ULTRA ANALOG VA-3

USER MANUAL

A|A|S

Applied Acoustics Systems

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Contents

1 Introduction

Ultra Analog VA is a virtual analog synthesizer that combines into a modern instrument features of the legendary vintage synthesizers. *Ultra Analog VA-3* generates sound by simulating the different components of the synthesizer through physical modeling. This technology uses the laws of physics to reproduce how an object or system produces sound. In the case of *Ultra Analog VA-3*, mathematical equations describing how analog circuits function are solved in real-time. *Ultra Analog VA-3* therefore uses no sampling nor wavetable, it just calculates the sound as you play in accordance with the controls it receives. This sound synthesis method ensures unmatched sound quality, realism, warmth and playing dynamics.

Before discussing the synthesizer in more detail, we would like to take this opportunity to thank you for choosing an AAS product. We sincerely hope that this product will bring you inspiration, pleasure and fulfill your creative needs.

1.1 System Requirements

The following minimum computer configuration is necessary to run *Ultra Analog VA-3*:

Mac OS

- Mac OS X 10.7 or later
- Intel Core i5 (circa 2012) or faster
- 64-bit DAW

Windows

- Windows 7 64-bit or later
- Intel Core i5 (circa 2012) or faster
- 64-bit DAW

Keep in mind that the computational power required by *Ultra Analog VA-3* depends on the number of voices of polyphony and the sampling rate used. These computer configurations will enable you to play the factory sounds with a reasonable number of voices but performances will vary depending on your specific computer configuration.

1.2 Installation and Authorization

Installation and authorization of *Ultra Analog VA-3* is quick and easy. For the installation of our different products we use so-called *custom installers* which include both the program itself and your licence information. Installation and authorization can therefore be carried out automatically in a single step and from a single file when your computer is online. AAS products use a copy protection system based on a a proprietary challenge/response key exchange and therefore their authorization does not rely on other third party software and/or hardware.

In order to start the installation process, simply double-click on the installer file that you have downloaded. This will first install the program and then use the licence information included in the custom installer file to to carry out automatically the challenge/response procedure.

Once the installation is completed, you can check your licence information by starting the program and clicking on the chevron icon at the top of the interface. This will open a dialog box in which you should see your serial number and the email address which you used in order to get the installer file. Note that your serial number is also sent to you by email when your custom installer is created.

If your computer is offline when running the installer, or if the authorization procedure could not be completed for another reason, the dialog box will will not show your serial number and you will be prompted to authorize the program. In that case, click on the *Authorize* button and follow the on-screen instructions. Note that it is possible to use the program during 15 days before completing the authorization process. After that period, the program will not function unless it is authorized.

1.3 Getting Started

1.3.1 Using *Ultra Analog VA-3* in Standalone Mode

Ultra Analog VA-3 comes with a standalone versions allowing you to play it without having to open your sequencer. This can be convenient to explore *Ultra Analog VA-3* and its library, play it live or do some sound design work. To start *Ultra Analog VA-3* in standalone mode, simply follow the instructions below:

- **Windows** - Select *Ultra Analog VA-3* from the **Start** menu.
- **Mac OS** - Double-click on the *Ultra Analog VA-3* icon located in the Applications folder.

Before you start exploring the program, take a moment to set up you audio and MIDI configuration as explained below.

Audio and MIDI Configuration

Audio and MIDI configuration tools are available by clicking on the **Setup** button located in the lower left corner of the *Ultra Analog VA-3* interface. The **Setup** dialog first allows you to select an audio output device from those available on your computer. Multi-channel interfaces will have their outputs listed as stereo pairs.

On Windows, the audio output list is organized by driver type. The device type is first selected from the *Audio Device Type* drop-down list. If you have ASIO drivers available, these should be selected for optimum performance. The **Configure Audio Device button** allows you to open the manufacturer's setup program for your audio interface when available.

Once the audio input has been selected, you can then select a sampling rate and a buffer size from those offered by your audio interface.

The list of available MIDI inputs appears at the bottom of the dialog. Click on the checkbox corresponding to any of the inputs you wish to use.

1.3.2 Exploring the Factory Sounds

Ultra Analog VA-3 comes with a factory library which amounts to a huge range of sounds before you have even turned a single knob. As you would expect, the best way of coming to grips with the possibilities *Ultra Analog VA-3* offers is simply to go through the sounds one at a time.

A sound or preset is a stored set of parameters corresponding to a given sound. The programs are grouped and organized in *packs*. The names of the currently loaded pack and sound are displayed at the top of the interface.

One navigates among the different sounds with the associated drop-down menu which is opened by clicking on the sound name. One can also browse sounds by using the left and right arrows which appear to the right of the sound name. The computer keyboard arrows can also be used to navigate through sounds but this control must first be selected by clicking on the arrows or the sound name. The arrows then become surrounded by an orange line.

Sounds are managed using the *Sound Browser* which is revealed by clicking on the *Browser* button just below the preset name. Playing sounds and organizing them is pretty straightforward, please refer to Chapter 4 for a complete description of the pack and sound management operations.

1.3.3 Using *Ultra Analog VA-3* as a Plug-in

Ultra Analog VA-3 integrates seamlessly into the industry's most popular multi-track recording and sequencing environments as a virtual instrument plug-in. *Ultra Analog VA-3* works as any other plug-in in these environments so we recommend that you refer to your sequencer documentation in case you have problems running *Ultra Analog VA-3* as a plug-in. Note that in plug-in mode the audio and MIDI inputs, sampling rate, and buffer size are determined by the host sequencer.

1.4 Getting Help

AAS technical support representatives are on hand from Monday to Friday, 9am to 6pm EST. Whether you have a question on *Ultra Analog VA-3*, or need a hand getting it up and running as a plug-in in your favorite sequencer, we are here to help. Contact us by phone or email at:

- North America Toll Free: 1-888-441-8277
- Worldwide: 1-514-871-8100
- Email: support@applied-acoustics.com

Our online support pages contain downloads of the most recent product updates, and answers to frequently asked questions on all AAS products.

1.5 About this Manual

Throughout this manual, the following conventions are used:

- Bold characters are used to name modules, commands and menu names.
- Italic characters are used to name controls on the interface.
- Windows and Mac OS keyboard shortcuts are written as Windows shortcut/Mac OS shortcut.

2 Architecture of *Ultra Analog VA-3*

2.1 Signal Flow of the *Ultra Analog VA* Engine

The general architecture of the *Ultra Analog VA* synthesis engine is presented in Figure 1. The primary sound sources of the synthesizer are two oscillators and a noise generator. These sources are mixed before being sent to two different multi-mode filter modules in series with an amplifier.The signal from these two lines is then mixed and sent to a multi effects module. Note that the two filtering lines are not totally independent since the output of the filter module from the first line can be sent to the input of the filter on the second line.

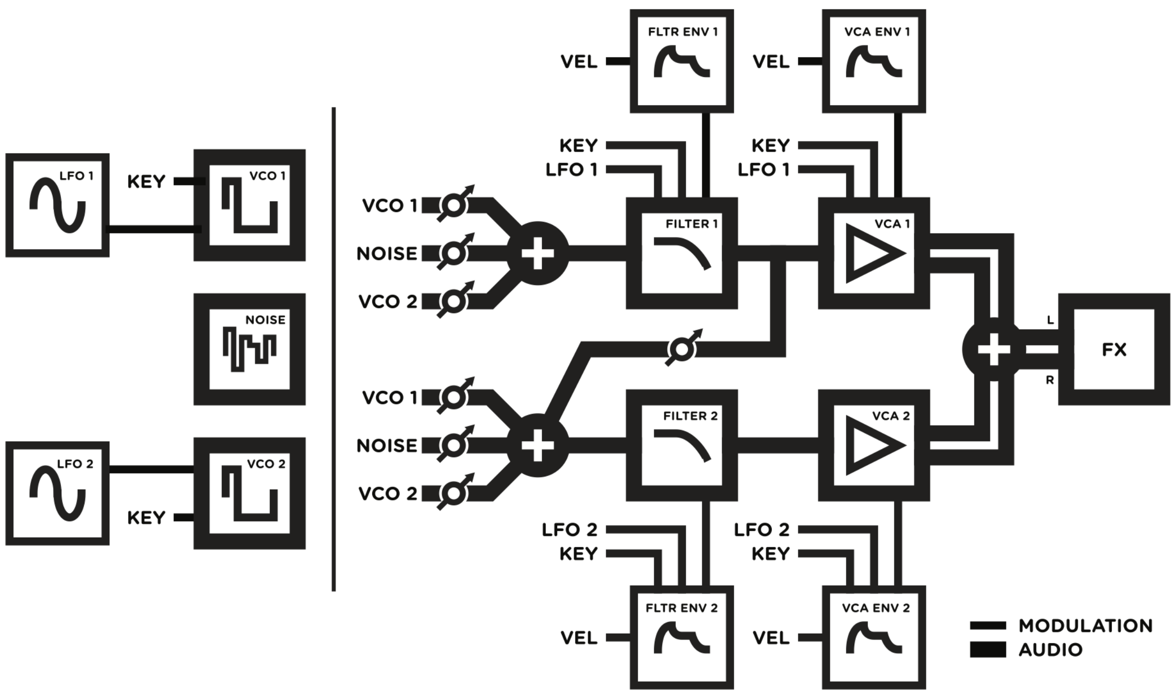


Figure 1: General signal flow of *Ultra Analog VA-3*.

This configuration is very flexible because, depending on the type of mixing applied, sources can be treated separately or in combination. For example, when the filters each receive signal from distinct sources, the right and left information from each source is preserved until the very end of the signal path. Sources can therefore be moved and positioned independently in the stereo space using the panning control on each of the amplifiers. On the other hand, combining the sources at the mixer level results in rich and complex tonal structures.

2.2 Multimbral Architecture

Ultra Analog VA-3 is a multitimbral synthesizer which can play two different timbres simultaneously either in layered or split keyboard mode. The general architecture of the synthesizer is shown in Figure 2 and consists of a MIDI routing module, two independent *Ultra Analog VA* engines in parallel, a mixer, and a multi effects module.

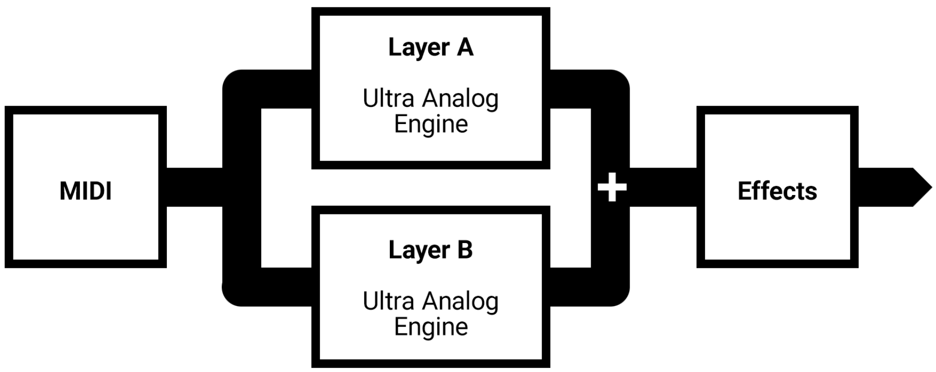


Figure 2: Multitimbral architecture of *Ultra Analog VA-3*.

2.3 Interface

The top part of the interface is called the *Utility* section and is shown in Figure 3. This section of the interface is always visible and will be described in more details in Chapter 7.

The center element of this section is a small sound browser which displays the currently loaded pack and sound. One can navigate through the pack by clicking on the sound name in order to open a drop-down menu with the list of sounds in the pack. One can also browse sounds by using the left and right arrows which appear to the right of the sound name. The computer keyboard arrows can also be used to navigate through sounds but this control must first be selected by clicking on the arrows or the sound name. The arrows then become surrounded by an orange line.

This section of the interface also includes a MIDI LED which is activated when the synthesizer receives MIDI signal and the master level meter allowing one to monitor the level of the output signal of the synthesizer. One also find buttons for the *Compare*, *Save*, *Save As*, and *History* commands.

The interface of the synthesizer is divided into four different views. Each one is accessed by clicking on the buttons labelled *Home*, *Browser*, *Editor*, and *Settings* respectively. We give a brief overview of these different views which will be followed by a more detailed description in the following chapters.

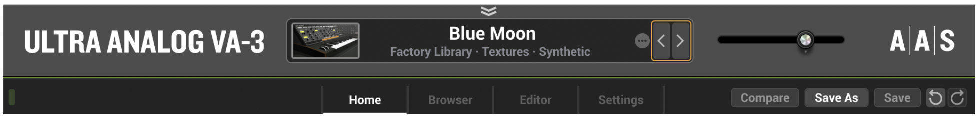


Figure 3: The *Utility* section.

2.3.1 The Home View

This is the main page for auditioning sounds and playing. The principal elements on this page are four controls allowing one to easily adjust high-level qualities or characteristics of the sound being played and modify it. These knobs, labelled *Modulation*, *Timbre*, *Envelope*, and *Effect*, control macro parameters in each layer which are mapped to specific synthesis parameters affecting the same characteristic of the sound. In other words, these controls add dimension by allowing one to obtain many different variations of a given sound. Controls on this view will be described in details in Chapter 3

Figure 4: The *Home* view.

2.3.2 The Browser View

Clicking on the *Browser* button reveals the sound manager. This view gives complete information on the sound library and this is where management tasks on sounds and sound packs can be carried out. A complete description of the sound manager is the subject of Chapter 4.

Figure 5: The *Browser* view.

2.3.3 The Editor View

Clicking on the *Editor* button gives access to the synthesis engine itself. The multitimbral architecture of *Ultra Analog VA-3* includes two identical instances of the synthesis engine in parallel, each corresponding to one layer or sound. The graphical user interface of each synth engine is identical and has been organized around three different pages as shown in Figures 8, 9 and 10. The *Synth* view also includes a layer mixer which is used to adjust the contribution of each layer in the resulting sound.

The first view, called the *Modes* view of the instrument, gives access to different performance parameters as well as to a step sequencer. The second and third views, called the *Synth* and *Effects* views respectively, are used for in-depth editing of the synthesis and effect parameters. One can switch from one view to the other by using the *Modes*, *Synth* and *Effects* tabs located just below the layer mixer.

The Layer Mixer

The layer mixer, shown in Figure 43, includes general controls which can be applied to each layer. It is also where the output level of each layer and the master output is monitored and adjusted using the different level meters and *Gain* sliders.

Each layer can be named using a label. In order to edit a label, click on it and type on the computer keyboard. Once a name has been entered, hit the *Return* key or click outside the label in

Figure 6: The *Synth* view.

order to deselect this region.

A layer can be switched *on* or *off* by clicking on the power switch icon located just before its label. Switching *off* a layer not only mutes its output but completely deactivates the synthesis engine of the layer. Note that in this case, the split layer feature of the keyboard is also deactivated. Each layer can also be muted or soloed by using the *M* and *S* buttons respectively. The level of a layer is adjusted using the corresponding *Gain* slider. The volume is adjusted by click-holding the mouse on the cursor and moving it. A specific level can also be reached at once by clicking directly on the slider rail. Note that it is possible to move the cursor of both layer by the same amount. In order to move the sliders together shift-click on a slider and move it.

The *Pan* knob is used to position the output of a layer in stereo space by adjusting the relative amplitude of signals sent to the left and right channels. When in its leftmost position, signal is only sent to the left channel while in its rightmost position signal is only sent to the right channel. When in its center position, an equal amount of signal is sent to both channels.

Each layer can also be transposed independently using the *Tune* controls. The adjustments are relative to the general tuning of the synthesizer which is specified in the *Settings* window as explained above. This control is composed of two numbers separated by a dot. The first number indicates a value in semi-tones while the second one indicates a value in cents (one hundredth of a semi- tone). The amount of transposition can be adjusted by click-dragging upward or downward on the semi-tone and cent controls. Double clicking on these controls brings back their value to zero.

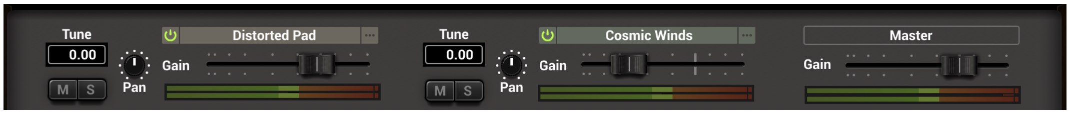


Figure 7: The layer mixer.

The Modes View

The lower section of this view includes a master clock, keyboard, unison, glide, vibrato and arpeggiator modules which will be described in more details in Chapter 5. Just below is a clickable seven octave ribbon allowing one to play different notes on the range of the piano which can be useful when no MIDI keyboard is connected to the computer.

Figure 8: The *Modes* view.

The Synth View

The *Synth* view gives access to the synthesis parameters described in details in Chapter 5 and allows one to really go under the hood.

The Effects View

The *Effects* view proposes a multi-effects module comprising an equalizer, a compressor, a reverb, and two additional effect modules. These five modules are in series and their respective order can be changed by click-holding on the handle on the left of the module and moving it to the desired position. The list of options for the two assignable effect modules includes a delay, distortion, chorus, flanger, phaser, wah wah, auto wah, a notch filter, a guitar amplifier, an equalizer, a compressor, a reverb, and a tremolo. The functioning of the effect modules is described in details in Chapter 5.

Figure 9: The *Synth* view.

Figure 10: The *Effects* view.

2.3.4 The Settings View

The *Settings* page is accessed by clicking on the *Settings* button. This is where general parameters and options for, polyphony, tuning, MIDI assigments and MIDI configuration are adjusted. The different setting options will be described in details in Chapter 6.

