ASC)

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

https://www.arturia.com/support/updates&manuals

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

Follow the installation instructions and then:

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

That's all there is to it!

2.1.2. SEM-Filter as a plug-in

SEM-Filter comes in VST, AU and AAX plug-in formats for use in all major DAW software such as Live, Cubase, Logic, Pro Tools and so on. You can load it as an insert or bus plug-in in your DAW. When inserted, you will observe the following:

- The plug-in will now be able to synchronize to your DAW's host tempo/bpm rate when tempo is a factor;
- You can automate numerous parameters using your DAW's automation system;
- You can use as many instances of SEM-Filter as you want in a DAW project, as long as your computer can handle them;
- It can be used along with any additional audio effects your DAW has available to further process the sound, including delay, chorus, other filters, etc.

2.2. QuickStart: Some patching tips

The steps below want to be a possible starting point for getting to know the SEM-Filter plugin. We will use the Band Pass filter and the Noise Input, modulated by the Envelope and the LFO, to create a rhythmic pattern that can be played in sync with the audio in a DAW host.

Please load the Init Preset. This ensures that you have all knobs in the correct starting position.

Now, let's give it try:

SEM-Filter after editing some settings to create a "Noise Rhythm"

- Load a four-beat clip into an audio track in your DAW (a bass track will be ideal, since it will be a great accompaniment for the rhythmic track we will create).
- Load an instance of SEM-Filter as an insert on that same track.
- Make the interface of the SEM-Filter visible by clicking over the name in the Mixer
- Now start your DAW, and the loop will sound in all its glory. By default the Sequencer sync switch is active and the filter cutoff frequency is set to maximum, with the filter mode in Low Pass position. That's OK with our demonstration.
- Stop the DAW. Set the Input Gain to -80dB (just turn the knob completely to the left) and the Input Noise to -20dB. For finer adjustment, you can use use Ctrl + Drag. In the lower toolbar, change the Noise Trig option to "MIDI Start". This makes SEM-Filter to produce noise as soon as the DAW starts running. Now, if you start your DAW again, you should just hear white noise, and no loop. It's OK. When Noise Trigger is set to "MIDI Start", noise only plays when we start the DAW, and since the Input Gain is set to the minimum, we will not hear any sound coming from the audio track.
- Now change the Filter Mode to BP (you have to drag the knob position to the lower left fixed "switch" position). In this mode, just a small range of frequencies with the center defined by the Frequency cutoff knob will be audible.
- Now change the Filter Frequency so that it shows a Cutoff around 160Hz. You can check the value in the lower toolbar. Again, for finer adjustment, you can use Ctrl + Drag. Check if Dry/Wet is set to full Wet (knob completely turned to the right).
- At this point, you still don't hear anything except noise. We need to modulate the filter. For that, we will use the Envelope and the Gate Sequencer (the envelope will not work if it doesn't get a trigger signal from the Gate Sequencer).
- So, we will start with the Gate Sequencer. As default, it will be set to 16 steps. We will create a rhythmic pattern in the Envelope row by turning On the 1, 3, 4, 5, 7, 8, 10, 12, 14 and 16. Now, we will switch the Sync switch to On, and change the Rate value to 1/16. The sequencer will advance one step every 16th note.
- Now in the Modulation Grid, we need to assign the Envelope as a modulation source. We will assign it to several sources, with the following values: Frequency 55; Mode 25 (this will not have an immediate action, since when Band Pass is selected, the filter mode can't change); and Noise 45.
- After this, we will also use the LFO to modulate the Envelope amplitude, to create some movement. For that, we will choose the S&H (sample & hold) waveform. Sample & Hold isn't technically a waveform, but since it's a function built in the LFO, it's here that we select that function. We'll also put LFO in Sync mode, and give it a rate of 1/16 too.
- To add some more movement, we will use the Gate Sequencer to also trigger the LFO. We turn On the 2, 4, 6, 7, 9, 11, 13 and 15 points in the LFO row. LFO phase will be reset at some regular intervals, to give it more motion.
- We still need to assign the LFO to modulate the Envelope. For that, we will locate the Env Amp destination cell in the LFO row of the Modulation Grid, and will give it a value of 25 (you can type directly in the cells the value you want, by clicking in the cell, typing 25 and press Enter or Return key in the computer keyboard). We will also use the LFO to modulate the Resonance with a value of 99 (although Resonance should still be at 0 in this moment, you will hear already some action because of this extreme modulation amount).
- Now, the final step. To have percussive but audible impulses fired at the Filter, we will raise the Decay segment of the Envelope to 0.060 seconds. If the output is not strong enough, try raise the Filter Out value.
- If you now start your DAW, you will get a rhythm pattern played entirely by the SEM-Filter. You can now adjust Resonance, and hear the timbre changing (what used to sound a bit like hi-hats will now sound something like synth toms).
- We will now try a different filter mode. We will move the Mode knob from the fixed Band Pass position to something around 66% Low Pass / 34% Notch (a little more than half-way between full Low Pass and Notch). The previous "synth toms" will now now sound again like hi-hats, but with a different timbre than before.

- Now turn the Input Gain a little to the right. The bass loop sound will become audible and will mix, in sync, with the rhythm track. Note that some kind of deep kick sounds will also become audible. This is the effect of the SEM-Filter in the bass loop audio. Control your mix and balance between processed and unprocessed sound using the Input Gain, Filter Out and the Dry/Wet control knobs, to see what is the best overall sound you can get.
- Turn the Soft Clip switch On to add saturation to the sound. Remember to balance the Input Gain, so that you don't cause clipping.
- Now you can try different waveforms in the LFO, to get different results. Sine is always a safe bet, but also the Saw Down produces good results (it works a bit like an AD envelope). And you can create new patterns by turn Off and On different steps in the Gate Sequencer, either for the Envelope and/or for the LFO.

We hope that, by now, you already got a pretty good knowledge of some of the things you can do with the SEM-Filter.

3. USER INTERFACE

The SEM-Filter is very flexible. That will always be the main focus of every Arturia product: to unleash your creativity while remaining easy to use.

3.1. The Control Panel

We will have a detailed look at the Control Panel of the SEM-Filter and all its different sections in the VCF [p.25] and subsequent chapters.

3.2. The Toolbar

The plug-in GUI (Graphical User Interface) has the usual Arturia toolbar that runs across the top edge, with the Arturia logo / plug-in name on the left (the colored part), followed by the Library button, the Library selection filter button, the Preset name on the center, and the MIDI button on the right. This toolbar is common to all current Arturia plug-ins, and gives access to many important functions. Let's look at them in detail.

The first seven of these options can be found by clicking on the Arturia SEM-Filter button at the top left-hand corner of the plug-in window. Since these options are also common to all Arturia current plug-ins, they may be already known to those that have some of them.

3.2.1. Save

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

3.2.2. Save As...

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

Save As dialog box in SEM-Filter plug-in

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

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 .sfix 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 to store presets.

3.2.4. Export Menu

You can export presets in a number of ways; as a single preset, as a bank or as a playlist.

- **Export Single Preset:** Exporting a single preset is handy when you want to share a preset with someone else. The default path to these files will appear in the "save" window, but you can create a folder at another location if you like. The saved preset can be reloaded with the import preset menu option.
- **Export Bank:** This option can be used to export an entire bank of sounds from the instrument, which is useful for backing up or sharing presets.
- **Export All Playlists:** This is an option you could use to prepare for a performance. It also allows you to transfer your playlists to another computer.

3.2.5. Resize Window options

The SEM-Filter window can be resized from 60% to 200% of its original size without any visual artifacts. On a smaller screen such as a laptop, you 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 control panel. The controls work the same at any zoom level, but they can be harder to see at the smaller magnification values, or when using high resolution monitors (like HD monitors or higher). The higher the resolution, the bigger the size that should be used.

3.2.6. Preset selection

The Preset browser [p.17] can be opened by clicking the library symbol on the toolbar. The filter, name field and left / right arrows in the toolbar all assist with preset selection.

3.3. MIDI Learn assignment

The MIDI plug icon at the far-right side of the toolbar places the instrument in MIDI learn mode. MIDI-assignable parameters will be shown in purple, which means you can map physical controls to those destinations inside the instrument. A typical example might be to map a real expression pedal to the Filter Out, or the Dry/Wet control, or knobs and faders on your controller to the different virtual knobs and switches of the plug-in.

SEM-Filter ready to "learn" MIDI assignments

In the image above one of the parameter dials is red. That means it has already been assigned to an external MIDI control. It can be reassigned, though.

> ♪: Remember that you can also assign the Preset Forward and Backward arrows to an external control.

3.3.1. Assign / Unassign controls

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

3.3.2. 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 Output Volume knobs to be controllable via hardware from 30% to 90%. If you make this setting (Min set to 0.30 and Max set to 0.90) your physical dial will be unable to alter the volume lower than 30% or higher than 90%, no matter how far you turn it. This is very useful to prevent you from accidentally making 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.

3.3.3. Relative Control option

The final option in this window is a button labeled "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 dial is turning, as opposed to sending a full range of values in a linear fashion (0-127, for example).

To be specific, a "relative" dial 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 dial) will change the software parameter by starting at its current setting, rather than being an "absolute" control and snapping it to some other value as soon as you start to move it.

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

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

3.4. MIDI controller configuration

There's a small arrow at the far right-hand side of the toolbar (after the MIDI icon) 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-Filter without having to build all the assignments from scratch each time you swap hardware.

SEM-Filter with the MIDI controller configuration opened

Note the check mark on the bottom of the menu: that indicates that the configuration with that name is the one currently active. Empty means that there's no configuration loaded.

3.5. The lower toolbar

When you are changing a parameter, you will see a readout showing the value or state of whatever control you are modifying at the left-hand side of the lower toolbar. It will also display the current value of a parameter when you place the mouse pointer over that parameter control in the control panel. This is handy, because you don't need to touch the parameter control to read the current value.

At the right-hand side of the lower toolbar are several small windows and buttons. These are very important features, so let's take a closer look at them.

3.5.1. Noise Trig

This is where we select the kind of triggering to be used by the SEM-Filter noise source. The amount of noise is controlled by the Input Noise knob in the Control Panel. By default, triggering is assigned to "Threshold", which means the noise source is activated by the incoming audio, when this passes the pre-determined threshold (currently around -37dBFS). The other option is "MIDI Start", in which case the noise source is activated every time the DAW starts.

The Threshold option allows us to use the SEM-Filter in situations where we don't have a DAW running (for example in a live rig). In this case, the noise source can react to and be triggered by the incoming audio.

3.5.2. Bypass

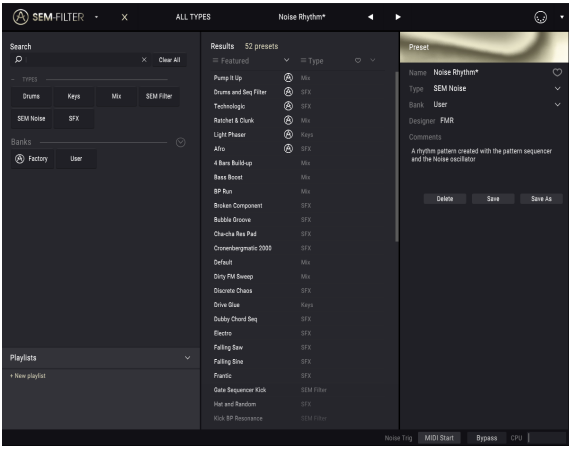
Activating the bypass option will disable SEM-Filter.

3.5.3. CPU meter

The CPU meter is used to monitor how much of your computer's CPU is being used by the plug-in. If you stress your computer too much, the global performance of your system and the audio may suffer.

3.6. The Preset browser

The preset browser enables you to search, load and manage preset configurations in SEM-Filter. It has a couple of different views but they all access the same banks of presets. You access the preset browser by clicking on the library symbol next to the Arturia logo / plug-in name on the left (it's the button immediately after the colored part of the toolbar).

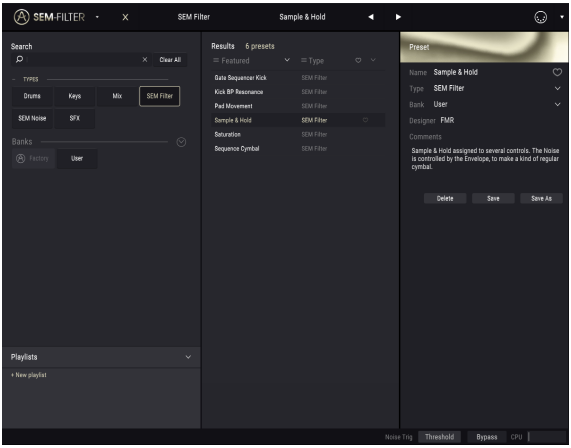


The SEM-Filter's Preset Browser

The Type category window in which the characteristics of a preset are listed may be collapsed and expanded using the symbol preceding it.

3.6.1. Searching Presets

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



Using the filter to search for presets

3.6.2. Using the filter

You can also search using different tags. So, for example by clicking on the SEM Noise tag in the Types field, you can show 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 select multiple 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 criterion and widen the search without having to go back and start again.