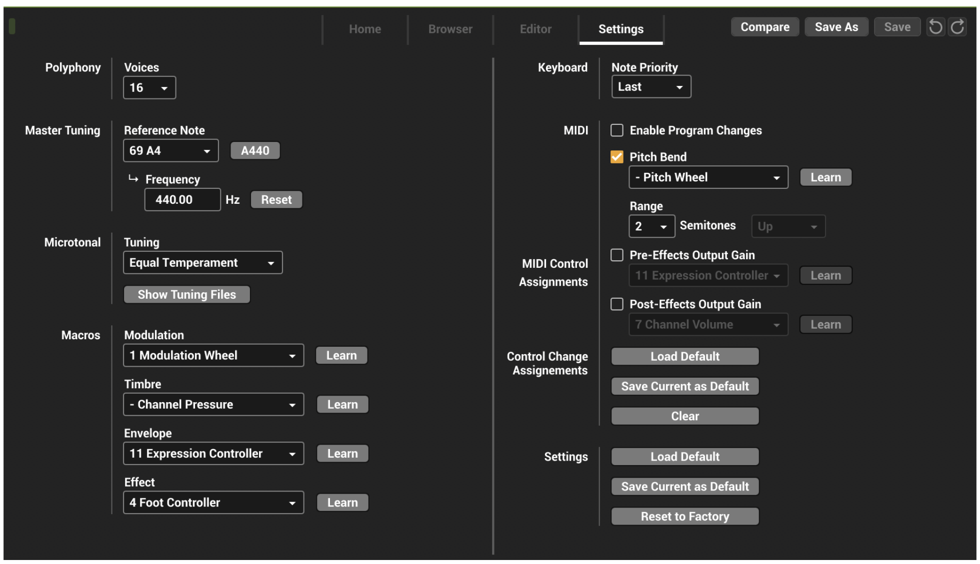


Figure 11: The *Settings* view.

3 The Home View

This is the main page for auditioning sounds and playing. The principal elements on this page are four controls allowing one to easily adjust high-level qualities or characteristics of the sound being played and modify it. These knobs, labelled *Modulation*, *Timbre*, *Envelope*, and *Effect*, control macro parameters in each layer which are mapped to specific synthesis parameters affecting the same characteristic of the sound. In other words, these controls add dimension by allowing one to obtain many different variations of a given sound. Note that the mappings to specific synthesis parameters are carried out at the layer lever as will be described in Chapter 5 when creating a sound.

Figure 12: The *Home* view.

The range of these knobs vary from -1 in their leftmost position to 1 in their rightmost position with a value of zero in their center position. The value of these parameter represents the amount of modulation applied to the synthesis parameters mapped to these macro controls. For example if the *Timbre* macro is mapped to the cutoff frequency of some filters, the value of this frequency is increased when the value of this knob is positive while it is decreased when the value is negative. When the knob is set to zero, the macro has no effect and the value of the mapped parameters is kept fixed. In our filter example, its cutoff frequency is not changed and remains the same as the one in the original sound. Note that the *Modulation* knob only varies between 0 and 1 since it is assumed that amplitude modulation or vibrato can only be added to a sound.

These four macro parameters can be assigned to external MIDI controllers for increased expressivity and playability. The mapping between the macros and the external controllers is carried out in the *Settings* view and will be described in Chapter 7. The name of the assigned MIDI controller is displayed below each macro knob and it is set to *None* when there is no assignment. When an external controller is used to control a macro, an orange line is displayed around the corresponding knob in order to indicated the actual value associated with the knob. If the knob is not set to a value of zero, the value received from the external control will be added to the value corresponding to the knob position. This is useful when a default value for the macro is desired.

The *Home* view also includes a virtual keyboard as well as a pitch bend wheel. This allows one to trigger sounds directly from the interface which is useful when no MIDI keyboard is connected to the computer.

4 The Browser View

In the context of *Ultra Analog VA-3*, a sound is a preset for the parameters of the entire synthesizer. Sounds are created by combining *layers*, each corresponding to a different instance of the synthesis engine. In this section we first review the browsing of sounds and their organization into *Sound Packs*. We then review the so-called *Layer Browser* which is used for the creation of new sounds. Finally we explain how to backup and share sounds and how to import sounds from *Ultra Analog VA-2*.

4.1 Sounds and Sound Packs

Sounds are stored in packs which basically act as folders. The name of the sound currently loaded is displayed at the top of the interface along with the name of the corresponding sound pack and its category as shown in Figure 13. The image associated with the currently loaded sound pack is also displayed to the left of the sound name.

The list of sounds in a given pack is revealed by clicking on the name of the sound. Clicking on a new name in the list loads this new sound into the synthesizer. One can also navigate through the list of sounds using the left and right-pointing arrows located on the right of the sound name. Note that after clicking on the name of the sound or selecting a new sound, the left or right-pointing arrows become surrounded by an orange line. This indicates that the arrows of the computer keyboard can also be used to navigate through the list of sounds. This feature is de-activated as soon as one clicks in another region of the interface or by using the *Escape* key on the computer keyboard.

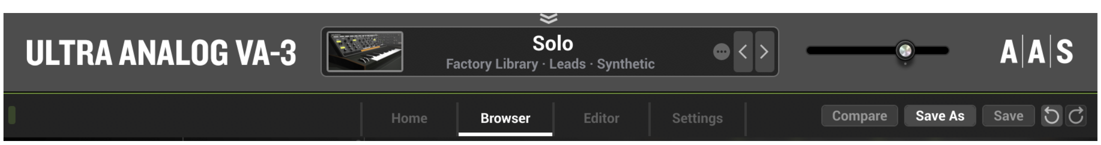


Figure 13: Currently loaded sound and sound pack.

4.2 Saving Sounds

Sounds are saved by clicking on the **Save** or **Save As** buttons located in the upper right corner of the interface (see Figure 13). When a sound has just been loaded, the **Save** button is greyed out and is therefore inactive. It is activated as soon as a parameter on the interface is modified. Clicking on this command replaces the stored version of the sound with the new one. Note that the **Save** command is never active when browsing AAS factory sounds since these are read-only and can not be modified. A new copy of the sound can be created, however, using the **Save As** command.

The **Compare** button, located just before the **Save** button, also becomes active as soon as a sound is modified. This command allows one to compare the modified version of a sound with the original version. This is useful when deciding if a sound should be replaced by a new version. Note that once the button has been switched *on* all further modifications to a sound are blocked. In order to allow edition again, the command must be switched *off*.

A new copy of a sound is saved by using the **Save As** command which is activated by clicking on the corresponding button which opens the **Save Sound** pop-up window. The name of the sound is entered at the top of this window. The destination pack is then selected. If necessary, a new pack can be created by clicking on the **New Pack** button. Sounds are saved with a *Category* and *Tone Quality* attribute. These are selected by using the corresponding drop-down lists. These attributes are useful for searches and display as will be described in the next section. These are followed by an entry for the name of the sound creator and finally a section for notes which can be useful for a description of the sound or playing indications.

4.3 The Sound Browser

Sounds and sound packs are managed using the **Sound Browser** which is opened by clicking on the **Browser** button in the top part of the interface (see Figure 13). Sounds can be browsed by pack, sound name, category, or creator by clicking on the tabs located in the upper left corner of the sound browser.

When browsing by packs, the list of available packs is displayed in the left section of the browser as shown in Figure 14. Packs are divided into two categories, AAS packs and user packs. AAS packs comprise the new *Ultra Analog VA-3* factory sounds, *VA-2* legacy factory sounds, as well as any expansion pack which might be installed on the computer. A pack is selected by clicking on its associated image in the left section of the browser. The AAS packs have read-only permission which means that their content can not be modified. The sounds from these packs can be edited but the new versions need to be saved into a user pack as will be explained in the section below. The list of user packs is always visible and appears below the *User* label in the left section of the browser. A user pack is loaded by clicking on its name which reveals the list of sounds included in this pack. Sounds in a pack can be organised by index number, name, category, or sound quality by clicking on one of these labels at the top of the sound list.

The entire library, including sounds from AAS and user packs, can be browsed by clicking on the *Sounds* tab in the upper left corner of the browser as shown in Figure 15 and which reveals the entire list of sounds in the library. Sounds can then be organised by name, category, sound quality, creator, pack, or date of creation. Similarly, the library can be browsed by sound category or sound creator, as shown in Figure 16 and Figure 17, by clicking on the **Categories** or **Creators** tab in the upper left corner of the browser. One then chooses a sound category or sound creator in the scrollable list which appears in the left part of the browser.

4.3.1 Managing User Packs

A new user pack is created by using the **Create Pack** command in the menu revealed by clicking on the ellipsis icon located to the right of the *User* label in the left section of the browser. One then

Figure 14: Browsing by sound pack.

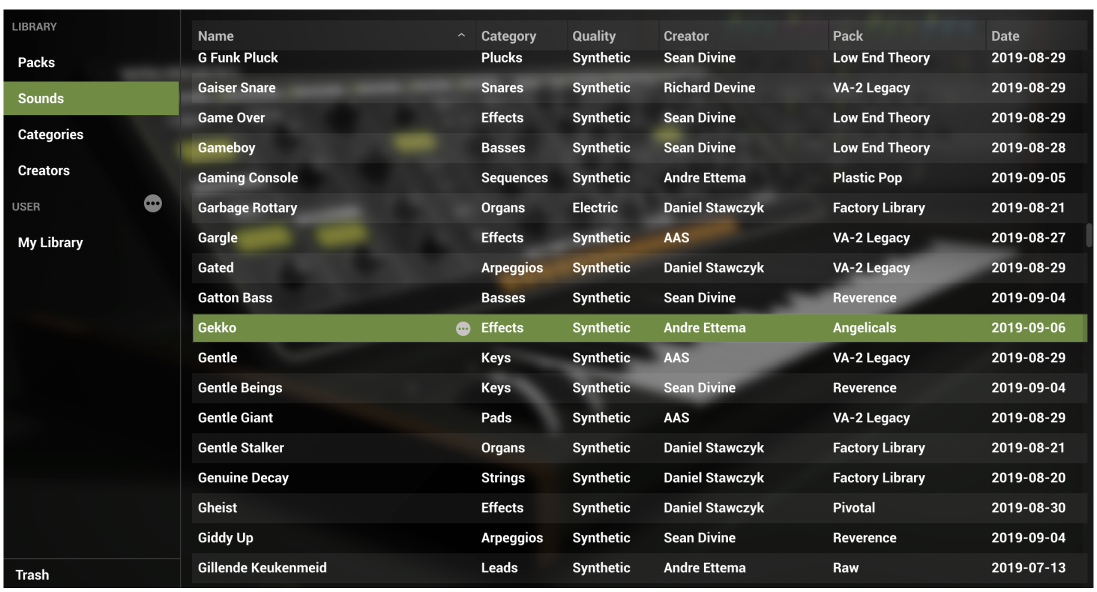


Figure 15: Browsing the entire library by sounds.

enters a name for the pack and clicks on the **Create** button. This creates an empty pack in the user pack list. Packs and the information corresponding to their sounds are stored as files in a *Packs* folder on your computer hard disk. In order to access this *Packs* folder, click on the **Show Packs Folder** command in the same menu. On the Windows operating system, this command will open

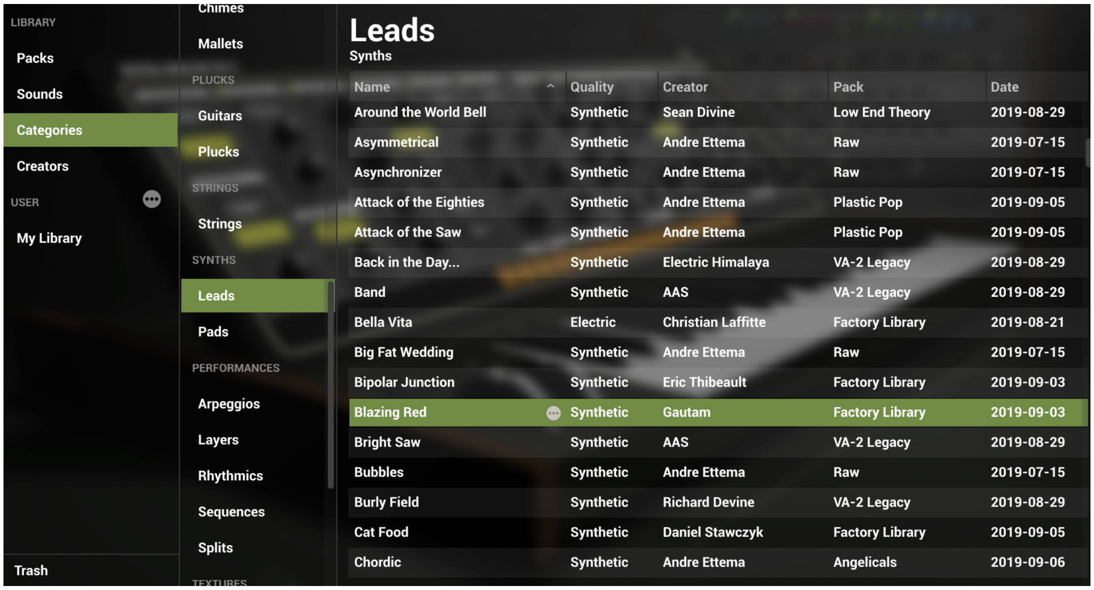


Figure 16: Browsing the library by sound category.

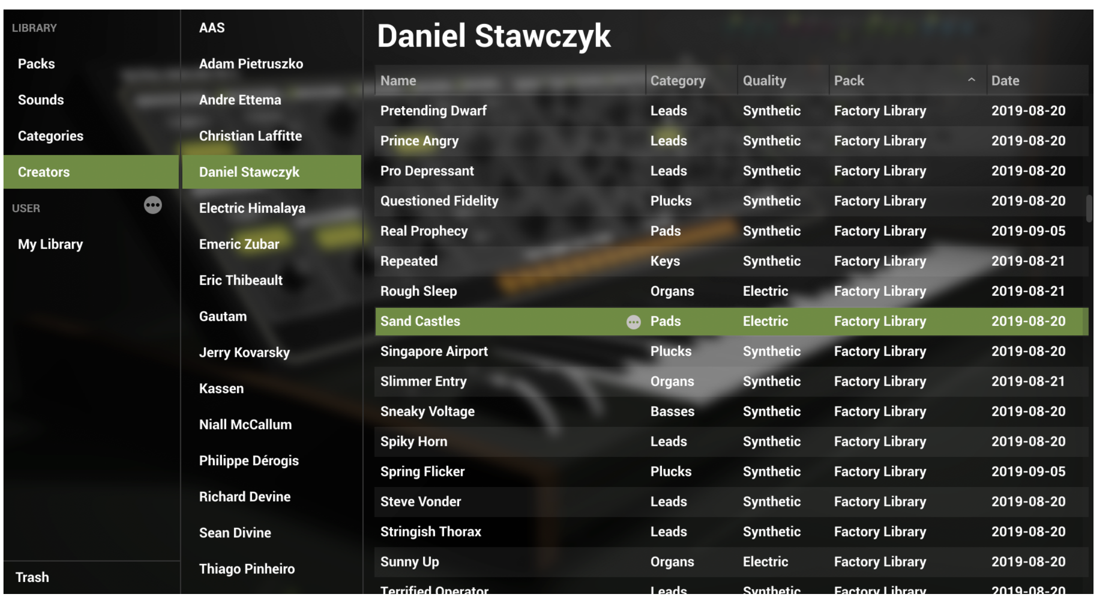


Figure 17: Browsing the library by creator.

an Explorer window at the location where the files are stored while on Mac OSX, the command opens a Finder window. All the pack file names follow the same format which consists of the pack name followed by the *VA-3 Pack* extension. These files can be used for backups or to exchange sounds with other users.

Different commands can be applied to a user pack once it has been selected. These are revealed by clicking on the ellipsis icon located to the right of the pack name at the top of the browser. A pack can be renamed by choosing the **Rename** command. A copy of a pack and its content is created by using the **Duplicate** command. The name of this new pack is entered in the pop-up window which appears after choosing this command. A pack is deleted by selecting the **Delete** command. Note that when a pack is deleted, it disappears from the browser and its entire content is not available anymore. A safety copy of the deleted pack is created however by *Ultra Analog VA-3* so it is always possible to recuperate a pack which has been deleted by mistake. The backup copy is accessed by using the **Show Packs Folder** command as explained above. At the same level as the *Packs* folder one finds a *Private* folder. The safety copies are located in the *Packs/Deleted* folder below the *Private* folder. In order to access a deleted pack again from the browser, copy it from the *Deleted* folder to the *Packs* folder.

4.3.2 Managing Sounds

A sound in a list is selected by clicking on its name or anywhere on its associated line in the list. It then becomes highlighted. Selecting a sound loads the corresponding preset data into the synthesizer and changes the current sound selection. A list of commands which can be applied to a selected sound appears by clicking on the ellipsis icon appearing to the right of the sound name. The **Add to Pack** command allows one to create a copy of the sound in a user pack. When this command is selected, the list of existing user pack is displayed and there is also the option to create a new one. Another way to create a copy of a sound in another pack is to use the **Copy** command, click on the name of the destination user pack and then by right-clicking in the sound list of this pack and using the **Paste** command.

Information for individual sounds is shown by clicking on the **Show Sound Details** command which opens the **Sound Information** display box. This includes the name of the sound, its category, tone quality, and its creator. All these fields can be modified directly from this information window. There is also a *Notes* field which is useful for entering a description or playing instructions. Note that the fields in this window can be edited for many sounds at once when using multiple selection of sounds.

A multiple selection consisting of adjacent sounds in a list is obtained by clicking on the name of the first sound to be selected and then, holding down the *Shift* key on the computer keyboard, and the clicking on the name of the last one. A non-adjacent multiple selection is obtained by holding down the *Ctrl/Command* computer key and clicking on the name of the different sounds to be selected. All sounds from a list can also be selected at once by using the **Select All** command from the command drop-down menu.