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.

3.6.3. Search Results window

Once you have a list of found presets in the search column, you can click the sort arrow to reverse the alphabetical order.

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

3.6.4. The Preset Info section

The Info column on the right side of the search field shows you information about any preset. The information for User Presets may be changed here: Name, Type, Bank, 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. You can now change the preset info in the "Save As" dialogue window.

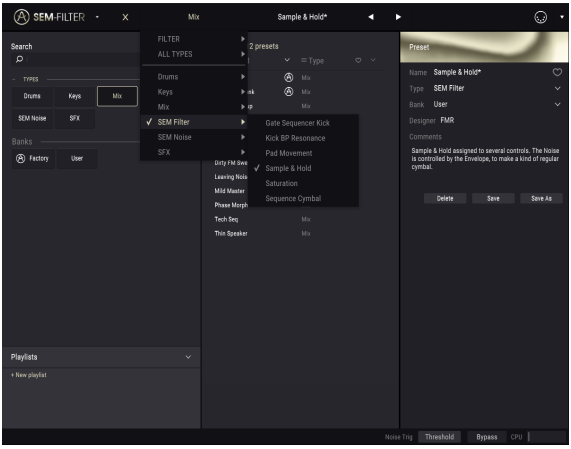
3.6.5. Preset selection: other methods

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

3.6.5.1. Selecting a Preset by type

Similarly, if you previously selected Type: SEM Filter in the Search field you would see the results of that search in this area instead.

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



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 preselected a Characteristic such as "Chaos", 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.

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

3.7. 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.

3.7.1. 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.

3.7.2. Add a preset

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

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

3.7.3. Re-order the presets

Presets may be reorganized within a playlist. For example, to move a preset from slot 1 to slot 3, drag and drop the preset to the desired location.

This will cause the other presets to be bumped up in the list to accommodate the new location of the preset being moved.

3.7.4. Remove a preset

To delete a preset from a playlist, select the preset and click the "x" at the end of the preset row.

3.7.5. Delete a playlist

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

4. SEM-FILTER OVERVIEW

The SEM (Synthesizer Expansion Module) was the first attempt of Tom Oberheim to enter the world of analog synthesizer builders, which was, by then, still at the beginning. It was a small and deceptively simple-looking module, with just the minimum components to make sound, and didn't even have a keyboard or any other way to trigger sound directly, since it was intended to be controlled by a sequencer or other synthesizer. Yet, it had a great sound, and that's what made it a triumph.

One of the main components of an analog synthesizer is the filter, and the SEM filter was completely different from the filters included in the ARP and Moog synthesizers, which dominated the market by then. This placed it apart from the competition, and contributed to grant the SEM its own place.

The synthesis method was subtractive, which means that some spectrum-rich sounds produced by oscillators were routed to one or more filters where some frequencies were cutoff and some were emphasized, and in that way changing the timbre. To make the sound more alive, some modulation sources were usually incorporated, most notably contour envelope generators and low frequency oscillators, but also sometimes sequencers, named this way because they would play a sequence of control voltages and trigger controls (usually eight or sixteen) in a row.

These modulation sources were then routed to one or more destinations, including the filter cutoff and also the resonance (sometimes also called emphasis).

The filters were usually Low Pass, which means that the frequencies above the cutoff point are attenuated in a certain slope, or roll-off (usually -24dB per octave, which means that frequencies one octave above the cutoff point are lowered by -24dB).

The SEM filter, however, didn't follow this path. Instead of a Low Pass filter, Tom Oberheim used a different circuit, known as State Variable Filter (SVF). This circuit is capable of several configurations, and, more important, it doesn't have to switch between one or another. Instead, the SEM filter gradually evolves from Low Pass to High Pass, with a Notch configuration in the middle. So, the filter changes gradually from Low Pass to Notch until it reaches the center position, and then gradually changes from Notch to High Pass. Only the Band Pass filter configuration is fixed, and is reached by rotating the Mode knob to a lower (switching) point on the left.

This circuit gave the filter much more versatility, allowing it not only to cut frequencies above the cutoff point but also to cut frequencies below the cutoff point, or even something in between one and the other and Notch.

In Notch configuration, frequencies are cut in a more or less narrow range, with the peak in the cutoff point, and a slope defined by the filter own slope and the cutoff intensity.

On the other hand, the Band Pass configuration (which, as we said, is the only fixed configuration in the SEM filter) is like the Notch inverted, which means that all frequencies are cut EXCEPT for a more or less narrow range, with the peak in the cutoff point, and a slope defined by the filter own slope and the cutoff intensity. Like the name says, in Band Pass configuration, the filter lets pass ONLY this range (band) of frequencies.

Besides these multiple configurations, the filter also had resonance, as was the case with all the synth filters at the time, which means that frequencies surrounding the cutoff point could be boosted, or emphasized (therefore the designation Emphasis that was also used).

Both the cutoff point and cutoff intensity, as well as the resonance amount, were usually modulated, which gave life and movement to the sound. The most common modulation sources were envelope generators and low frequency oscillators, as we said before.

By cutting frequencies above or below a certain point, or even using a more or less narrow band, emphasizing frequencies surrounding that cutoff point, and changing dynamically that same cutoff point as well as the cutoff intensity and the resonance amount, by means of envelope generators and low frequency oscillators, we could achieve a great amount of dynamic timbre transformation over time.

Now imagine what we can do with this same sound processing applied to real audio. That's what the SEM-Filter plug-in was created for. We even have the previously mentioned modulation sources to give life and movement, including a step sequencer (here named Gate Sequencer).

4.1. Animating the filtering

A filter modifies sound by removing frequencies beyond the cutoff point (usually above, but in the case of the SEM-Filter we have several other choices, as we saw). Doing this manually is not very effective, although we can hear the result, and have an idea of what's possible. What turns the filter into an interesting musical tool is the fact that we can change (modulate) dynamically several parameters at once, including the cutoff point, the filter intensity and resonance amount, by means of one or more modulators.

In the case of SEM-Filter, we can even change dynamically the filter mode, making it sweep from Low Pass to High Pass through Notch and vice-versa.

To achieve the mentioned dynamic control, we have an Envelope, an LFO and a Gate Sequencer to use as modulation sources. We also have an extra noise source which we can add to the signal or use exclusively as source to be processed by the filter to produce sound. Finally, we have an extra Modulation Grid to create more complex modulation routings. Additionally, we can use MIDI sources to create even more variety. Please refer to the Envelope [p.31], LFO [p.33], Gate Sequencer [p.35], and Modulation Grid [p.37] chapters for more details.