A sound can be copied to another pack by selecting it and using the **Copy** command. This command is available from the command drop-down menu or by right-clicking on the selected sound. The destination pack is then selected by clicking on its image and the sound copied by using the **Paste** command. A sound can also be moved to another pack by selecting it and then dragging and dropping it onto the image of a pack. Be careful however as this command, unlike the copy command, copies the selected sound to the destination pack but also deletes it from the original pack. This is only true however for sounds from user packs. AAS Packs can not be edited so drag and dropping sounds from these packs to a user pack is equivalent to a **Copy** command. Note that the **copy**, **delete**, and **move** commands can be applied on single sounds or multiple selections.

Sounds can be deleted from a user pack by using the **Move to Trash** command. This deletes the sounds from the pack but creates a safety copy in the so-called *Trash* pack which appears in the lower left corner of the browser. This might be useful when a sound has been deleted by mistake. This pack can be emptied from time to time by selecting it and deleting its sounds.

4.4 Backing up of Sound Packs

User packs are stored on disk as files which contain all the information corresponding to the sounds they include. These files can be displayed directly from *Ultra Analog VA-3* by opening the sound browser and using the **Show Pack Folder** command as explained above. The simplest way to create a backup of your packs is to make a copy on an external media of the above mentioned folders. Individual packs can be backed-up by making copies of individual pack files.

4.5 Exchanging Sound Packs

Sounds can easily be shared with other *Ultra Analog VA-3* users. This operation simply involves the exchange of the above mentioned user pack files. When a new pack file is copied to the pack folder on the destination computer, it is automatically available in *Ultra Analog VA-3*.

Note that individual sounds can not be exported. Sounds always appear inside a pack file. If you only wish to share a few sounds, create a new pack, copy the sounds you wish to exchange to this pack and share the corresponding pack file.

4.6 The Layer Browser

Sounds in *Ultra Analog VA-3* consist of one or two layers, each layer corresponding to one instance of the *Ultra Analog VA-3* synthesis engine. Sounds can be modified by changing individual parameters in the *Synth* section of each layer but they can also be changed by loading presets for the entire synthesis engine corresponding to each layer. Presets for each layer slot are loaded using the *Layer Browser*, shown in Figure 18. It is opened by clicking on the **Browse** command revealed by clicking on the ellipsis icon to the right of each layer name in the layer mixer.

Layer presets from library sounds are browsed by pack. The list of layers included in the selected pack are displayed on the right of the pack list. Layers are organised in sound categories and are listed using the name of the sound they come from and their associated slot (layer A or layer B). One can jump from one sound category to another by using the category drop-down menu at the top of the layer list. Layers can have a label but this is optional. When a layer has no label

Figure 18: The layer browser.

the default labels *Layer A* and *Layer B* appear in the layer mixer. One can navigate in the layer list by scrolling up or down the list or using the arrows in the top right corner of the *Layer Browser*.

A layer is loaded as soon as it is selected so it is possible to audition the effect of different layers by browsing the layer list. In order to hear the changes, make sure the power switch of the corresponding layer, located just before the label name in the layer mixer, is in its *on* position. It may indeed be in its *off* position if you are editing a sound that initially had just one layer slot used. Once a choice has been made, one clicks on the *OK* button in order to replace the content of the layer in the current sound. Clicking on the *Cancel* button closes the *Layer Browser* and reverts to the original version of the layer. If a new layer has been chosen and you are satisfied with this new version of a sound click on the *Save* or *Save As* button in the top right corner of the interface in order to save the changes.

For convenience, layer presets which you often use can be stored separately in a user section. These can simply be copies of existing layers or layers which were modified by tweaking the parameters in the *Play*, *Synth*, and *Effects* section of the synth engines and which you wish to keep for use in other sounds. A layer preset is created by using the **Save** command in the layer command menu opened by clicking on the ellipsis icon to the right of the layer name. In the same way, a preset is loaded into a layer slot by using the **Load** command.

4.7 Importing Sounds from Previous versions of *Ultra Analog VA*

Ultra Analog VA-3 includes a converter that allows one to import sounds from *Ultra Analog VA-2*. The conversion operation simply involves copying an *Ultra Analog VA-2* pack file into the *Ultra*

Analog VA-3 sound pack folder. The conversion operation is then triggered automatically when *Ultra Analog VA* detects a pack from a previous version.

Ultra Analog VA-2 sound packs, which were then actually called banks, can be found by opening *Ultra Analog VA-2*, clicking on the *Manage* button at the top of the *Ultra Analog VA-2* interface in order to open the manager, and then clicking on the *Show Files* button at the bottom of the manager window. On the Windows operating system, this command will open an Explorer window at the location where the *Ultra Analog VA-2* sound pack files are stored while on Mac OSX, the command opens a Finder window

Once you have the *Ultra Analog VA-2* files you wish to import, go back to *Ultra Analog VA-3*, click on the **Browser** button in order to open the browser, click on the ellipsis icon next to the *User* label in the left part of the browser and choose the *Show Packs Folder* command. Again, this will allow you to access the location where sound pack files are stored using a Finder or Explorer window on Mac OSX or Windows respectively. Copy the *Ultra Analog VA-2* files to be converted to this location and they will be automatically converted to *Ultra Analog VA-3* sound packs and appear in the browser.

While the great majority of sounds should be recuperated without noticeable differences, the conversion program is not infallible which means that some sounds might need some readjustments after the conversion. This is due to the fact that the mapping of the parameters from different versions of *Ultra Analog VA* is not direct as a result of changes in the architecture, modules and the effect section between the different versions.

Note that AAS expansion sound packs for *Ultra Analog VA* which were installed on your computer prior to the installation of *Ultra Analog VA-3* should not be converted in this way. The *Ultra Analog VA-3* installer you will have downloaded from our server should indeed also include your expansion sound packs and take care of their installation automatically. If this is not the case, or some packs are missing, please go back to your account on the AAS user portal and download the latest installer for these sound packs, they have indeed all been updated and optimized for this new version of *Ultra Analog VA*.

5 The Editor View

This section can be used as a reference for the controls appearing on the different modules of the synthesis engine of *Ultra Analog VA-3*. These modules are accessed from the *Editor* view which is displayed by clicking on the *Editor* button located at the top of the interface.

This synthesizer is two-voice multitimbral and is therefore based on two instances of the synthesis engine. These two instances are identical and this documentation therefore applies indifferently to both layers. The different modules corresponding to each layer are grouped under a *Modes*, *Synth*, and *Effects* view each accessed by clicking on the corresponding tab.

We begin by describing the behavior of the different types of controls appearing on the interface and then describe the parameters of each module of the synthesizer.

5.1 General Functioning of the Interface

5.1.1 Knobs

The synthesizer parameters are adjusted using controls such as knobs, switches and numerical displays. A specific control is selected by clicking on it. A coarse adjustment is obtained by click-holding the parameter and moving the mouse, or the finger on a track pad, either upwards and downwards or leftwards and rightwards. The value of the parameter replaces its label while it is being adjusted.

Fine adjustment of a control is obtained by holding down a modifier key of the computer keyboard (Shift, Ctrl, Command or Alt key) while adjusting the parameter.

Double clicking on a knob brings it back to its default value when available.

5.1.2 Switches

Switches are turned *on* or *off* by clicking on them. They are used to activate or deactivate modules and the *sync* feature of some parameters.

5.1.3 Drop-down Menus

Some displays reveal a drop-down menu when clicking on them. Adjustment of the control is obtained by clicking on a selection.

5.1.4 Modulation Signals

Different parameters can be modulated by the signal from different sources including the MIDI keyboard, envelope generators and the **LFO** modules. Modulation signals are controlled with small gain knobs located below the corresponding modulated parameters.

The *Key* modulation knobs are used to modulate a parameter depending on the note played on the keyboard. When in its center position, the value of the corresponding parameter is equal across the whole range of the keyboard. Turning the knob to the left increases the value of the parameter for low notes while decreasing its value for high notes. The variations are applied relative to the middle C (MIDI note 60) whose value is always that corresponding to the settings of the actual parameter knob. Turning the modulation knob to the right has the opposite effect and increases the value of the parameter for high notes while decreasing it for low notes.

The *Vel* modulation knobs are used to modulate the value of a parameter depending on the velocity signal received from the keyboard so that the value of a parameter increases as notes are played harder on the keyboard. The position of the knob is used to adjust the amount of modulation applied to the parameter. In its leftmost position, the modulation source is turned *off* and the value of the parameter does not vary with the velocity signal from the keyboard. Turning the knob clockwise increases the effect of the modulation signal on the value of the parameter.

Modulations using the signal from the **LFO** and **Env** modules are controlled using the *LFO* and *Env* gain knobs respectively. The amplitude of the modulation is zero when the knob is centered. It is increased by moving it from its middle position clockwise or counter-clockwise. When turning it counter-clockwise, the phase of the modulating signal is inverted while it is preserved when moving it clockwise.

5.1.5 Synchronisation

The rate of the **Arpeggiator**, **LFO** and certain effect modules can be synchronized to the clock of a host sequencer when the program in used in plug-in mode. To do so, simply turn *on* their *Sync* switch. Synchronization values are adjusted with the *Sync Rate* parameter and range from 4 whole notes (16 quarter notes) to a thirty-second note (1/8 of a quarter note) where the duration of the whole note is determined by the host sequencer clock. The effect can also be synced to a triplet or dotted note. To adjust this parameter, click on the *Sync Rate* button and choose a rate value from the drop-down menu.

In standalone mode, when the *Sync* switch of an effect of module is switched *on*, the duration of a whole note is adjusted using the *Rate* control of the **Clock** module on the **Play** view.

5.1.6 Module Commands

All modules in the *Play*, *Synth*, and *Effects* views share a common command menu which is opened by clicking on the ellipsis icon located on the right of the module label. The *Copy* command is used to copy the settings of a module and the *Paste* command is used to paste them onto another instance of the module in either layer of a the current sound or in another sound.

Settings which are used often can be saved as presets by using the **Save** command. One then chooses a name and the presets is then created. Presets for a module are loaded using the *Load* command which opens the module preset window as shown in Figure 19. One navigates through the list of available presets by scrolling up or down the list or using the arrows in the top right corner of the window. A preset is loaded once it is selected. It is possible to compare the new settings with the original ones by clicking on the *Revert* button in the right section of the window. A preset can also be renamed or deleted by using the *Rename* and **Delete** buttons respectively. Once a preset has been chosen, it can be applied to the current module by clicking on the **OK** button. This closes the window and applies the new settings to the module. Clicking on the **Cancel** button closes the window but reverts to the original settings of the module. Note that if the settings of the module have been changed following a **Load** command, the **Save** command in the top left corner of the interface must be used in order to save this new version of the sound.

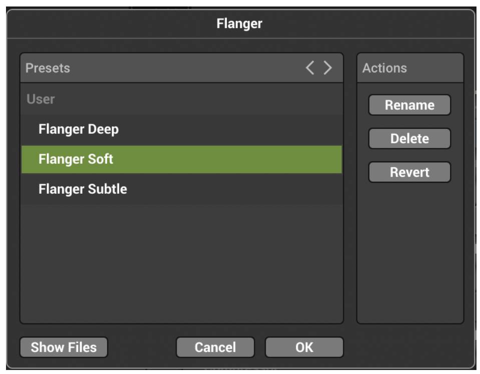


Figure 19: The module preset window.

Presets for modules are kept in text files on your computer disk. Clicking **Show Files** button in the lower left corner of the module preset window opens a Finder or Explorer window on Mac OSX or Windows respectively at the locations where these files are kept. This might be useful for backuping presets or exchanging them with other users.

5.2 The Modes View

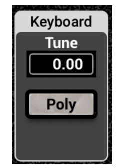
The **Play** view is where the main performance oriented modules are located. Key parameters from the **Synth** and **Effects** view are also included for quick access. This view is loaded when starting the instrument and can be accessed from another view by clicking on the *Play* button on the top part of the interface.

5.2.1 The Clock Module

This module is used to control the tempo of the different effects of the FX section as well as that of the **LFO** and **Arpeggiator** modules when their respective *sync* button is switched *on*. When *Ultra Analog VA-3* is launched in standalone mode the clock tempo, in bpm, is set by using the *Rate* knob. The tempo can also be adjusted by clicking at the desired tempo on the *Tap Tempo* pad of the module. Once the new tempo is detected, the value of the *Rate* knob is automatically adjusted.

When using *Ultra Analog VA-3* in plugin mode, the *Tap Tempo* pad is replaced by a *Sync To Host* switch. In its *on* position, the rate is synchronized with that of the host sequencer. When switched *off*, the tempo is determined by the value of the *Rate* knob.

5.2.2 The Keyboard Module



The **Keyboard** module controls how the synthesizer voices respond to the events coming from an external MIDI keyboard or from a MIDI sequencer.

The keyboard can be monophonic, allowing one to play only one note at a time, or polyphonic, allowing one the play chords. This behavior is adjusted using the *Poly* button. The keyboard is in polyphonic mode when this button is switched *on*.

The *Tune* control is used to transpose the frequency of the keyboard. This control is composed of two numbers separated by a dot. The first number indicates a value in semi-tones while the second one indicates a value in cents (one hundredth of a semi- tone). The amount of transposition can be adjusted by click-dragging upward or downward on the semi-tone and cent controls. Double clicking on these controls brings back their value to zero. When the value of the Tune parameters is set to 0.00, the frequency of notes are calculated relative to A4 with a frequency of 440Hz.

5.2.3 The Unison Module

The unison mode allows one to stack voices, in other words, play two or four voices for each note played on the keyboard. This mode creates the impression that several instruments are playing the same note together, adding depth to the sound. It is switched *on* by clicking on the LED located in the upper right corner of the module.

Each voice can be slightly detuned relatively to the others by using the *Detune* knob. Turning this knob clockwise increases the amplitude of the error. Furthermore, voices can be desynchronized by adding a small time lag between their triggering with the *Delay* knob. There is no delay when the knob is in its leftmost position and it increases (units in ms) as it is turned clockwise.

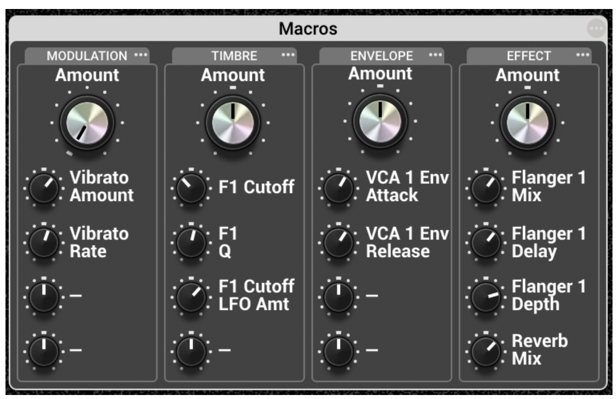
5.2.4 The Glide Module

The **Glide** module is used to make the pitch slide between notes rather than changing immediately from note to note. The *Time* knob sets the amount of time necessary for the pitch to slide over one octave. The *Mode* drop-down menu enables one to choose between the *Constant* or *Proportional* mode. When in *Constant* mode, the time necessary for the pitch to slide from one note to another is always the same regardless of the interval between the notes. When set to proportional, the slide time becomes proportional to the width of the interval between the two notes.

Clicking on the *Legato* button switches the module into legato mode and the sliding between two notes then only occurs if the second note is played before the first one is released. When a note is played staccato, or in other words if a key is released before the next one is depressed, there is no glide effect. Note that even though the glide effect is available when the **Key** module is in polyphonic mode, it is mostly dedicated to monophonic playing. In polyphonic mode, the sames rules apply to individual voices and the overall result is less predictable.

5.2.5 The Macros Module

The **Macros** module includes four different macros associated with different high-level qualities or characteristics of the sound. These macros are used to easily control general aspects of the sound and therefore rapidly obtain different variations. These modules, labelled *Modulation*, *Timbre*, *Envelope*, and *Effect*, correspond to the macro controls found on the *Home* view introduced in section 2.3.1. The modules are used to create the assignments between specific synthesis parameters and each macro and control the amount of possible modulation for each parameter.



The *Modulation* macro is used to control the amount of modulation in the sound and it is usually assigned to parameters from the vibrato module (frequency modulation) or the tremolo effect module (amplitude modulation). The *Timbre* macro is used to modify the tone quality of a sound and is typically mapped to parameters from the filter modules of the synthesizer. The *Envelope* macro is used to adjust the time-domain characteristics of the sound and make it shorter or longer for example. This parameter is usually mapped to the parameters from the different ADSR modules of the synthesizer. Finally, the *Effect* macro allows one to controls the amount of effect applied to the sound. This knob is typically mapped to key parameters of the effect modules used in each sound. Note that there is formally no restriction on the parameters which can be assigned to a macro and they do not necessarily need to correspond to the macro label. These labels have been chosen as indications and corresponds to sound qualities which are normally most relevant in the majority of sounds. Assignments for the sounds from the factory library have been chosen to follow these categories as much as possible.

Up to four synthesis parameters can be assigned with each macro. Assignments are created by clicking on the label located above the four small *range* knobs in each macro module (or the small horizontal line when there is no assignment yet). This reveals a drop-down menu with a list of possible destination parameters. These *range* knobs are used to specify the amount of allowed modulation of the corresponding destination parameter. Their range varies between -100 % in their leftmost position and 100 % in their rightmost position with a value of 0 % when in their center position. This value represents the maximum possible percentage of variation of the destination parameter, in terms of its range, around its current value. A value of 100 % therefore means that the destination parameter can vary over its full range while a value of zero means that its value will remain fixed. Negative values are used to invert the direction of the variation, in other words decrease the value of the destination parameter when a positive modulation signal is received and vice versa for negative modulation signals.