Tom Oberheim's state variable filter, as we mentioned, became known because of its versatility and musicality. Not only it was possible to use several configurations, we also had a gentler slope (instead of attenuating the sound by -24dB per octave, as was common in the other filters of the time, it only had a slope roll-off of -12dB per octave). It's this soft and gentler slope that also contributed to its "sweetness". It also makes it great for pure audio processing, in our opinion.

The presence of several modulation sources contributes to give variety and movement to the processing.

4.2. SEM-Filter sections

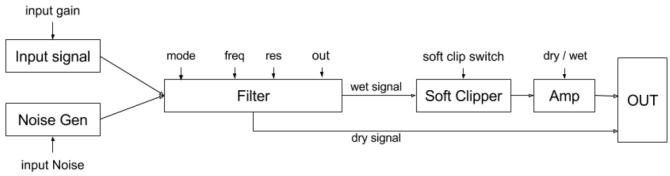
The SEM-Filter Control Panel consists of five sections:

1. The VCF [p.25] (Voltage Controlled Filter)
2. The Envelope [p.31]
3. The LFO [p.33] (Low Frequency Oscillator)
4. The Gate Sequencer [p.35]
5. The Modulation Grid [p.37]

4.3. Signal Flow

The VCF [p.25] is the heart of the plug-in. This is the section that receives the external audio signal and processes it. It also has a noise oscillator, that can be added to the signal or used alone to generate a noise signal that can be processed by the filter.

With the proper modulations, we can create interesting rhythmic tracks by using just the Noise Oscillator processed by the filter.



Signal Flow diagram

External audio is received in the plug-in by opening the Input Gain knob (rotating it to the right). Noise can be added by opening the Input Noise knob. The entire signal will then be routed to the filter, and from there to the Filter Output, through a Soft Clipper. Finally, there is a Dry/Wet control that allows you to control the mix of processed and unprocessed audio that goes to the Main Out (the dry signal is always routed to the Main Out too so, whenever the Dry/Wet control is not at full wet position, we will hear also the unprocessed signal).

The filter can be modulated by the LFO and/or the Envelope, which can also modulate the Wet amount. On the other hand, both the Envelope and the LFO can also be modulated by the Gate Sequencer, and can as well be modulated by themselves or each other, via the Modulation Matrix Grid in the bottom. Some quite complex modulation routings can be achieved by means of this Modulation Grid, as we will see.

4.4. Controls

Before proceeding with the analysis of the different sections one by one, just a few words about the SEM-Filter controls.

The majority of the knobs in the SEM-Filter plug-in are unipolar, which means that their minimum value is reached when the knob is turned all the way to the left, and we increment values by rotating it to the right. We can rotate the knobs by clicking them and dragging the mouse up or down. Dragging up will make them rotate to the right, and dragging down will make them rotate to the left.

While we're at it, take note that, by pressing Ctrl + Drag, we can achieve finer editing. This is good for when we want to assign a precise value. This way, values move slower, and we are able to precisely set the value we want.

However, some knobs are bipolar, which means that they have a center value at 0.00 and assume negative values when turned to the left and positive values when turned to the right. This is the case with the volume control knobs, namely the Input Gain knob and the Filter Out knob. In this case, the default value is 0.00 and is in the center position.

Also, not all of the unipolar knobs have their default value set to the minimum. For example, the Frequency cutoff knob default value is set to the maximum (in this case 15kHz) which, in conjunction with the filter Mode knob set to the default position of full Low Pass (completely turned to the left) will let basically all audio pass through.

The Dry/Wet control knob also defaults to the max value, which means full wet signal.

On the other hand, all the values in the Modulation Grid are bipolar, which means that we can have both positive and negative modulation values, and all of them default to 00 (neutral value). The Modulation Grid also allows you to enter exact values using the computer keyboard. We just need to click the cell we want to edit, type the number and press the Return or Enter key.

Double clicking on any control will reset it to the default value (this also works for the Modulation Grid cells).

5. THE VCF (VOLTAGE CONTROLLED FILTER)

The SEM-Filter is packed with great features and in this and the subsequent chapters, we'll take a tour and show you what it can do. We think you'll be amazed by the range of filtering options and editing possibilities this plug-in is capable of.

A filter is a device that removes ranges of frequencies from any audio source routed through it. Since it "subtracts" partials, the synthesis technique based on filtering is called subtractive. However, since this was the most common synthesis technique used in analog synthesizers since they appeared in the 1960s, it became also known as analog synthesis (although nothing prevents an analog synthesizer from using other synthesis techniques).

By removing certain frequency ranges and also boosting the frequencies around the cutoff point, the filter alters the global timbre of the sound, sometimes in a way so drastic that it becomes unrecognizable.

The SEM-Filter is a State Variable Filter. This kind of filtering circuit allows multiple configurations without the need to switch from one configuration to another. Therefore, we can start with a Low Pass Filter (frequencies above the cutoff point are removed) and sweep gradually to a High Pass Filter (where the frequencies being removed are the ones below the cutoff point).

This can be achieved by manually rotating the filter mode knob of the SEM-Filter, but that's not very practical. Instead, we will probably use one or more of the modulation sources at our disposal, to achieve that same transformation dynamically and in a musically controllable way.

The frequencies around the cutoff point can also be boosted by using the Resonance control, which contributes again to alter the original timbre of the sound. Bear in mind that if we modulate the cutoff point (another possibility), the resonant frequencies follow that modulation, since they are tied to that point.

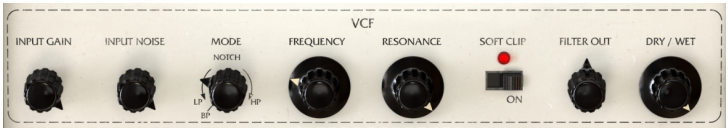
The SEM-Filter

Another word about the Resonance control. The original SEM filter didn't self oscillate, even if we pushed the resonance to its maximum amount. This characteristic was preserved in the SEM-Filter plug-in. So, we can push resonance at will without fear of it enter self oscillation. However, resonance creates a boost in frequencies around the cutoff point, which may or may not be desirable so, use that feature carefully.

That said, we also have a Noise source (labeled Input Noise, although it is self generated) which we can mix with the audio input, or use as a self contained generator for some special effects. This is particularly effective when setting the filter to Band Pass (switching the Mode knob to the extreme-left point, labeled BP) and modulating the cutoff and resonance with the Envelope and LFO, while having these triggered by the Gate Sequencer. Some great rhythm patterns can be created this way.

But it's time to check all the controls available in the VCF section, one by one, and see what we can achieve with them. The controls we have at our disposal are:

- Input gain
- Input Noise
- Mode
- Frequency
- Resonance
- Filter Out
- Soft Clip
- Dry/Wet



The SEM-Filter VCF section

5.1. Input Gain

The Input Gain controls the volume of the input signal, as expected. It starts at -8OdB and goes up to +24dB. We need to be very careful to not overload the audio circuit, since it is easy to reach distortion (there are a lot of gain stages). In our tests, we never went up to more than +16dB, and that was with the plug-in used only for inducing saturation.

Input Gain cannot be modulated by any internal control, but can be controlled by MIDI (see MIDI Learn Assignment [p.13] in Chapter 3). The default value is OdB (center position).

5.2. Input Noise

The SEM-Filter includes a noise source, with which we can achieve very interesting results, when combined with the filter and resonance. Its gain goes from -Infinite to -6dB.

The noise can be triggered by the incoming audio or by MIDI start. The trigger source is selected in the lower toolbar. By default, it is assigned to "Threshold", which means the incoming audio. For the audio to trigger the Noise input, it needs to have a volume at least around -37dBFS, the pre-determined "threshold". When we switch the option to "MIDI Start", the noise source is triggered by the DAW (it only sounds when the DAW is running).

Noise volume can be modulated by both the Envelope and the LFO. Modulation values can be positive or negative.

The default value is -Infinite (knob points to the lowest point on the left).

5.3. Mode

This is where we configure the filter behavior. As we have been saying, the SEM-filter is state variable, which means it doesn't have a fixed mode, like for example the famous Bob Moog ladder filter. We have a switch position at the lower left of the knob control to activate Band Pass filter mode. This mode is fixed, which means that, when it is On, we can't vary the mode, but when this isn't activated, the filter varies continuously from Low Pass filter to Notch, is fully Notch at the center, and then varies continuously from Notch to High Pass filter.

This allows very interesting variations, especially with complex spectra sounds, and when we use the different modulation sources to add movement to the filter mode. We can modulate Mode using the Envelope, the LFO or both. We can also use several MIDI sources (again, see MIDI Learn Assignment [p.13].

If the modulation range is high enough, we can even have the filter varying continuously from Low Pass to High Pass and back. Whether this is desirable or not depends on the aims and the audio source being processed.

Envelope and LFO modulations can have positive or negative values.

The default position of the filter Mode is Low Pass (knob points to the lowest point on the left, before the Band Pass switch position).

VCF filter Mode knob controller

Band Pass switch is at the lower left of the Mode knob, separated from the normal range. This is because this is a switch point (we cannot reach it by simply rotating the knob). Bear in mind that this is a fixed switchable point, therefore, when in the BP position, the Mode knob will not respond to modulation.

5.4. Frequency

Once we set the mode for the filter, it's time to set the base cutoff frequency. Remember that, unlike what happens in a Low Pass filter, this frequency will have different actions according to the selected filter mode. Therefore, if the filter mode is set, for example, to High Pass, and the cutoff frequency is set very high, we will hardly have any sound (only the frequencies above the cutoff point will pass without attenuation).

Also, if we have the filter at the Notch point (center) we will have just a narrow band cutoff. If we choose Band Pass (have the Mode knob at the switched BP position) then we will only have a narrow band of frequencies around the cutoff point audible.

> ♪: The Band Pass filter only lets a narrow range of frequencies pass. This, modulated by the Gate Sequencer and the Envelope may create great rhythmic patterns, if used with the Noise source, as we already saw.

Of course, since the slope of the filter is soft (just -12dB per octave), there are some frequencies above/below the cutoff point that will pass nonetheless, although attenuated.

So, with the filter mode control in the Low Pass region, we must have the cutoff frequency set to a relatively high value. On the other hand, if we have the filter mode control in the High Pass region, we must have the cutoff at a relatively low value. The lower the value, the more frequency bands that pass. This depends, of course, of the results one want to achieve.

Cutoff frequency range goes from 20Hz (when the control knob is pointing to the extreme left) to 15kHz (the control knob points to the extreme right). The default position is 15kHz (knob is pointing to the extreme right).

With the Frequency knob at its default position, and the filter Mode in its default position too, the filter lets you hear basically all the spectra of the signal coming through the Input. Only very high frequencies (over 15kHz) will be cutoff. The vast majority of audio will not have partials above this frequency. But we can still make a quick check, by activating the plug-in Bypass in the lower toolbar, or by rotating the Dry/Wet knob fully to the left.

Frequency cutoff can be modulated by the Envelope, the LFO or both. Modulation amount values can be positive or negative. We can also use several MIDI sources (again, see MIDI Learn Assignment [p.13] in Chapter 3).

5.5. Resonance

Resonance is a special feature present in synthesizer filter modules since almost the beginning. It is also one of the most appreciated timbre modification weapons. As we said earlier, the filter simply cuts frequencies above or below the cutoff point (depending if it's configured to Low Pass or High Pass) or even simply a range band (if it's configured for Notch, which is why it is also known as Band Reject).

What the Resonance control does is boost the frequencies around the cutoff point, creating a boosting peak bell around that point. This drastically changes the timbre of the sound. The bigger the amount of resonance, the higher the peak of the bell rises. In some filters, it may even start to self oscillate, producing a sine wave. It is not the case of the SEM-Filter though.

A high resonance value will create an audible effect in the sound, whose nature depends on the kind of filtering applied (if the filter is completely open, we will not get any effect from Resonance). This effect may or may not be pleasant, and may even be hurtful in some circumstances, so, beware when using resonance, and where and how you apply it.

So, if we have the filter set for Low Pass, the Frequency set for a relatively high value, and we set the Resonance to a value high enough (let's say 0.900), when we lower the cutoff value, we will start to hear a kind of ringing whistle, which corresponds to the higher partials being boosted.

This behavior is different if we configure the filter to High Pass, and then set the cutoff to a low value (a low value will allow more frequencies to pass the filtering, since it is configured for High Pass). Picking a drum track, for example, if we look for the region where the kick sounds and set the cutoff point in that region, by raising the resonance, we can achieve a really artificial and heavy punching kick.

> ♪: Since the SEM-Filter is so versatile, it is good to experiment with different kinds of filtering and resonance, especially with content that is spectrally complex, where we can use a more specific filter type, like Notch or Band Pass. Select a specific range of frequencies and apply a boost around the cutoff point using the Resonance. Dry/Wet control will help in achieving the ideal mix between processed and unprocessed sound.

Resonance values go from 0.00 to 1.00. This value doesn't have an absolute meaning. We may think of it like a percentage, where 0.00 is 0%, or no resonance, and 1.00 is 100% or full resonance. The default value for the Resonance is 0.00, or no resonance.