At the top of each macro module is an *Amount* knob which controls the amount of modulation applied to the destination parameters associated with the macro. These controls are gain knobs which behave exactly in the same way as those on the *Home* view and described in Chapter 2.3.1. The range of these knobs vary from -1 in their leftmost position to 1 in their rightmost position with a value of zero in their center position. The only exception is the the *Modulation* knob which varies between 0 and 1.

The actual amount of modulation applied to a destination parameter is determined by the multiplication of the value of the *Amount* knob and that corresponding to the range of the destination parameter. Since the value of the *Amount* knob varies between -1 and 1, it can be interpreted as a fraction of the allowed range of the destination parameter. When set to a value of 1, the amount of modulation applied to the destination parameter will be equal to value set by the range parameter, when set to 0.5 it will be equal to half the value allowed by range knob , with a value of zero the modulation will be nil and the value of destination parameter will remain unchanged. The effect is the same for negative values but the modulation of the value of the destination parameter is inverted. Note that the value of the *Amount* know associated with the *Modulation* macro can only be positive because it is assumed that a vibrato or amplitude modulation effect can only be added to a sound.

In the case where the *Timbre* macro is mapped to the cutoff frequency of some filters, for example, the total amount of variation of the cutoff frequency is determined by the value of the corresponding *range* knob. If set to 0.25 for example, the maximum allowed amount of variation will correspond to a quarter of the full range of this parameter. When the *Amount* knob is in its zero position, the frequency remains unchanged but it starts to increase when the knob is turned clockwise. When the *Amount* knob reaches a value of 1, the value of the cutoff frequency will have been increased by a quarter of the parameter range, in other words a quarter of the maximum possible value for this parameter. If the *Amount* knob is set to negative values, the cutoff frequency will be reduced in a similar manner.

A good way to adjust these parameters is to set the *Amount* knob to its maximum value by turning it fully clockwise and then adjusting the value of the range knobs to get the maximum desired amount of modulation effect. The *Amount* knob is then set back to its zero position.

Note that there is a relationship between the *Amount* knobs of the **Macro** modules and those on the *Home* view. The amount of modulation signal applied to a destination parameter corresponds to the sum of the values specified by these two knobs. As the modulation knobs on the *Home* view can be assigned to external MIDI controllers, the total amount of modulation signal is in fact equal to the sum of the values of these two knobs plus that associated with the value sent by the mapped external controller. An orange line is displayed around the *Amount* knobs in order to indicate the total amount of the modulation signal applied to destination parameters. Mapping of **Macro** modules to external MIDI controllers is described in Chapter 7.

One may wonder why there is an *Amount* knob both on the *Home* and *Synth* views. The reason for this is that the *Amount* knobs on the *Home* view affect both layers of the synthesizers at the same time while those in the same view only affect individual layers. The knobs in the *Synth* view are useful when using saved layers to create sounds. A certain amount of modulation may then be desired in a given layer and the corresponding *Amount* control would then be used. In this way the *Amount* controls on the *Home* view, and eventually external MIDI controllers, would affect the *Macros* of both layers differently. Note that macros for a given layers can be disabled at once by using the controls in the *Layer Settings* window which is opened by clicking on the gear icon located next to the *Split* button in the lower right corner of the *Play* view of each layer.

As a last remark on *Macro* modules, we mention that the list of destination parameters for macros include *Pre* and *Post-effect Gains*. These two parameters are useful to control the overall level of the sound when modulation is applied. Indeed the level of the sound may vary when, for example, changing the cutoff frequency of a filter or modulating the depth of an effect. These gain parameters can then be used to compensate the level variation and keep the volume constant.

5.2.6 The Vibrato Module

The vibrato effect is equivalent to a periodic low frequency pitch modulation. This effect is generally obtained by using an LFO to modulate the pitch signal of an oscillator. In *Ultra Analog VA-3* , a dedicated module is provided for this effect. The *Rate* knob sets the frequency of the vibrato effect from 0.3 Hz to 10 Hz. The *Amount* knob sets the depth of the effect, or in other words the amplitude of the frequency variations. In its leftmost position, there is no vibrato and turning the knob clockwise increases the amount of pitch variation.

The vibrato can be adjusted not to start at the beginning of a note but with a little lag. This lag, in seconds, is set by the *Delay* knob. The *Fade* knob allows you to set the amount of time taken by the amplitude of the vibrato effect to grow from zero to the amount set by the *Amount* knob.

5.2.7 The Arpeggiator Module

The **Arpeggiator** module allows one to play sequentially all the notes that are played on the keyboard. In other words, arpeggios are played rather than chords. The modules allows one to produce a wide range of arpeggios and rhythmic patterns and to sync the effects to the tempo of an external sequencer.

Arpeggio Patterns

The arpeggio pattern is set by the combination of the value of the *Range*, *Span* and *Order* controls. This module is switch *on* by clicking on the small LED located in the upper right corner of the module.

The *Range* control is used to select the number of octaves across which the pattern is repeated. When the range is set to 0, there is no transposition and only the notes currently depressed are played. If set to a value between 1 and 4 (its maximum value), the notes played are transposed and played sequentially, over a range of one or more octaves depending on the value of the *Range* parameter. The direction of the transposition is set with the *Span* drop-down menu. This parameter can be adjusted to *Low* for downwards transposition, to *High* for upwards transposition or *wide* for transposing both upwards and downwards. Finally, the *Order* control sets the order in which the notes are played, therefore determining the arpeggio pattern. When set to *Forward*, the notes are played from the lowest to the highest. When set to *Backward* the notes are played from the highest to the lowest. In the two last modes, *Rock and Roll exclusive* and *Rock and Roll inclusive*, the notes are played forward from the lowest to the highest and then backward from the highest down to the lowest. When using the *RnR exclusive* mode, the highest and the lowest notes are not repeated when switching direction but in RnR inclusive mode these notes are repeated. Finally, in *Chord* mode, all the notes are played at once.

Rhythmic Patterns

Rhythmic patterns can be added to the arpeggio pattern by using the 16-step *Pattern* display. Notes are played as the step display is scanned and the corresponding step is selected (red button *on*). Notes are played regularly when all the steps of the display are turned *on* and rhythmic patterns are created by selecting only certain steps. The arrow button below each step is used to fix looping points from which the rhythmic pattern starts being played again from the beginning.

Rate and Synchronization

The rate at which the arpeggiator pattern is scanned is set by the *Rate* knob of the **Arpeggiator** module or can be synced to the master clock of the *Clock* module. The *Rate* knob is only effective when the *Sync* control is set to *off*. When the *Sync* control is *on*, the rate (tempo) is fixed by the master **Clock** module (see 5.2.1) in standalone mode or the host sequencer in plugin mode. The rhythmic value of each step is set using the *Steps* parameter. Values can range between a quarter note and a thirty-second note with binary and ternary beat division options. One can then fix the metric of the pattern by setting the loop point of the step display appropriately.

Latch mode

The **Arpeggiator** module is toggled in latch mode by clicking the *Latch* button to its *on* position. In this mode, the **Arpeggiator** keeps playing its pattern when the notes on the keyboard are released and until a new chord is played.

5.2.8 Ribbon

The lower part of this view includes a ribbon controller. The ribbon covers seven octaves and notes are played when clicking on the ribbon. The ribbon is useful to test sounds when no MIDI keyboard is connected to your computer.

5.2.9 Split Keyboard

When two layers are used in a sound, one can enable the split keyboard mode in order to play them in different regions of the keyboard. This mode is activated by clicking on the *Split* button. When this mode is activated, a coloured line appears above the ribbon keyboard of the interface in order to indicate the range of each split region of the keyboard. The left portion of the keyboard is associated with layer A while the right portion is associated with layer B. The split point on the keyboard can be adjusted by clicking on the ellipsis icon located to the right of the ribbon. The split note can be chosen from the *Split Note* drop down menu. Alternatively, the *Learn* function can be activated and the desired split note played on the MIDI keyboard connected to *Ultra Analog VA-3*.

5.2.10 Layer Settings

Clicking on the gear icon located next to the *Split* button in the lower left corner of the *Play* panels opens the *Layer Settings* window. Command buttons allow one to quickly enable and disable external controller assignments used in conjunction with the **Modulator** modules of each layer. One can also turn *on* of *off* pitch bend and the sustain pedal (MIDI cc number 64) for each layer independently. Layer settings also include adjustment of the split note which is used when the split keyboard mode is used.

5.3 The Synth View

The **Synth** view is activated by clicking on the *Synth* button. This view allows one to go 'under the hood' and tweak the core synthesis modules of *Ultra Analog VA-3* and therefore to customize its tone and behavior.

The two oscillators of *Ultra Analog VA-3* appear on the left of this view. They are followed by the two filter-amplifier lines and their associated mixers. One can switch from one line to the other by clicking on the tabs to the right of the oscillators. This reveals the main parameters of the filter and amplifier modules as well as that of the **LFO** module and envelope generators used as modulation sources.

To the right of the interface one finds the detailed or expert view of each of the synthesis modules giving access to all their parameters. One can switch to any of these modules by clicking on the corresponding tab to the left of the modules.

5.3.1 The Oscillator Module

The **Oscillator** module is the main sound source of *Ultra Analog VA-3* and offers the features of the most reputed analog oscillators. It provides a fine control on the pitch, standard waveforms, a sub-oscillator, and hard synchronization. The **Oscillator** module is implemented using precise modeling algorithms rather than wave tables providing alias free sources with clean pulse width modulation and synchronization.

Waveform

The *Shape* drop down menu is used to choose the wave form generated by the oscillator. Wave forms include sine wave, saw tooth, rectangular and white noise as shown in Figure 20.

When the *Shape* is set to rectangular, one has additional control and can adjust the pulse width of the rectangular wave. This parameter is controlled using the *PW* knob. In its leftmost position the pulse is very narrow while in its rightmost position the wave is perfectly square and only the odd harmonics are heard. Note that the pulse width of the rectangular waveform can be modulated with the signal received from the **LFO** module. The amount of modulation, around the value set with the *PW* knob, can be set from the **LFO** module using the *VCO PW* knob.

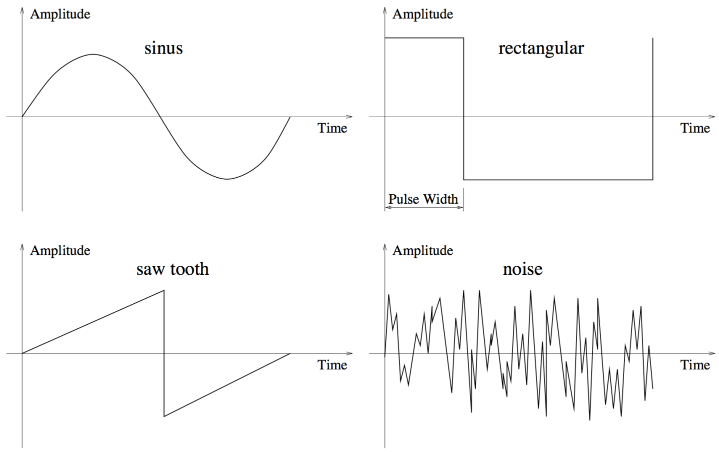


Figure 20: Choice of wave forms provided by the **Oscillator** module.

Pitch

The pitch of the **Oscillator** module can be adjusted using the *Octave*, *Semi* and *Detune* controls. The *Octave* control allows one to transpose the pitch by octaves (upper or lower) while the *Semi* control is used to transpose the pitch by semitones. To adjust these parameters, click on the corresponding button to select it and drag the mouse (or use the finger on a track pad) up or down to select a value. The *Detune* knob is used to slightly detune the oscillator. When the *Detune* knob is in its center position, there is no detune while turning it clockwise or anti-clockwise increases or decreases the pitch respectively.

The pitch of the **Oscillators** module can be modulated by the pitch signal received from the MIDI keyboard and/or the **LFO** module. The amount of modulation from these sources is adjusted by using the *Key* and the *VCO Detune* gain knob from the **LFO** module. In order for the oscillator to respond to the keyboard with an equal tempered scale, the *Key* knob of the **Oscillator** module must be set to its center position.

The **Oscillator** module includes a ramp generator which allows one to obtain a pitch variation when a key is depressed. The *Ramp* modulation knob is used to set the starting pitch of the oscillator while the *Decay* knob determines the time for the pitch to glide from this starting value to the effective pitch. The starting pitch can be set from -72 to +72 semi-tones with respect to the effective pitch by turning the *Ramp* knob clockwise. There is no pitch change when this control is in its middle position.

Sub Oscillator

The **Oscillator** module is equipped with a sub-oscillator which generates a wave one octave below the pitch of the oscillator. The shape of the sub-oscillator signal is a square wave when the wave shape of the **Oscillator** is set to *Rect* or *Saw* and a sine wave when it is set to *Sine*. The amount of this sub-oscillator signal in the output signal of the oscillator is adjusted with the *Sub* knob. When this knob is fully turned to the left, the sub-oscillator is muted. Note that when the oscillator is used in its **Sync** mode (see section 5.3.1 below), the sub-oscillator becomes unavailable.

Synchronization

The **Oscillator** module features a hard synchronization mode which is toggled *on* or *off* with the help of the *Hard Sync* button. Synchronization involves a master and a slave oscillator. In this mode, the signal from the slave **Oscillator** is reset, or in other words restarted, at the beginning of each period of the waveform of the master oscillator which therefore acts as a master clock as shown in Figure 21.

The perceived pitch of the final output of the oscillator module is the same as that of the master oscillator while the frequency of the slave or synced oscillator only affects the harmonic content and therefore the timbre of the resulting signal. The *Shape* control applies to the shape of the slave oscillator. The shape of the oscillation of the master is indeed irrelevant as it is only used to generate a master clock signal.

In hard sync mode, the frequency of the master oscillator is determined by the settings of the *Octave*, *Semi* and *Key* controls. The frequency of the slave oscillator is adjusted using the *Tune* knob and its value is presented as a ratio between the frequency of the slave and that of the master oscillator. It can take values between 1 and 20 and can be modulated by the ramp signal of the oscillator envelope using the *Ramp* knob and the output from the **LFO** module.

Note that in sync mode, the sub-oscillator knob is not available.

5.3.2 The Filter Module

Ultra Analog VA-3 is equipped with two multi-mode filters. The filters are patched in a flexible way in order to allow their use in parallel, in series or any combination of both. For even more flexibility, the cutoff frequency of the **Filter 2** can also be locked to that of **Filter 1**.

Each of the multi-mode filters include a resonant low-pass, band-pass, high-pass, notch and two formant filters which can be selected using the *Type* drop-down menu. The order of the filters can be adjusted to 2 (-12 dB/oct for low-pass and high-pass and -6 db/oct for band-pass) or 4 (-24 dB/oct for low-pass and high-pass and -12 db/oct for band-pass) with the help of the *Order* parameter. Furthermore different saturation algorithms, selected with the *Drive* drop-down menu, can be applied to the filters. The resonance or cutoff frequency of the filters is adjusted with the *Frequency* knob while the amount of resonance is controlled with the *Resonance* knob. When a formant filter is used, the *Resonance* knob is used to cycle between the vowels (a, e, i, o, u).

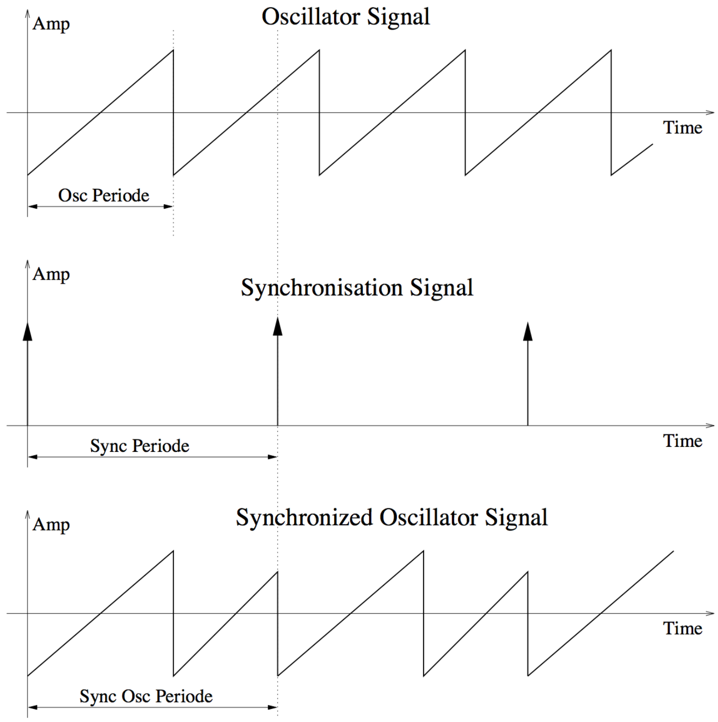


Figure 21: Synchronization of the oscillator.

The cutoff frequency and resonance of the filters can be modulated with different modulation sources. The *Key* knob allows one to adjust the amount of variation with the pitch signal from the keyboard while the *Env* knob is a gain control applied to the signal received from the **Filter Env** module. These parameters can also be modulated by the signal from the LFO using the *Filter Freq* and *Filter Res* knobs on the **LFO** module.

Let's now have a closer look at the different filter types available.