Of course, using a modulation source like the envelope is handy, especially when we are trying to achieve something like we described above. We can use the Envelope, the LFO or both. Modulation amount values can be positive or negative. We can also use several MIDI sources (see MIDI Learn Assignment [p.13]). We will get more into modulation when we check the Modulation Grid options.

5.6. Soft Clip

The Soft Clip adds 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. However, within a reasonable range, the induced saturation may sound very musical and pleasant.

> ♪: Try the Soft Clip with some audio with a wide dynamic range, like for example a drum track. In our tests, we got some great results just with the Soft Clip turned On and some gain added in the Input. Use the Dry/Wet knob to look for the ideal mix between the processed and unprocessed signal.

Soft Clip control is a toggle switch, which means it turns On or Off by simply clicking it. The switch can also be toggled by MIDI. By default, the control is Off.

5.7. Filter Out

The Filter Out knob adjusts the final output level of the filtered signal. Its range goes from -infinite to +24dB, so be careful, because it may cause audio clipping if set too high.

The Filter Out only affects the filtered signal, therefore, it will not have any effect on the Dry signal. It is a final volume control stage where we can compensate for some gain losses due to the filtering of the audio.

This controller can also be modulated by the Envelope, the LFO or both. We can also use MIDI, of course (see MIDI Learn Assignment [p.13] in Chapter 3).

5.8. Dry/Wet

The Dry/Wet control simply controls the amount of dry signal versus "wet" (processed) signal that goes to the plug-in output. So, we may think of it as a mixer control. Remember that the Dry signal reaches this stage unaltered, which means that, if we added some gain to the Wet signal, when mixing we can perceive a decrease in the overall volume. Again, we have relative values (this time even presented in percentage). The percentage shown refers the amount of processed (wet) signal that goes to the plug-in output. When the knob is completely turned to the right, we have 100% wet signal (only processed signal goes to the plug-in output). When we have the knob completely turned to the left, we have 0% wet signal (only the dry input sound goes to the plug-in output). Any value in between is a mix of the two signals. The higher the percentage value, the more processed signal we get in the output.

> ♪: If we turn the Dry/Wet knob completely to the Left (0% or Dry position), we bypass the entire plug-in process.

The default value is 100% (full wet). This parameter can only be modulated through MIDI.

6. THE ENVELOPE

In nature, sounds are not constant. They have an audio contour and partial content that changes over time, and that's what gives sound certain timbral characteristics, and makes the different sounds recognizable to us. Sound is produced by a body put in vibration by some kind of action (usually blowing, bowing, plucking or striking the vibrating body). We can easily identify some stages in almost all sounds:

1. The attack stage, where a lot of audio frequencies are produced when a certain action puts something in vibration (it's the vibration that produces the sound), and that differ according to the kind of physical body that was put in vibration, how the vibration was activated, and the kind of resonant/amplification body that the vibrating body is connected to (which can simply be the air).
2. A decaying stage, where the initial energy that created the vibration fades away, and the waves decay. In this stage, most partial waves fade completely, and only a few remain vibrating. This is the stage where the timbre of the sound changes the most, until reaching stabilization. The decaying stage is usually exponential, with a faster decay in the beginning, progressively slowing down.
3. A sustain stage (which may or may not be present, depending of the kind of vibrating body and the kind of action used to produce the vibration). In this stage, the number and kind of partials remains more or less stable, although sometimes we can identify some inner movement in this stage, usually some kind of "undulation" which can be tremolo or vibrato (two different phenomena that sound similar).
4. A release stage, which can be longer or shorter depending on the vibrating body and the resonant body. This also varies a lot, but eventually the vibration decays so much that we can't hear it anymore. In many sounds, Decay and Release stages are tied, as in a long segment, and we have no Sustain. That's the case, for example, of all plucked strings, the piano, or any mallet instrument.

All these stages occur within a single sound, and they can take more or less time, depending on the kind of sound (vibrating body) and how it is produced.

When synthesizers appeared, in the middle of the sixties, one of the first modules that was added was an envelope contour generator.

According to history, it was Vladimir Ussachevsky that gave Robert Moog the idea to build the Envelope Generator, to artificially produce a dynamic contour to the sound over time that would approximate the synthetic sounds of their natural counterparts. It was among the two that even the four important segments described above became somewhat standardized. Yet, apparently it was ARP that came up with the designation that became the synonym of Contour Envelope Generator: ADSR, the initials of Attack, Decay, Sustain and Release.

In fact, in the first Moog modular systems, the Envelope segments were labeled T1 (for the Attack segment), T2 (for the Decay segment) T3 (for the Release segment) and a fourth special "extra control" named Esus (from Envelope Sustain), this way separating that segment from the other, tempo based, segments.

So, the Envelope Generator became an essential part of any synthesizer since the very early days. Usually, the Modular Systems that were the first synths had several Envelope Generators. Not all of them were ADSR, though. Some were simpler (parts were not cheap in those days, everything was assembled by hand, and the more complex the circuit was, the more expensive it would be). So, sometimes we could have just a simple AD (Attack and Decay) envelope.

Some well known synthesizers, for example, only had three segment envelopes: Attack, Decay and Sustain, with a switch to activate the Decay segment again upon key release. We can think of the Release segment, as just another Decay, this time reaching silence, although with ADS envelopes we could not have independent slopes for Decay and Release segments.

The SEM featured this kind of ADS envelopes, a characteristic that was preserved in Arturia's SEM emulation, the SEM V.

However, for the SEM-Filter, Arturia introduced a slightly modified version of the envelope. We still have three segments, but we don't have a Decay after the Attack. Instead, we have a Hold segment, and the Decay is the last segment (therefore, we may think of it more as a Release segment, although these types of envelopes have been traditionally known as AHD, or Attack-Hold-Decay, or trapezoid envelope). One famous synth that had an envelope of this kind was the VCS3.

SEM-Filter Envelope section

Since the SEM-Filter is a processing plug-in, we have to remember that the audio to be processed already has its own (natural) envelope contour. So, the envelope is just for modulation purposes, and will "transform" the original audio envelope through filtering modulation. The LED next to the Envelope label will light whenever the Envelope is in action.

Each segment has a range between 0 and 10 seconds. The values indicated in the lower toolbar are real time values, in milliseconds. Attack and Decay segments are exponential, which means that they rise or fall exponentially. The Envelope needs a trigger signal to run, therefore, it will only work if it receives such a control signal. The only way to trigger the Envelope is via the Gate Sequencer. If the Envelope doesn't receive a trigger signal, no matter what destinations it is addressed to, it will not have any effect.

The Envelope can be applied to one or more of the following destinations: Filter Frequency Cutoff, Filter Resonance, Filter Mode, Input Noise, LFO Rate, LFO Amp, Env Amp (the envelope will modulate its own amplitude) and Filter Out.

Destination and modulation amount are defined in the Modulation Grid in the lower part of the Control Panel. Modulation amount can be positive or negative, varying between -99 and 99. These are relative values, where 99 or -99 are the max modulation amount (positive or negative).

> **NOTE: The Envelope needs a trigger signal to run. The only way to trigger the Envelope is via the Gate Sequencer**.

7. THE LFO (LOW FREQUENCY OSCILLATOR)

Like we can read in the title, LFO stands for Low Frequency Oscillator.

In the past (and we will travel again to the beginnings), there was no distinction between a regular oscillator (usually called VCO, from Voltage Controlled Oscillator) and a Low Frequency Oscillator.

This was because the VCOs could oscillate low enough to be used as LFOs, and a modular system would be more versatile this way - any oscillator could perform audio tasks or modulation tasks. This would lower the costs and give more power and flexibility to the user.

Actually, it seems that the first uses of an oscillator with a low frequency to modulate a signal were accidental. Synthesists became aware that patching a signal from the output of an oscillator into the input of another oscillator could create some great effects. And in those days, everyone felt adventurous and curious, and the strangest anyone could be the better.

So, they started messing with the modular systems, which basically allowed anything to be patched into anything. There was no distinction between audio signals and modulation signals.

Eventually someone found that by lowering the frequency of an oscillator and patching its output to one of the modulation inputs of another oscillator it was possible to produce vibrato (the base frequency of the modulated oscillator would vary according to the frequency and shape of the modulating wave). Actually, and just out of curiosity, this is the same principle of Frequency Modulation synthesis - it only differs in the frequency range of the modulator signal.

So, if that happened with the oscillators, then some similar effect could be achieved by doing the same modulation to the filter... and to the amplifier. As long as a module had modulation inputs, an oscillator output could be routed into them. All this was achievable with regular oscillators, so, the first modular systems didn't have dedicated LFO modules.

SEM-Filter LFO waves

But there was something that a regular VCO couldn't produce, and that was Sample & Hold. Sample & Hold is a kind of random sequence of pulses, created by capturing the voltage of a continuously varying analog signal, holding that value during a small period of time, defined by the rate parameter, and then capturing another voltage, and so on. This module, when routed to other modules, produces an equally random reaction on the module being modulated.

This function was so important that the original Moog catalog of modules, although it didn't have a dedicated LFO module, already had a Sample & Hold module. Even today, there are many Sample & Hold dedicated modules for modular synthesizer systems, including the ones inspired by Don Buchla, which gave Sample & Hold the poetic name of "Source of Uncertainty". This says enough about the importance this function had since the early days.

Anyway, we ended up having dedicated LFO modules, and eventually (later) had the Sample & Hold function incorporated in some of these modules (but only in self contained synthesizers). Curiously enough, even today, the vast majority of LFO modules destined for modular systems don't have Sample & Hold, which remains reserved to dedicated modules.

The Oberheim SEM didn't have Sample & Hold either, although it already had a dedicated LFO (a very simple one with just a sine wave).

That's not the case of the SEM-Filter though. Here, we have a modern complex LFO, with many waves to choose from, and, of course, with Sample & Hold.

The SEM-Filter LFO section

The SEM-Filter LFO is capable of oscillation in a range the goes from O.O1Hz (very, very slow) up to 2kHz (which is very, very fast), with a selection of waves comprised of sine, triangle, saw down, saw up (ramp), square, and finally Sample & Hold (S&H).

The sample rate may also be synchronized to the host, which is very important, because this way we will have modulations that are in sync with the audio/music. To activate synchronization, we have to click the Sync switch. As with the other switches in SEM-Filter, this one is also a toggle switch, which means it changes each time we click it.

When in sync, the rate of the LFO is controlled by the host. We therefore have the rate defined in music values, rather than frequency. We have many values to choose from, from the very, very slow 4/1 (which means four 4/4 bars or 4 whole notes), through 2/1 (two 4/4 bars or two whole notes), 1/1 (four beats, or whole note), 1/2 (two beats, or half note), 1/4 (one beat, or quarter note), 1/8 (eighth note), 1/16 (sixteenth note), and so on.

Of course, we also have dotted values, starting at half note. So, we have 1/2D (dotted half note). And we have triplet values also, starting at 1/4. Therefore, we have 1/4D (dotted quarter note), and 1/4T (triplet quarter note), 1/8D and 1/8T, 1/16D and 1/16T. From the 32nd note onwards, we will have only single values and triplet values, except when we reach the 1/128, which is the final value - therefore, no 1/128 triplet.

The shortest value available is 1/128, which means very, very fast (so fast that to be perceptible, the base musical tempo has to be really slow.

We finally have a Smooth parameter, to soften the transition between values (useful for example if we are using fast rates).

LFO can be used to modulate one or more of the following destinations: Filter Frequency Cutoff, Filter Resonance, Filter Mode, Input Noise, LFO Rate (the LFO will modulate its own rate), LFO Amp (the LFO will modulate its own amplitude), Envelope Amplitude, and Filter Out.

Destination and modulation amount are defined in the Modulation Grid in the lower part of the Control Panel. Modulation amount can be positive or negative, varying between -99 and 99. These are relative values, where 99 or -99 are the max modulation amount (positive or negative).

8. THE GATE SEQUENCER

The Sequencer was another module that was developed early on, and included in Modular Synthesizer Systems since almost the beginning.

What the Sequencer did was send triggering signals one after the other, in sequence (hence the name), coupled with control voltage signals. In the beginning these trigger signals were sent to the modules by some kind of controller device. These controller devices included keyboards, foot pedals, and also one or more special modules called... a Sequencer.

To play notes, an analog oscillator needed to receive two kinds of data: a control voltage to know which pitch to play, and a gate to trigger the sound. That's why we see many times the label CV/Gate, because keyboards and other devices, like the sequencers, usually sent these two kinds of signals. If the amplitude was controlled by an envelope, it's the envelope that needs to be triggered with a Gate, of course.

The Gate Sequencer, as its name implies, does not play notes, it just sends trigger messages.

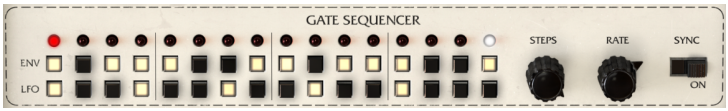
So, what kind of events are going to be triggered by the Gate Sequencer?