Resonant Low-Pass Filter

A low-pass filter is used to remove the higher spectral components of the signal while leaving the lower components unchanged. The frequency at which attenuation begins to take effect is called the *cutoff* frequency and it is controlled using the *Frequency* knob. In a resonant filter, frequencies located around the cutoff frequency can also be emphasized by an amount called the *quality factor* or *Q-factor* of the filter as illustrated by Figure 22. This parameter is adjusted using the *Resonance* knob. The higher the Q-factor, the louder and sharper the response of the filter around the cutoff frequency. When the Q-factor is set to 1 (*Q* knob fully turned to the left),

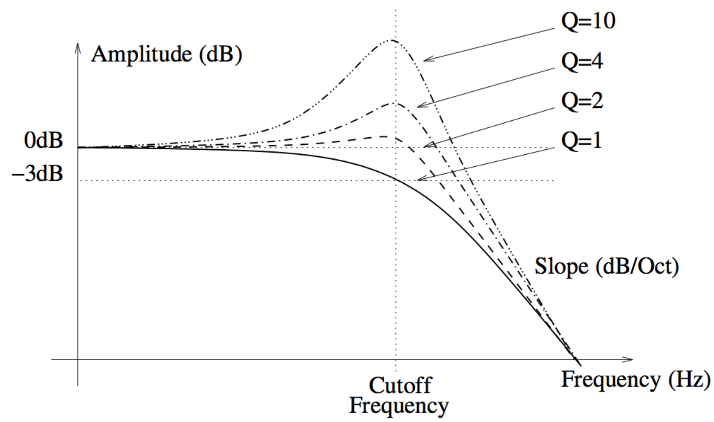


Figure 22: Frequency response of the low-pass filter.

there is no emphasis around the cutoff frequency and the attenuation is -3dB at the cutoff frequency. The attenuation for frequencies located above the cutoff frequency depends on the order of the filter which is set by the *Order* menu, a slope of -12dB/Oct corresponding to a second order filter and a slope of -24dB/Oct to a fourth order filter.

Resonant High-Pass Filter

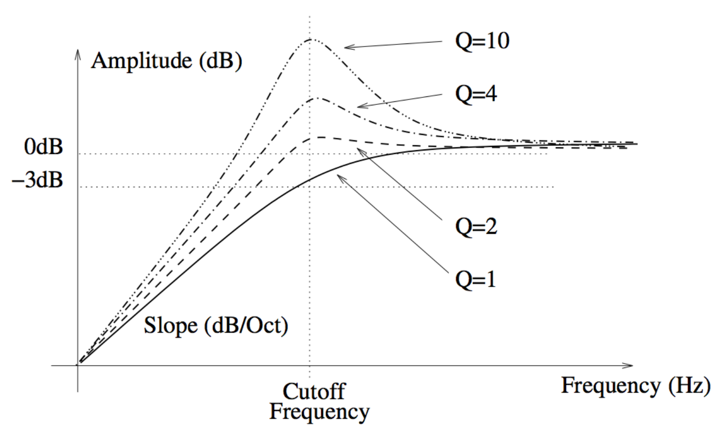


Figure 23: Frequency response of the high-pass filter.

The high-pass resonant filter works exactly in the opposite manner as the low-pass resonant filter by removing the frequency component of a signal located below the cutoff frequency while leaving those above the cutoff frequency unchanged. Similarly to the low-pass filter, the *Q-factor* controls the emphasis of frequencies located around the cut-off frequency.

Band-Pass Filter

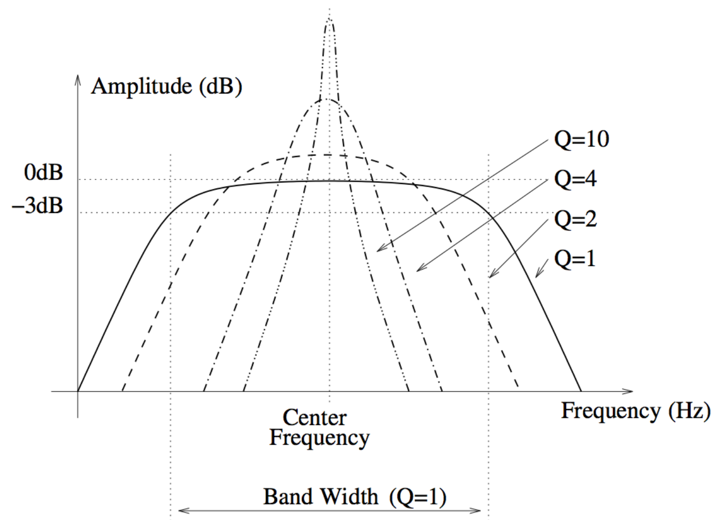


Figure 24: Frequency response of the band-pass filter.

The behavior of a band-pass filter is to let the frequencies in a band located around a center frequency and to attenuate the frequencies outside of this band as shown in Figure 24. The bandwidth of the band-pass filter is set with the *Resonance* knob while the center frequency is set with the *Frequency* knob. The *Order* control sets the order of the filter. This parameter affects the slope of the roll-off on both sides of the center frequency. For a second order filter the slope is -6dB/Oct while for a fourth order filter it is -12dB/Oct.

Notch Filter

The notch filter, does essentially the opposite of the band-pass filter. It attenuates the frequencies in a band located around the center frequency and leaves those outside of this band unchanged as shown in Figure 40. The *Frequency* knob is used adjust the center frequency and the *Resonance* knob sets the bandwidth of the notch. Note that the center frequency is totally removed from the spectrum of the output signal of the filter.

Formant Filter

The formant filter reproduces the filtering effect of the vocal tract in the human voice. By changing the position of the tongue, the opening of the mouth and opening or closing the nasal cavities one can change the filter applied to the glottal signal and thus produce the different vowels. Measurements have shown that this filter can be modeled by three peaking EQ filters corresponding to the three main cavities of the vocal tract as shown in Figure 26 and also known as formants. By moving

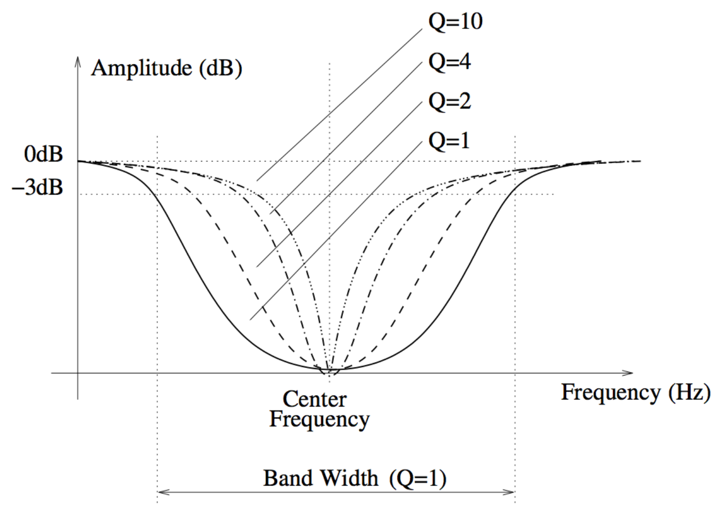


Figure 25: Frequency response of the notch filter.

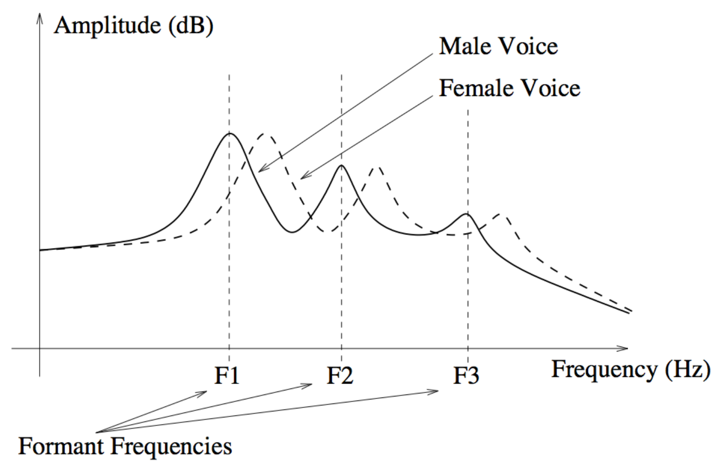


Figure 26: Frequency response of the formant filter.

the parameters of these three filters (frequency, amplitude and Q-factor) one can cycle between all the vowels. The effect of the *Frequency* knob on the formant filter is to offset all the formants by the same factor and it is used to switch between male voice (left position), female voice (center) and child (right position). The *Resonance* knob is used to cycle between vowels. Note that changing these parameters can be automated by using the different modulation signals.

Filter Drive

As seen in the preceding sections, some filters can boost the signal in some regions of the spectrum. Theoretically, with a high Q-factor the amplification can reach up to 50dB, but in real life, electronic components saturate before this level is reached and saturation is introduced in the signal. In other words, the output signal of a circuit can raise linearly up to a certain value but then begins to raise more slowly up to a value where the output is clipped to the maximum output value allowed by the circuit. This effect is typical of the sound signature of vintage circuits and is implemented in *Ultra Analog VA-3* through the *Drive* control as shown in Figure 27. There are 6 different saturation patterns implemented, the three first being symmetrical, meaning that the saturation is the same for positive and negative values of the signal while for the three remaining the distortion pattern applied is not the same for positive and negative values of the signal.

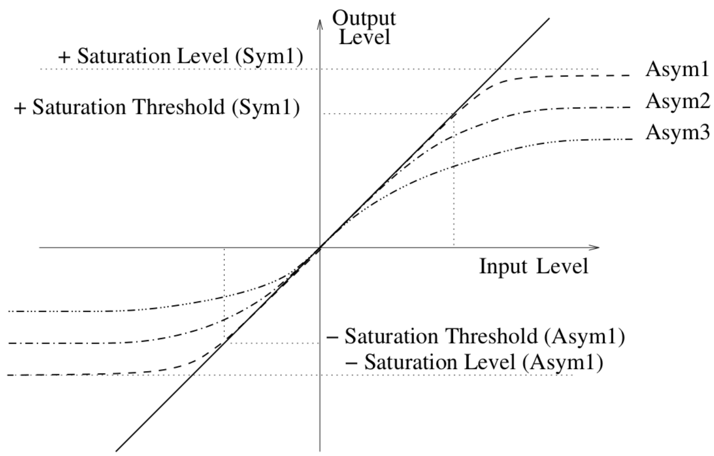


Figure 27: Saturation curves of the resonant filters. For asymmetrical curves, the saturation level and threshold are different for positive and negative values of the input level.

The distortion algorithms are selected with the *Drive* drop-down menu and are numbered from one to three for both symmetrical and asymmetrical saturation. These distortion algorithms range from very low distortion (clean circuit) to very high distortion.

Slave Filter 2

The **Filter 2** module can be enslaved to the **Filter 1** module. This can be done by clicking the *Slave* toggle switch on the **Filter 2** module. In this mode, the frequency of the **Filter 2** module follows exactly that of the **Filter 1** module. The *Frequency* knob on the **Filter2** module is then used to adjust the offset frequency between the **Filter 1** and **Filter 2** modules. When slaved, the cutoff frequency of **Filter 2** module follows exactly that of the **Filter 1** module including the effects of the modulation signals from the keyboard, LFO or envelope generator acting on the **Filter 1** module.

Note that the modulation entries on the **Filter 2** module are kept active so do not forget to disable them in order to have the desired effect.

Signal routing to the filters

The filter mixers allow one to control how the signal from the different sources, the two oscillators and the noise source, are routed to the filters.

The amount of signals received by the filters from the the **VCO1** and **VCO2** modules is adjusted by the *Vco1* and *Vco2* knobs. These control simply apply a gain factor to the signal received from the oscillators. The gain is increased by turning the knobs clockwise from a value of zero (-inf dB) to a factor of about 3 (10 dB). In the same way, the amount of signal from the noise source is adjusted using the *Noise* gain knob.

Finally, signal from the **Filter 1** module can be routed to the input of the **Filter 2** module. The amount of signal is controlled using the *Filter 1* knob appearing on the mixer of the second filter. In order to have the two filters strictly in series, this knob should be turned clockwise and the *VCA 1* switched *off* as the output of the first filter is hard wired into the input of the **VCA 1** module. In this way, the output signal only comes out from the second amplifier after going through the two filters.

5.3.3 The VCA Module

After filtering, the signal is routed to an amplifier in order to add an amplitude envelope and panning effect to the sound. Both the **Filter 1** and **Filter 2** modules have their own amplifier section, the **VCA 1** and **VCA 2** modules, which allows one to keep the signal coming from each filter totally independent from the other.

The two main controls of the amplifier module are the *Level* knob and the *Pan* knob. The *Level* knob is used to adjust the overall level of the amplifier while the *Pan* knob is used to position the sound in the stereo field. The source can be positioned from left to right by turning the *Pan* knob clockwise; it is centered when the knob is in its center position.

The level of the sound is always modulated by the signal from the **Amp Env** module which is hard-wired to the **Amp** module. The level of the sound across the overall range of the keyboard can also be adjusted using the *Key* modulation knob. In its center position, the sound level is the same across the keyboard. Turning the *Key* knob clockwise boosts high notes and decreases the amplitude of low notes while turning it to the left boosts low notes while decreasing the amplitude of high notes. Note that the middle C key (C3, MIDI note 60) always sounds a the same level regardless of the position of the *Key* knob. The output signal from the **LFO** module can be used for adding a tremolo effect to the sound. This is achieved by using the *VCA Gain* knob on the **LFO** module.

The pan value can be modulated by signals from the **LFO**, the **Amp Env** modules and the pitch of the note played. The modulation signals moves the source relative to the source position determined by the settings of the *Pan* knob. Negative values of the modulation signal move the source to the left of the source while positive values move it toward the right.

Using the **LFO** module as a modulation source moves the signal from left to right at the frequency of the signal from the **LFO** module. This is achieved by using the *VCA Pan* knob on the **LFO** module. Using the signal from the **Env** module as the modulation source moves the source to one side of the source and brings it back to its original position. Note that since the sign of the envelope signal is always the same, the source would need to be positioned completely to the left or to the right (*Pan* knob in its leftmost or rightmost position) in order for the source to sweep the entire space. Finally, the pitch signal from the keyboard positions notes depending on their pitch, low notes to the left and high notes to the right where the middle C is the reference note positioned at the location determined by the *Pan* knob.

5.3.4 The Noise Generator Module

The **Noise** module generates white noise and is followed by a -6dB/Oct low-pass filter used to adjust the frequency content of the noise. The *Color* knob is used to vary the cutoff frequency of the built-in low-pass filter. Turning this knob to the right increases this frequency and therefore increases the high frequency content of the noise until, in its rightmost position, the filter is fully opened. The *Noise* knob on the filter mixers is used to control the amount of noise sent to each filter.

5.3.5 The LFO Module

The LFO module is used as a modulation source for the **Oscillator**, **Filter** and **Amplifier** modules. On the **LFO** module, one can adjust the waveform, the phase of the signal and fade-in behavior.

Wave Shape

The waveform of the **LFO** is selected with the *Shape* drop-down menu. The possible values are *Sine* for sinus, *Tri* for triangular, *Rect* for rectangular and *Rnd* and *Rnd 2* for the two random modes. In the case of the triangular and rectangular waves, the *PW* (Pulse Width) knob is used to control the symmetry of the wave. This allows one to go from a pulse to a square wave when a rectangular waveform is selected, and from a saw-tooth to a perfectly triangular wave when the triangular waveform is selected. Note that the *PW* knob has no effect on the *Sine*, *Rnd* and *Rnd 2* waveforms. When the *Shape* control is set to *rnd*, the LFO outputs random values at the rate determined by the *Sync* control or the *Rate* knob. In this case, the output value from the LFO remains constant until a new random value is introduced. The *Rnd 2* mode reacts almost like the preceding mode except that the **LFO** module ramps up or down between successive random values instead of switching instantly to the new value.

Rate

There are two ways to adjust the rate, or frequency, of the output of the **LFO** module. If the *Sync* switch is in its *off* position, the rate is fixed with the *Rate* knob. When the *Sync* switch is *on*, the frequency of the oscillator is fixed relative to the frequency (tempo) of the host sequencer or the master clock (see 5.2.1) in standalone mode. Sync values are adjusted using the *Sync Rate* control and range from 1/8 of a quarter note (a thirty-second note) to 16 quarter notes (4 whole notes). The **LFO** module can also be synced to a triplet (t) or a dotted note (d). Note that when the *Sync* control is depressed, the *Rate* knob has no effect.

Phase and Reset Mode

The LFO module behaves in a polyphonic way which means that a low frequency oscillator is associated with each voice of the polyphony. This allows the LFO module to control notes played on the keyboard individually. The gate signal received from the keyboard is used to reset the **LFO** waveform when a note is played on the keyboard. The specific point in the waveform where the **LFO** module starts generating signal is determined by the position of the *Phase* knob and whether or not the **LFO** is in reset mode.

The reset mode is enabled by clicking on the *Reset* button. In this mode, the phase of the output signal of the LFO corresponding to a polyphonic voice is fixed and adjusted with the *Phase* knob. This means that every time a note is depressed on the keyboard and a gate signal is received from the **Keyboard** module, the **LFO** module starts generating signal for this specific voice starting at a specific point in the cycle of the waveform. The initial phase of the signal is determined with the *Phase* knob which enables one to select values over a full period of the waveform. The value of the phase lag is increased by turning the knob clockwise and is equal to from 0 to 100 % (0 to 360 degrees). This feature enables, for example, the triggering of filter sweeps or panning effects that always start at the same point every time a note is played.

When the **LFO** is not in *Reset* mode (Reset button off), the phase of the signal is random and determined within a range fixed by the *Phase* knob. Turning this knob clockwise increases the range to values located at different point of the wave period as explained above. The main interest of this mode is to keep voices uncorrelated when chords are played. Indeed, the LFO modulation signals corresponding to the different voices then start at different points even if all the notes are triggered at the same time. Note that when the **LFO** module is not in *Reset* mode and the *Phase* knob is fully turned to the left (phase of 0 degree), the synthesizer behaves as if the *LFO* was monophonic and all the voices played were following this single *LFO*.

Fade-In

One more feature of the **LFO** module is the possibility to add a fade-in effect to its output signal. The duration of this fade-in can be adjusted within the range of 0 to 5 seconds, as determined by the *Fade* knob. Turning this knob fully to the left results in a value of 0 which is equivalent to removing the fade-in effect. The time at which the LFO signal is introduced can even be controlled by adding a delay to the fade in. This parameter can also be set to values varying between 0 and 5 seconds, as determined by the *Delay* knob. Note that this knob is effective even if the *Fade* value is adjusted to zero. In this case, the signal from the **LFO** module si simply delayed.

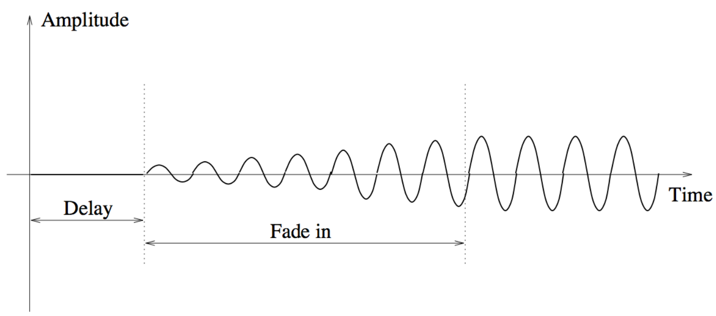


Figure 28: Fade in feature of the LFO.

Modulation Destinations

The signal from the **LFO** module can be used to modulate parameters in the **VCO**, **Filter**, and **VCA** modules using the controls located on the left part of the module. Available modulation destinations on the **VCO** module are the cutoff frequency and pulse width parameter controlled with the *VCO Detune* and *VCO PW* knobs respectively. The *Filter Freq* and *Filter Res* knobs are used to control modulation of the cutoff frequency and resonance of the **Filter** module. Finally the *VCA Gain* and *VCA Pan* knobs are used to control the modulation of the gain and panning of the **VCA** module.

5.3.6 The Envelope Module

Each row of *Ultra Analog VA-3* is equipped with two envelope generators, the **Filter Env** and **Amp Env** modules which are used to modulate the **Filter** and **Amp** modules. Both envelope generators have the same user interface and offer the same functions. Envelopes are generated through the use of a standard ADSR (attack, decay, sustain, release) approach including MIDI velocity modulation and looping capabilities.

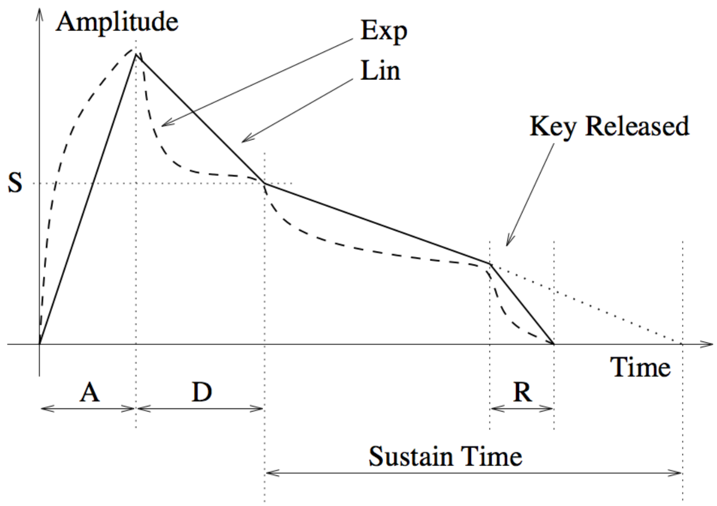


Figure 29: Response curve of an envelope generator. Broken line: exponential, full line: linear.

The envelope modules generate a four segment envelope: attack, decay, sustain, release. The attack time is adjusted using the *Attack* knob. The attack time can also be modulated with the velocity signal received from the MIDI keyboard to make the attack time shorter as the velocity increases, the intensity of this effect being controlled using the *Vel* knob on the left of the *Attack* knob. When this knob is in its leftmost position, the attack is only determined by the value of the *Attack* knob, turning the knob clockwise increases the influence of the velocity signal until the attack time is strictly determined by the inverse of the velocity signal when the *Vel* knob is fully turned to the right. The decay time is set with the *Decay* knob. The sustain phase of the envelope generator lasts from the end of the decay phase until the key is released. When the *Sustain* knob is fully turned to the left, the sustain level is zero and there is no sustain phase while fully turned to the right, the sustain level is at maximum and there is no decay phase. When the key is released, the envelope generator toggles to the release phase and the envelope signal decreases from the value at the end of the sustain phase to zero in a time set by the *Release* knob.

Note that during the sustain phase, the envelope signal can be made to decrease even if a key is still depressed. The time taken to go from the level set by the *Sustain* knob to a value of zero is then determined by the value of the *Time* knob located below the *Sustain* knob. When the *Time* knob is fully turned to the left, the envelope signal falls to zero right after the end of the decay phase; when the knob is in it rightmost position, the sustain level is held as long as the key is depressed. Finally, the overall level of the envelope can be controlled with the velocity signal from the keyboard through the *Vel* slider located to the left of the *Sustain* knob and behaves in exactly the same way as the *Vel* parameter associated with the *Attack* knob.

Shape of the Envelope Segments

We have so far determined the shape of the envelope by adjusting the duration of the different phases as well as their level. The envelope signal can further be modified by adjusting the shape of the envelope segments. These are linear when the *Exp* button is in its *off* position and become exponential when it is switched *on* as shown in Figure 29.

Free Run Mode

The *Free* button sets the envelope into free-run mode. This allows one to bypass the sustain phase of the envelope or in other words, to go directly from the decay phase to the release phase regardless of the amount of time a key is depressed.

Legato Mode

This mode is used to choose how the envelope generator reacts when a new note is played before the end of the preceding one. When this occurs, two strategies are possible. The first one consists in triggering a new envelope, from the attack phase, when the new note is played. The second consists in applying the current envelope signal of the first note to the second note which produces a legato effect. The first strategy is adopted when the *Retrig* button switched *on* while the second is applied when it is in its *off* position. Note that when the keyboard is in polyphonic mode and the envelope generator in retrig mode, the envelope generator behaves in a monophonic manner, reacting to a logical OR between all the gates signal from the different notes played on the keyboard. In other words the envelope generator is only triggered by the first note played on the keyboard and then remains in the sustain stage until the last note is released.

Loop Modes

The envelope generator features three loop modes: *AD*, *ADR* and *Once*. These modes allow the envelope to cycle between several stages of the envelope until the key is released and are selected from the *Loop* drop-down menu.

In the *AD* mode, the envelope begins with the attack and decay phases as usual, but rather than holding the sustain level, it repeats the attack (from the sustain level) and the decay phases until the note is released. When the note is released the envelope signal returns to zero following the release phase. The *ADR* mode is quite similar to the preceding mode and only differs in that the looping is done by cycling trough the attack, decay and release phases. Finally, in the *Once* mode, the envelope is played normally but the envelope generator cycles once more through the attack and release phase when the key is released.

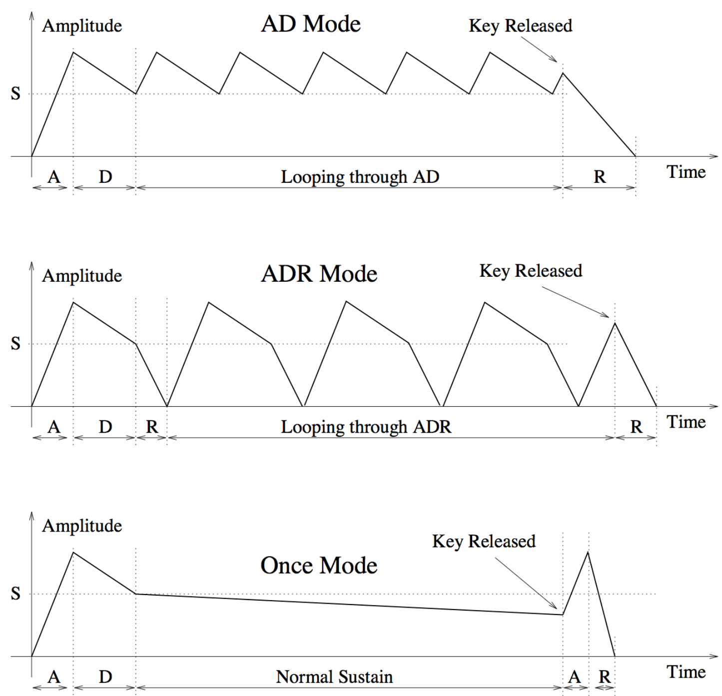


Figure 30: Loop mode of an envelope generator.

5.4 The Effects View

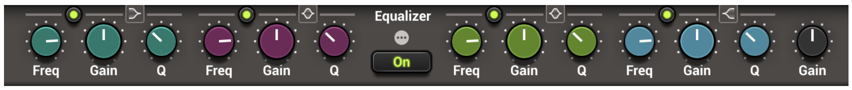
The **Effects** view is displayed by clicking on the *Effects* tabs in the layer mixer section and is based around a **Multi-effects** module. Note that there is a **Multi-Effects** module at the output of each layer and one at the output of the synthesizer after the layer mixer. The individual effects modules are identical in each of these **Multi-Effects** modules.

The **Multi-Effects** module allows one to process and shape the signal. This module comprises an **Equalizer** and a **Compressor** in series with two configurable effect processors and a **Reverb**. The effect processors can be set to a different type by using the drop-down menu located at the center of each module for a wide range of possibilities. They are turned *on* or *off* by using the *On* button located just below this menu. The effect list includes a **Delay**, **Distortion**, **Chorus**, **Flanger**, **Phaser**, **Wah Wah**, **Auto Wah**, **Guitar Amplifier**, **Tremolo**, and a **Notch** filter.

The **Multi-Effects** module is also visible from the **Play** view just below the utility section. This allows one to see rapidly which effects are selected for a given sound, turn the effects *on* or *off* and rapidly adjust the amount of each effect. The **Compressor**, **Equalizer** and **Reverb** can also be adjusted from this view.

5.4.1 Equalizer

The **Equalizer** module provides equalization over the low, mid, and high frequency bands. It is composed of a low shelf filter, two peak filters, and a high shelf filter in series, labelled **LF**, **LMF**, **HMF**, and **HF** respectively.



The functioning of the low shelf filter is depicted in Figure 31. The filter applies a gain factor to low frequency components located below a cutoff frequency while leaving those above unchanged. The cutoff frequency of this filter is adjusted using the *Freq* knob and can vary between 40 and 400 Hz. The *Gain* knob is used to adjust the gain factor applied to the signal in a $\pm$15dB range. In its center position there is no attenuation (0 dB). Turning it clockwise boosts the amplitude of low frequencies while turning it counter-clockwise reduces it.

The high frequency content of the signal is controlled with a high shelf filter that works in the opposite manner as the low shelf filter as illustrated in Figure 31. The filter applies a gain factor to components located above a cutoff frequency while leaving those below unchanged. The cutoff frequency of this filter, located above 1 kHz, is adjusted with the help of the *Freq* knob while the gain factor applied to the signal, in a $\pm$15dB range, is adjusted using *Gain* knob. In its center position there is no attenuation (0 dB). Turning it clockwise boosts the amplitude of high frequencies while turning it anti-clockwise reduces it.

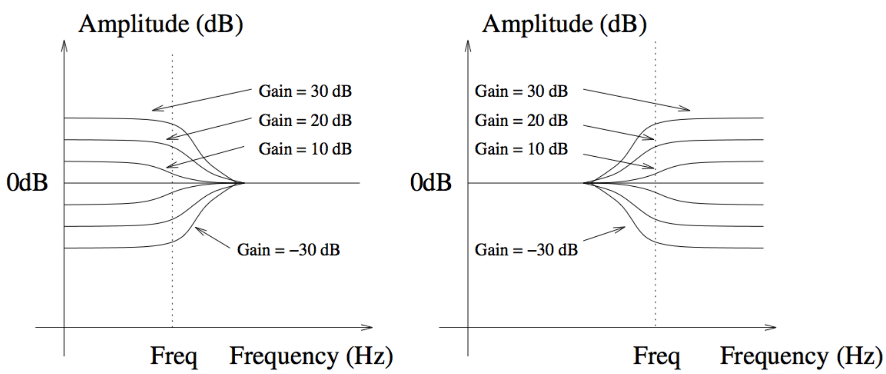


Figure 31: Low and high shelf filters.

The **Equalizer** module features two peak filters, labeled **LMF** and **HMF**, allowing to shape the signal in two frequency bands as illustrated in Figure 32. The filters apply a gain factor to frequency components in a band located around the cutoff frequency of the filters. This cutoff frequency is adjusted using the *Freq* knob and can vary between 100 Hz and 10 kHz. The gain factor applied a the cutoff frequency is controlled by the *Gain* knob and can vary in a $\pm$15 dB range. When in its center position there is no attenuation (0 dB). Turning it clockwise boosts the amplitude of frequencies located around the cutoff frequency while turning it anti-clockwise reduces it. The

Q knob is used to adjust the so-called quality factor of the filter which controls the width of the frequency band on which the filter is active. In its leftmost position, the frequency band is wide and it gets narrower as the knob is turned clockwise.

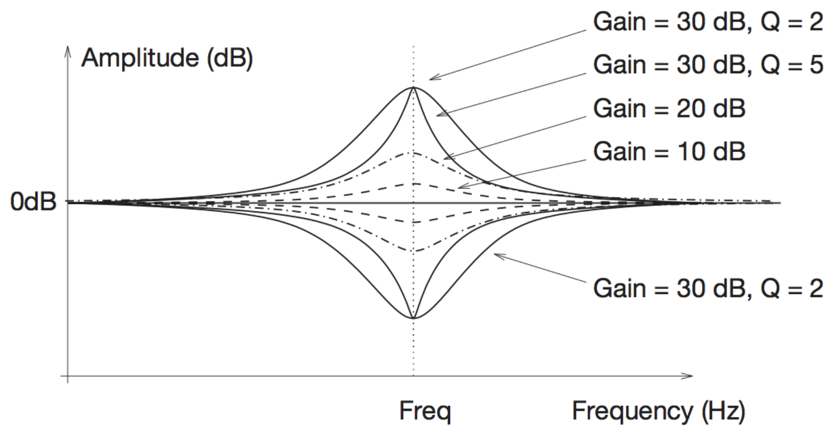


Figure 32: Peak filter.

The *Gain* knob is used to adjust the output level of this module. In its center position, the level is left unchanged, it is decreased by turning the knob counter-clockwise and increased by turning it clockwise.

5.4.2 Compressor

The **Compressor** module is used to automatically compress, in other words reduce, the dynamics of a signal. This module receives two input signals. The first one is the signal to be compressed while the second one is a control signal which triggers the compression process when it rises above a given level.

Tuning

The level at which the **Compressor** starts to enter into action is determined by the value of the *Threshold* parameter. This value is in dB and corresponds to the amplitude of the input signal as monitored by the first level meter of the module.

The amount of compression applied to the part of the signal exceeding the threshold value depends on the *Ratio* parameter which varies between value of 1:1 and 1:16. This parameter represents the ratio, in dB, between the portion of the output signal from the compressor above the threshold value and the portion of its input signal also exceeding the threshold value. As one might expect, increasing the ratio also increases the amount of compression applied to the signal. For example, a ratio of 1:5 means that if the input signal exceeds the threshold by 5 dB, the output signal will exceed the threshold by only 1 dB. Note that the *Ratio* parameter can also be adjusted from the **Play** view.

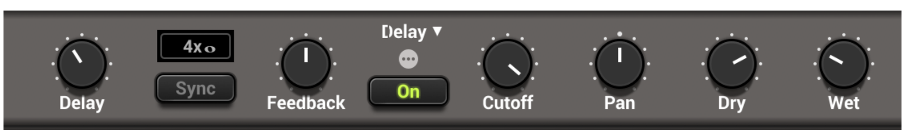
Two other controls affect the behavior of the **Compressor**. The *Attack* knob is used to set the time, in milliseconds, before the **Compressor** fully kicks in after the level of the input has exceeded the threshold value. A short value means that the compressor will reach the amount of compression as set by the *Ratio* knob rapidly. With a longer attack, this amount will be reached more gradually. In other words, the attack time is a measure of the attack transient time of the compression effect. The **Release** parameter is similar and represents the amount of time taken by the **Compressor** to stop compressing once the amplitude of the input signal falls below the threshold value.

The **Makeup** knob is used to adjust the overall level at the output of the **Compressor** module and is used to compensate from an overall change in signal level due to the compression effect.

The attenuation or gain reduction level meter, located in the middle of the module, indicates the amount of compression applied by the module. It is the difference between the input and output signals of the module before makeup gain is applied.

5.4.3 Delay

The **Delay** module consists in a stereo feedback loop with a variable delay in the feedback. It is used to produce an echo effect when the delay time is long (greater than 100 ms) or to color the sound when the delay time is short (smaller than 100 ms).



The *Delay knob* is used to adjust the amount of delay, in seconds, introduced by the effect. Turning this knob clockwise increases the delay. The *Feedback* parameter is a gain factor, varying in the range between 0 and 1, applied to the signal at the end of the delay lines. It controls the amount of signal that is re-injected in the feedback loop. In its leftmost position, the value of this parameter is 0 and no signal is re-introduced in the delay line which means that the signal is only delayed once. Turning the knob clockwise increases the amount of signal re-injected at the end of the feedback loop and therefore allows one to control the duration of the echo for a given delay time. In its rightmost position, the gain coefficient is equal to 1 which means that all the signal is re-injected into the feedback loop and that the echo will not stop. In addition to this gain factor, low pass filtering can also be applied to the signal re-injected into the feedback loop. The cutoff frequency of this filter is controlled using the *Cutoff* knob.