The Gate Sequencer is used to trigger two other modules in the SEM-Filter - the Envelope and the LFO. It isn't exactly used to "play" anything, like the name Sequencer could imply. In Fact, the Gate designation is extremely precise, because it's exactly what the sequencer does - it sends Gate signals to trigger the Envelope and/or the LFO.

However, its presence is vital for the SEM-Filter to work. In fact, the Envelope needs to be triggered to work, and although the LFO is free running (which means it will always act), if we want to have its phase reset at precise intervals, we will have to enable triggering. So, what the Gate Sequencer does to the LFO is restart its phase.

To sum it up, the Gate Sequencer is what makes the Envelope work, by triggering it, and also what makes the LFO phase restart.

> **NOTE: Bear in mind that, if no trigger data is sent to the Envelope, no matter what it is assigned to, we will not hear any effect. On the other hand, since it's free running, the LFO only needs triggering to reset phase.**



SEM-Filter Gate Sequencer

As you can see from the image pictured above. the Gate Sequencer controls are very simple. We have two rows of sixteen steps (it's a 16-step sequencer) labeled Env and LFO. The upper row (labeled ENV) controls the Envelope, of course, and the lower row controls the LFO. Each step is enabled by clicking it, and when enabled it becomes lighted.

The ending step is marked by a white LED, visible only when we click the Steps knob.

We then have two knobs, labeled Steps and Rate. The Steps knob controls the number of steps the Gate Sequencer will play (by default, it is positioned in 16 steps, which means all the steps will be played). We can have any number of steps enabled, from 1 to 16.

The Rate knob is a little more complex. Its behavior depends on the Sync switch. When the Sync switch is ON, the Gate Sequencer is synced to the host, and the Rate values are displayed in musical steps. We can sync it from 4/1 (meaning one step each four whole notes) up to 1/128 (128th note - a value so short that's almost never used in music, since to be perceptible, the music would need to be at a very slow tempo).

We have many values to choose from, from the very, very slow 4/1 (which means four 4/4 bars or four whole notes), through 2/1 (two 4/4 bars or two whole notes), 1/1 (four beats, or whole note), 1/2 (two beats, or half note), 1/4 (one beat, or quarter note), 1/8 (eighth note), 1/16 (sixteenth note), and so on.

Of course, we also have dotted values, starting at half note. So, we have 1/2D (dotted half note). And we have triplet values also, starting at 1/4. Therefore, we have 1/4D (dotted quarter note), and 1/4T (triplet quarter note), 1/8D and 1/8T, 1/16D and 1/16T. From the 32nd note onwards, we will have only single values and triplet values, except when we reach the 1/128, which is the final value - therefore, no 1/128 triplet.

When not in sync (Sync switch is OFF), the Rate value can be as low as 0.100Hz (very, very slow) and all the way up to 30.0Hz, which is already pretty fast. Anyway, the Gate Sequencer will probably be mostly used in Sync mode, because it's what makes more sense musically. To enable Sync mode, we will have to toggle ON the Sync switch.

Like all switches in SEM-Filter, this is a toggle switch, which means it changes position each time we click it. By default, it is OFF.

Gate Sequencer controls can be modulated by MIDI, like all the other controls. See MIDI Learn Assignment [p.13] in Chapter 3 to learn how to do it.

9. THE MODULATION GRID

The Modulation Grid is the last section of the SEM-Filter. It is placed in the lower part of the Control Panel.

It is a matrix with two rows of eight columns. Each row corresponds to one modulation source, and each column corresponds to one modulation destination. When the box displays "-", it means that there is no modulation assigned to that destination, by the source in the corresponding line.

SEM-Filter Modulation Grid

The top line is for the LFO, and the bottom line is for the Envelope. Modulation destinations can be Cutoff (Filter Frequency Cutoff), Res (Filter Resonance), Mode (Filter Mode), Noise (Input Noise), LFO Rate, LFO Amp (Amplitude of the LFO), Env Amp (Amplitude of the Envelope), and Filter Out.

Modulation values can be positive or negative, and go from -99 to 99. These values are abstract, but we can think of them as percentage values. This way, -99 would mean something like -100% which means full negative modulation. Likewise, 99 means something like 100% or full positive modulation. The neutral value is shown as a - sign.

All values can also be modulated by MIDI. See MIDI Learn Assignment [p.13] in Chapter 3 to learn how to do it. When adding MIDI modulation, this will modulate the amount of modulation already assigned in the grid cell. As we saw in the previous chapters, we also can assign MIDI modulation controls to any SEM-Filter parameter control directly.

Negative or positive modulation can act very differently, depending on the source and the destination. So, if we have, for example, the Envelope modulating the Filter Frequency Cutoff, with this one positioned in the lower part, and we assign a negative value to the Envelope modulation, we will probably not hear any result. But if you position the Cutoff knob at the center, you will hear a more or less fast close (depending on the envelope - for better results, put the Attack at 0 and give a generous amount of Decay).

♪: To have an idea of what we can do with the Modulation Grid controlling both the Envelope and the LFO, try the following:

1. Load a pad sound in your DAW, with four bars, and put it in Loop. Make sure the volume is low;
2. Place the Filter Mode in the Notch position;
3. Set the Filter Frequency around 2.2kHz;
4. Put the Resonance control at the maximum;
5. Put the Dry/Wet knob at full Wet (default position);
6. Now set the Envelope Attack to 0, Hold around 300ms and Decay around 1.2 seconds;
7. Set the LFO to Sine, put it in Sync, with a Sync Rate of 1/16 and Smooth around 0.020s;
8. Set the Gate Sequencer to four steps, put it in Sync mode with a Sync Rate of 1/4 (sync to the beat);
9. Activate all four Gate Sequencer beats for the Envelope and the first beat for the LFO;
10. Now, at the Modulation Grid, assign -99 from the Envelope to Cutoff, +75 from the LFO to the Resonance, -10 from the LFO to the Mode, +99 from the Envelope to the LFO Rate, +99 from the Envelope to the LFO Amp and +99 from the Envelope to the Filter Out;

You will hear the Pad undulating with a decreasing rate from the start of each bar, and a kind of soft muffled kick in each beat. All this was produced by the SEM-Filter. You can try changing values to see how that affects the overall sound. **NOTE: Remember that all values of the modulation grid can also be further modulated by assigning MIDI controls to them, which greatly expands the capabilities of the modulation.**

The Modulation Grid allows you to enter values directly, using the computer keyboard. To do this, we just need to click the cell we want to edit, type the value and press the Return or the Enter key. Remember that value ranges are from -99 to 99 (00 is assumed as -). To enter negative values, we have to start by typing the - signal.

This Grid allows for some very complex modulations, especially if we combine MIDI with the two modulation sources. Study the examples, study the presets, and let your imagination fly.

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