The *Pan* knob is used to balance the input signal between the left and right channels. In its leftmost position, signal will only be fed into the left delay line and one will hear clearly defined echo first from the left channel and then from the right channel and so on. In its rightmost position, the behavior will be similar but with the first echo coming from the right channel. These two extreme position correspond to the standard ping pong effect but a a less extreme behavior can be obtained by choosing an intermediate position. In particular when the *Pan* knob is in its center position, an equal amount of signal is sent in both channels.

The output signal from the **Delay** module can include a mix of input signal (dry) and delayed signal (wet). The *Wet* and *Dry* knobs are used to adjust the amplitude of each component in the final output. The amplitude of each component is increased by turning the corresponding knob clockwise from no signal to an amplitude of +6dB. Note that the *Wet* parameter is also adjustable from the **Play** view.

5.4.4 Distortion

The **Multi-Effect** module includes three different types of distortion which are selected using the *Shape* selector knob. The *Warm Tube* effect applies a smooth symmetrical wave shaping to the input signal resulting in the introduction of odd harmonics in the signal. The *Metal* distortion is similar to the *Warm Tube* effect but is slightly asymmetrical resulting in the introduction of even and odd harmonics in the signal. The *Solid State* distortion applies an aggressive symmetrical clipping to the signal thereby adding high frequency harmonics and resulting in a harsh sound.

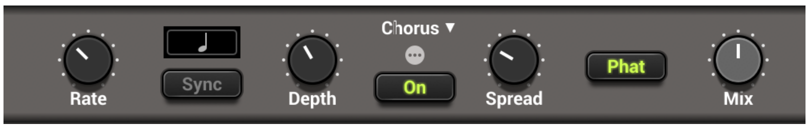
The *Drive* control is a gain knob acting on the input signal. This parameter allows one to adjust the amount of distortion introduced in the signal by controlling how rapidly the signal reaches the non-linear portion of the distortion curve applied on the signal. In its leftmost position, the amplitude of the input signal is reduced by -6 dB; turning this knob clockwise allows one to increase its amplitude. Note that the *Drive* parameter is also adjustable from the **Play** view.

The *Tone* knob is used to adjust the color of the signal after the distortion algorithm has been applied. In its leftmost position, high frequencies will be attenuated in the signal while in its rightmost position low frequencies will be filtered out from the signal. In its center position, the signal will be left unchanged.

The *Volume* knob is a gain knob acting on the amplitude of the distorted signal. Finally, the *Mix* knob allows one to control the amount of dry and wet (distorted) signal in the final output signal from the **Distortion** module. In its leftmost position, there is only dry signal in the output while in its rightmost position one only hears the distorted signal. In its center position, there is an equal amount of dry and wet signal in the output.

5.4.5 Chorus

The chorus effect is used to make a source sound like many similar sources played in unison. It simulates the slight variations in timing and pitch of different performers executing the same part. The effect is obtained by mixing the original signal with delayed version obtained from the output of delay lines as shown in Figure 33. In the case of a chorus effect, the length of the delay lines must be short in order for the delayed signals to blend with the original signal rather than be perceived as a distinct echo. The length of the delay line can be modulated introducing a slight perceived pitch shift between the voices.



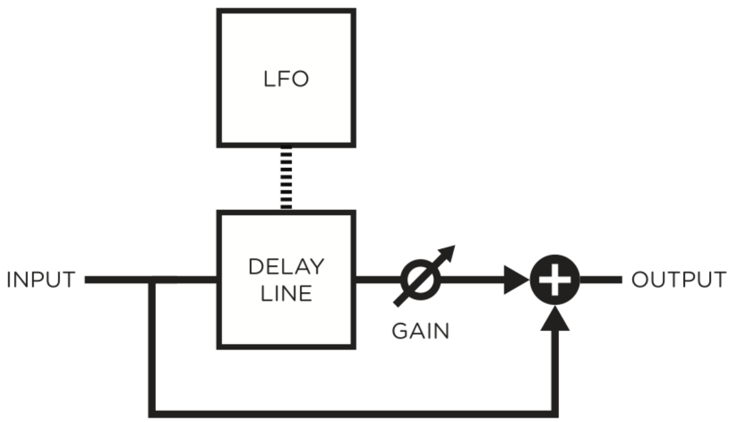


Figure 33: **Chorus** module.

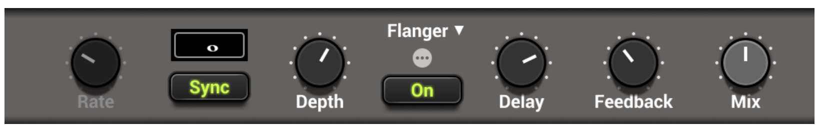
Tuning

The amount of modulation of the length of the delay lines is adjusted using the *Depth* knob. In the left position, there is no modulation and the length of the delay lines remains constant. As the knob is turned to the right, the length of the delay line starts to oscillate by an amount which increases as the knob is turned clockwise thereby increasing the amount by which the different voices are detuned. The frequency of the modulation is fixed with the *Rate* knob.

The *Fat* button is used to control the number of voices in the chorus effect. Switching this button *on* increases the number of voices. The *Spread* knob is used to adjust the amount of dispersion of the different voices in the stereo field. When in its leftmost position, there is an equal amount of left and right output signal on each channel. In other words the signal is the same on both channels. In its rightmost position, there is complete separation between the channels, the left output from the chorus is only sent to the left channel while the right output of the chorus is only sent to the right channel. Finally, the *Mix* knob allows one to mix the dry and wet signals. In its leftmost position, there is no output signal from the chorus and one only ears the dry input signals. In its rightmost position, one only ears the wet signal from the chorus module. In its center position, there is an equal amount of dry and wet signal in the output signal from the module.

5.4.6 Flanger

The **Flanger** module implements the effect known as *flanging* which colors the sound with a false pitch effect caused by the addition of a signal of varying delay to the original signal.



The algorithm implemented in this module is shown in Figure 34. The input signal is sent into a variable delay line. The output of this delay is then mixed with the dry signal and re-injected into the delay line with a feedback coefficient.

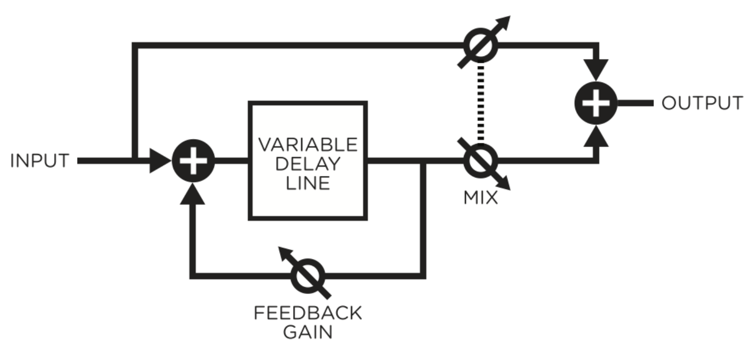


Figure 34: **Flanger** algorithm.

The effect of the **Flanger** module is to introduce rejection in the spectrum of the input signal at frequencies located at odd harmonic intervals of a fundamental frequency as shown in Figure 35. The location of the fundamental frequency $f0$ and the spacing between the valleys and peaks of the frequency response is determined by the length of the delay line ($f0 = 1/(2\text{delay})$), the longer the delay, the lower is $f0$ and the smaller the spacing between the harmonics while decreasing the delay increases $f0$ and hence the distance between the harmonics.

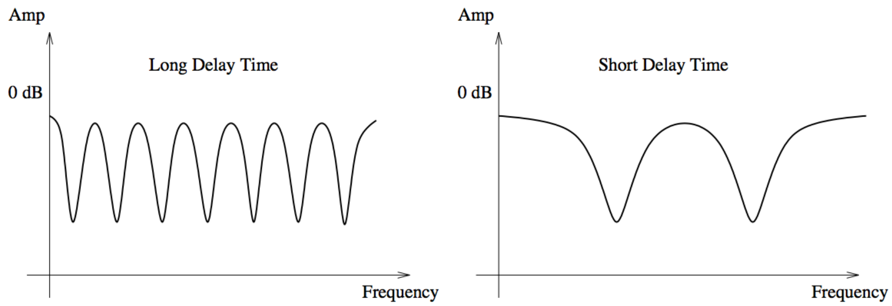


Figure 35: Frequency response of a **Flanger** module. Effect of the length of the delay line.

The amount of effect is determined by the ratio of wet and dry signal mixed together as shown in Figure 36. As the amount of wet signal sent to the output is increased, the amount of rejection increases. Finally, the shape of the frequency response of the **Flanger** module is also influenced by the amount of wet signal re-injected into the feedback loop as shown in Figure 37. Increasing the feedback enhances frequency components least affected by the delay line and located at even harmonic intervals of the fundamental frequency. As the feedback is increased, these peaks become sharper resulting in an apparent change in the pitch of the signal.

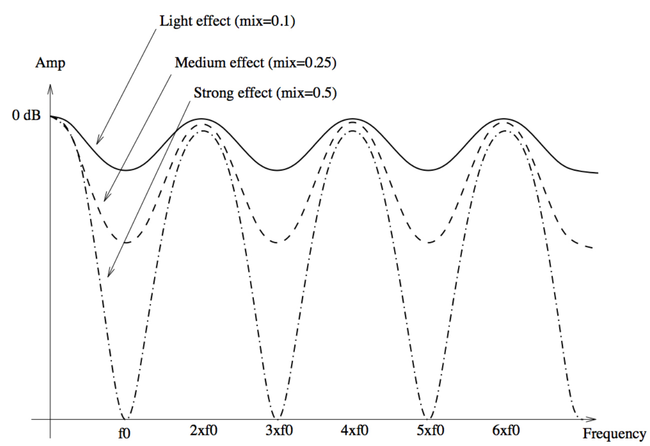


Figure 36: Effect of the mix between wet and dry signal on the frequency response of a **Flanger** module

Tuning

The delay length, in milliseconds, is adjusted with the *Delay* knob. The length of this delay can be modulated by a certain amount depending on the adjustment of the *Depth* knob. In the left position,

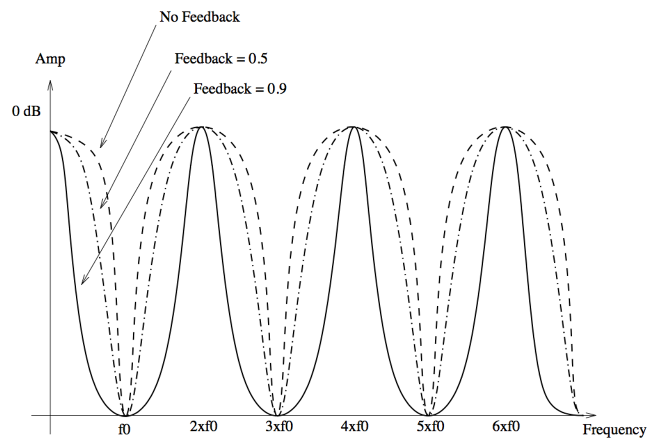
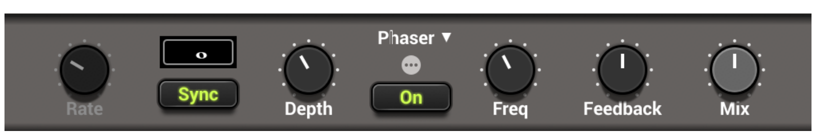


Figure 37: Effect of the amount of feedback on the frequency response of a **Flanger** module.

there is no modulation and the length of the delay line remains constant. As the knob is turned to the right, the length of the delay line starts to oscillate by an amount which increases as the knob is turned clockwise and at a frequency fixed with the *Rate* knob. The *Feedback* knob is a gain knob used to fix the ratio of wet signal re-injected into the delay. Finally, the *Mix* knob determines the amount of dry and wet signal in the output signal from the module. When this knob is adjusted in its leftmost position, only dry signal is sent to the output, in its center position, there is an equal amount of dry and wet signal in the output signal while in its rightmost position, only wet signal is sent to the output. Note that the *Depth* parameter is also adjustable from the **Play** view.

5.4.7 Phaser

The **Phaser** module implements the effect known as *phasing* which colors a signal by removing frequency bands from its spectrum. The effect is obtained by changing the phase of the frequency components of a signal using an all-pass filter and adding this new signal to the original one.



The algorithm implemented in this module is shown in Figure 38. The input signal is sent into a variable all-pass filter. This wet signal is then mixed down with the original dry signal. A feedback line allows the resulting signal to be re-injected into the filter. The effect of the **Phaser** module is to introduce rejection in the spectrum of the input signal depending on the tuning of the filter.

The all-pass filter modifies a signal by delaying its frequency components with a delay which increases with the frequency. This phase variations will introduce a certain amount of cancellation when this wet signal is mixed down with the original dry signal as shown in Figure 39. The rejection is maximum when the phase delay is equal to 180 degrees and a given component is out

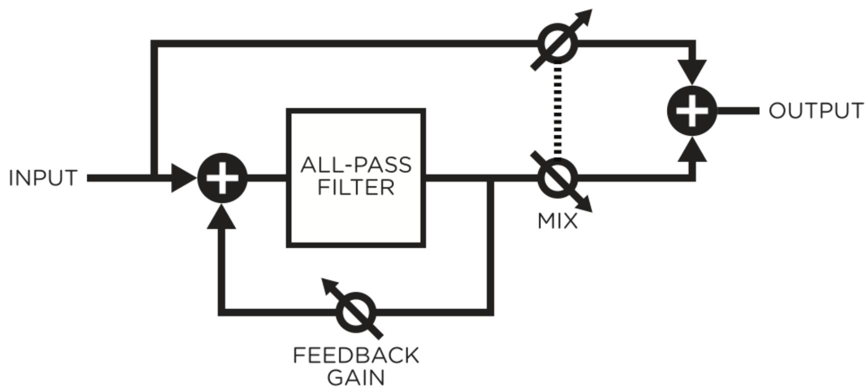


Figure 38: **Phaser** algorithm.

of phase with that of the original signal. The amount of effect is determined by the ratio of wet and dry signal mixed together as shown in Figure 39. As the amount of wet signal sent to the output is reduced, the amount of rejection increases. The shape of the frequency of the Phaser module is also influenced by the amount of wet signal re-injected into the feedback loop. Increasing the feedback enhances frequency components least affected by the all-pass filter. As the feedback is increased, these peaks become sharper. The functioning of the **Phaser** is very similar to that of the **Flanger** module. The filtering effect is different however, since the **Phaser** module only introduces rejection around a limited number of frequencies which, in addition, are not in an harmonic relationship.

Tuning

The location of the first notch in the frequency response of the module is adjusted with the *Frequency* knob This frequency can be modulated by an amount controlled with the *Depth* knob. In its leftmost position, the location of the first notch is fixed but it starts to oscillate by an amount which increases as the *Depth* knob is turned clockwise. The frequency of the modulation is controlled using the the *Rate* knob. The *feedback* knob is used to fix the amount of wet signal re-injected into the delay. Finally, the *Mix* knob determines the amount of dry and wet signal sent to the output. When this knob is adjusted in the left position, only dry signal is sent to the output, in its center position, there is an equal amount of dry and wet signal in the output and in the right position, only wet signal is sent to the output.

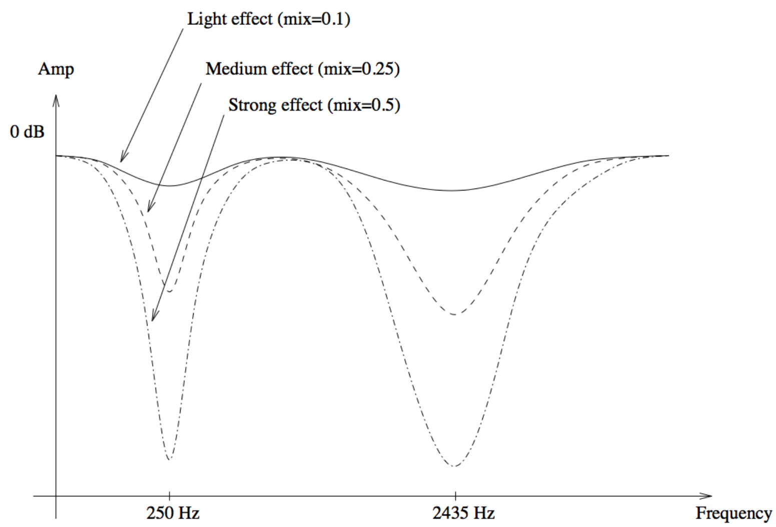
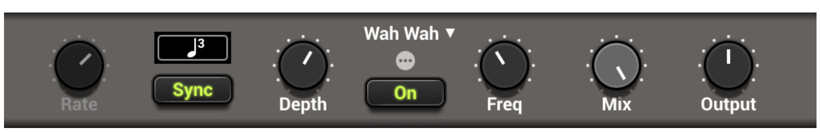


Figure 39: Frequency response of a **Phaser** module. Effect of the mix between wet and dry signal on the frequency response.

5.4.8 Wah Wah

The Multi-Effect module includes 2 different types of *Wah* effects: wah wah, and auto wah. These effects are used to enhance a frequency band around a varying center frequency using a bandpass filter. In the wah wah effect, the center frequency of the bandpass filter varies at a rate fixed by the user. In the case of the auto-wah, the variations of the center frequency is controlled by the amplitude envelope of the incoming signal.

The *Freq* knob is used to control the central frequency of the filter. Turning this knob clockwise increases the center frequency. In the case of the *Wah Wah* effect, the center frequency will oscillate around the value fixed by the *Freq* knob while with the *Auto Wah* effect, the setting of the *Freq* will fix the starting point value of the varying center frequency.



The *Depth* knob controls the excursion of the center frequency of the filter. In the case of the *Wah Wah* effect, this excursion is applied around the value fixed by the *Freq* knob while in *Auto Wah* effect the value of the center frequency increases from the value fixed by the *Freq* knob.

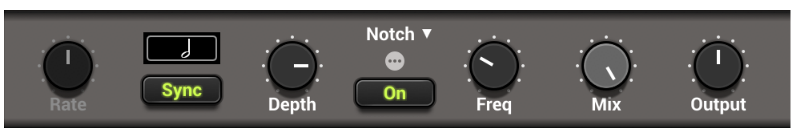
Turning this knob clockwise increases the excursion of the center frequency. Note that the *Depth* parameter is also adjustable from the **Play** view.

The *Rate* knob controls the frequency or rate of the modulation of the center frequency of the filter. In the case of the *Wah Wah* effect, turning this knob clockwise increases the rate of the modulation. In the case of the *Auto Wah* filter, this knob is labeled *Speed* and controls the time constant of the envelope follower. Turning this knob clockwise decreases the time constant, or in other words the reaction time, of the envelope follower.

The *Mix* knob allows one to mix the dry and wet signals. In its leftmost position, there is no output signal from the chorus and one only ears the dry input signals. In its rightmost position, one only ears the wet signal from the chorus module. In its center position, there is an equal amount of dry and wet signal in the output signal from the module. Finally, the *Output* know is used to adjust the output level of this module.

5.4.9 Notch Filter

The *Notch Filter* does essentially the opposite of a band-pass filter. It attenuates the frequencies in a band located around the center frequency and leaves those outside of this band unchanged as shown in Figure 40. As was the case for the *Wah Wah* effect, the filter can be modulated.



The *Freq* knobs is used to control the central frequency of the filter. Turning this knob clockwise increases the center frequency. The *Depth* knob controls the excursion of the center frequency of the filter around its center frequency. Turning this knob clockwise increases the excursion of the center frequency. Finally, the *Rate* knob controls the frequency or rate of the modulation of the center frequency of the filter. Turning this knob clockwise increases the rate of the modulation. Note that the *Depth* parameter is also adjustable from the **Play** view.

The *Mix* knob allows one to mix the dry and wet signals. In its leftmost position, there is no output signal from the chorus and one only ears the dry input signals. In its rightmost position, one only ears the wet signal from the chorus module. In its center position, there is an equal amount of dry and wet signal in the output signal from the module. Finally, the *Output* know is used to adjust the output level of this module.

5.4.10 Guitar Amplifier

The **Guitar Amplifier** module is a versatile 2-channel amplifier with speaker cabinet and spring reverb. With relatively few parameters, this amplifier module allows one to obtain a rich variety of sounds for different music styles.

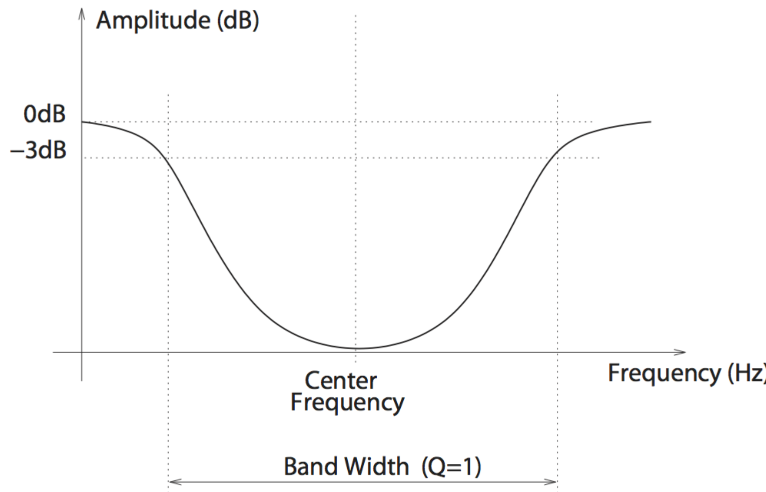


Figure 40: Frequency response of a notch filter.

The amplifier section of this module is switched *on* or *off* by clicking on the LED located in the top right corner of the section labelled *Amp* on the left of the module. The *Channel* LED allows one to switch between the two channels of the amplifier. Channel one offers clean to semi-dirty sound while channel two is well-suited when strong distortion is required. The *Drive* knob is used to adjust the amount of distortion in the sound. The sound becomes more and more distorted as the knob is turned clockwise. The *Mid* knob is used to set the amount of mid-range frequencies in the sound. In its middle position, the sound is not modified, mids are cut or boosted by up to ± 12 dB by turning this knob to the left or right. The *Level* knob is a gain knob which is used to adjust the overall volume of the amplifier. Note that the effect of this control on the frequency response of the amplifier is different for each channel.

The *Low* and *High* parameters are used to boost or cut low and high frequencies respectively by up to pm 18dB by turning the knob from its center position. These controls have a similar behavior for both channels. Additional control on the frequency response is obtained by using the *Bite* control which is switched *on* or *off* by clicking on the *Bite* LED just before to the channel selector. This parameter boosts high frequencies while cutting some low frequencies for a brighter sound.

The low-cut (or high-pass) filter is used to remove from the output sound of the instrument frequency components below the cut-off frequency. The cut-off frequency of the filter is increased by turning the knob clockwise. when this knob is in its leftmost position, the filter has no effect on the sound.

The spring reverb is turned *on* or *off* by using the *spring* LED in the top right of the section labelled *Spring*. The *Mix* knob is used to set the amount of wet signal in the mix, turning the knob clockwise increasing the amount of reverberation in the signal.

Finally, the speaker cabinet is switched *on* or *off* by clicking on the LED in the upper right corner of the cabinet section of the module. This part of the module simulates the effect of both the speaker and the cabinet on the frequency response of the amplifier module. The back of the cabinet can be open or closed using the *Type* selector. Opening the back of the cabinet allows waves to travel from the back of the cabinet and interfere with those traveling from the front part of the cabinet resulting in a more colored sound.

5.4.11 Tremolo

The **Tremolo** module is used to modulate the amplitude of the sound. The *Rate* knob is used to control the speed (frequency) of the modulation while the *Depth* knob controls its amplitude. The waveform knob is used to change the shape of the waveform used to modulate the sound. In its leftmost position, the waveform is a triangular and as the knob is turned clockwise it changes to a smoothed square wave. The *stereo* button is used to switch between stereo and mono mode. When the button is in its *on* position, the module is in stereo mode and the output signal from bounce with a 180 degrees phase difference between the left and right channels. In mono mode (button switched *off*) the signals in the left and right channels are the same.

5.4.12 Reverb

The **Reverb** module is used to recreate the effect of reflections of sound on the walls of a room or hall. These reflections add space to the sound and make it warmer, deeper, as well as more realistic since we always listen to instruments in a room and thus with a room effect. This module is located at the very end of the effects chain in the signal flow.

Impulse Response of a Room

The best way to evaluate the response of a room is to clap hands and to listen to the resulting sound. Figure 41 shows the amplitude of the impulse response of a room versus time. The first part of the response is the clap itself, the direct sound, while the remaining of the response is the effect of the room which can itself be divided in two parts. Following the direct sound, one can observe a certain amount of echoes which gradually become closer and closer until they can not be distinguished anymore and can be assimilated to an exponentially decaying signal. The first part of the room response is called the early reflexion while the second is called the late reverberation. The total duration of the room response is called the reverberation time (RT).

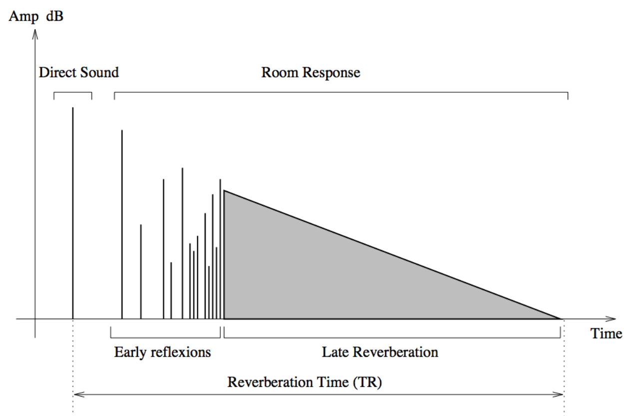


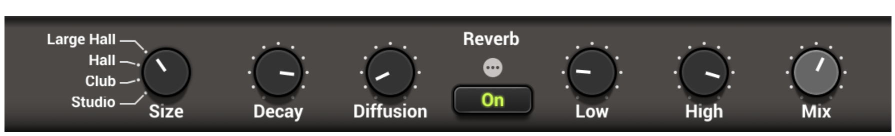
Figure 41: Impulse response of a room.

Adjusting the room effect

The size of a room strongly affects the reverberation effect. The *Size* selector is used to choose between the *Studio*, *Club*, *Hall* and *Large Hall* settings each reproducing spaces of different volumes from smaller to larger.

The duration of the reverberation time depends on both the size of the room and the absorption of the walls, which is controlled with the *Decay* knob. In a real room the reverberation time is not constant over the whole frequency range. As the walls are often more absorbent in the very low and in the high frequencies the reverberation time is shorter for these frequencies. These parameters are adjusted with the *Low* and *High* knobs respectively.

Another parameter which affects the response of a room is its geometry; the more complex the geometry of a room, the more reflexion are observed per unit of time. This quantity is known as the time density and can be set trough the *Diffusion* knob. In a concert hall, the time density is supposed to be quite high in order not to hear separate echoes which are characteristic of poor sounding rooms. The last parameter which affects our listening experience in a room, is the distance



between the sound source and the listener. While the room response is quite constant regardless of the position of the source and the listener, the direct sound (the sound which comes directly from the source) depends strongly on the position of the listener. The farther we are from the sound source the quieter is the direct sound relatively to the room response. The ratio between the direct sound and the room response is adjusted with the *Mix* knob which in other words is used to adjust the perceived distance between the source and the listener. In its leftmost position, only the direct sound is heard while when fully turned to the right, one only hears the room response. Note that the *Mix* parameter is also adjustable from the **Play** view.

Studio Mix

The *Studio Mix* knob on the left of the module is used to add a subtle reverb effect to the sound. The reverb preset used for this effect can not be modified and is completely independent from the other reverb effect adjusted with the other parameters of the **Reverb** module. This knob reverb is always active, even when the **Reverb** module is turned *off*. The *Studio Mix* knob is a mix knob which allows one to adjust the amount of wet and dry components in the signal. In its leftmost position, there is only dry signal in the output while in its rightmost position one only hears the input signal processed through this reverb preset. In its center position, there is an equal amount of dry and wet signal in the output.

5.4.13 Output Gain

In order to ensure a proper gain staging, the output level of the *Effects* section, in other words the post-effects signal level, should be between 0 and +4 dBr when playing a musical phrase mezzo forte (moderately loud), assuming of course that the pre-effects level is also correctly adjusted. It should be possible to achieve this using the different gains of the different effects modules. An extra gain parameter is provided in case this proves to be difficult to achieve. This gain is controlled using the *Output* knob located on the right of the **Reverb** module.

A coloured LED located just above the *Output* knob gives an indication of the level at this point in the signal flow. This LED is turns to light green when the signal is in the 0 to +4 dBr zone. It will turn to yellow and then red as the output level increases. For a more precise reading, a level meter is displayed when clicking on this LED. It is hidden by clicking again on this LED. For more details on general levels and level meters, please refer to section 7.4.1.

5.4.14 Unity Gain

Ideally, the effects modules should be adjusted to provide unity gain. In other words, their input and output levels should be equal or said differently turning them *on* or *off* should not change the output volume of the effect chain. This should be possible to achieve with the gain parameters provided with each modules. In some effect modules, an extra gain parameter is provided for more flexibility and appears in the lower right corner of the module. This gain is adjusted by clicking on the gain value (in dB) and entering a new value manually.

6 The Settings View

6.1 Settings

Clicking on the *Settings* button opens the settings window, shown in Figure 42 which is where some general parameters of the synthesizer, such as tuning, number of polyphony voices, pitch bend range, and external macro modules assignments, are fixed. The value of these parameter is not saved in a sound preset and therefore apply to all sounds. In other words, these parameters do not vary when new sounds are loaded. In standalone mode, the last saved configuration is always reloaded. In plug-in mode, these parameters are saved with a project which mean they can vary from one project to the other but are fixed within one project.

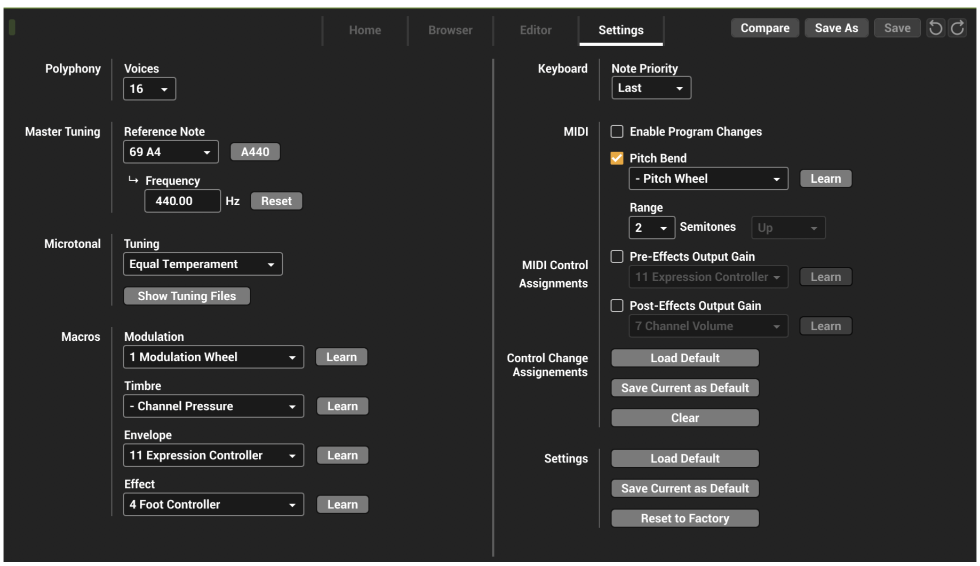


Figure 42: The *Settings* window.

6.1.1 Polyphony

The *Voices* control located at the top of the *Settings* window allows one to adjust the number of polyphony voices used by *Ultra Analog VA-3*. The number of voices is adjusted by clicking on the control and selecting the desired number of voices. In general, a higher number of voices is desirable but keep in mind that the CPU load is proportional to the number of voices used.

6.1.2 Master Tuning

Musical instruments are usually tuned based on a fixed reference, such as A440, and a temperament. The reference is a note whose frequency is fixed, for example 440 Hz for the A above the middle

C of the keyboard in the case of A440. A tuning fork or electronic tuner is typically used to give a reference note. A temperament is a tuning system, or a set of rules, which establishes how the octave is subdivided and allows one to calculate the frequency of all the other notes in a scale.

By default, *Ultra Analog VA-3* is tuned using equal temperament and a frequency of 440 Hz for the A4 note (MIDI note 69). These values are displayed under *Reference Note* and *Frequency* in the *Master Tuning* section. One can transpose, or in other words raise or lower the frequency of all the notes on the keyboard, by changing the frequency (in Hertz) of the reference note with the *Frequency* parameter. Any note can be chosen as a reference by clicking on the *Reference Note* drop-down menu. When choosing a new note, the value of the *Frequency* parameter is updated to the current frequency of this new note in A440 and equal temperament.

The A440 button to the right of the *Reference Note* control is used to revert to A440 when both the reference note and its frequency were changed. The *Reset* button next to the *Frequency* control brings back the frequency of the reference note to its original value.

Note that the tuning parameters in the *Settings* window are used to fix the general tuning of the synthesizer. The tuning of each layer, with respect to this reference tuning, can be adjusted independently using the *Tune* controls in the layer mixer as will be discussed below.

6.1.3 Microtonal Tuning

An interesting feature of *Ultra Analog VA-3* is that it can be tuned according to different temperaments using Scala micro-tuning files. Temperament files are loaded by clicking on the *Tuning* drop-down menu which shows a list of available temperament. By default, the only temperament available is the equal temperament. Other temperaments can be added by clicking on the *Show Tuning Files* button just below the *Tuning* parameter and copying Scala files in this folder.

Selecting a new Scala file with the *Tuning* automatically triggers the loading of the corresponding temperament. The reference note used as the base note for the scale described in the Scala file as well as its frequency are set using the *Reference Note* and *Frequency* parameters of the *Master tuning* section as explained above.

6.1.4 Macros

The *Macros* settings is used to assign an external MIDI continuous controller to the four**Macros** modules used in each layer. One can select a specific controler from the list of controller numbers appearing when clicking on the drop-down menus of this section for each of the **Macros** module. The **Learn** command can also be used to assign a controller. When one of the *Learn* buttons is switched *on*, the corresponding modulator will be assigned to the first continuous controller from which a message is received. Note that when the *None* option is chosen, the corresponding *Modulator* will not respond to any controller.

For more information on the **Macros** modules, please refer to section 5.2.5.

6.1.5 Keyboard

The *Note Priority* setting sets the behavior of the keyboard when several notes are depressed at the same time when the *Keyboard* module is set to monophonic mode or when the maximum number of polyphonic voices has been reached in polyphonic mode. In monophonic mode, the Priority determines which of the lower, last, or higher note has precedence when several notes are played. In polyphonic mode, this control determines which of the lowest, highest, or oldest note is muted in order to replace it with the newest note played once the maximum of polyphonic voices has been reached. Note that since this parameter determines the note priority, the stolen note will be the opposite of what appears in the control display.

6.1.6 MIDI

MIDI Program Changes

Ultra Analog VA-3 responds to MIDI program changes when the *Enable Program Changes* option of the *MIDI Program* setting is turned *on*. When this is the case *Ultra Analog VA-3* will load, when it receives a MIDI program change message, the sound having the same index number, in the currently selected sound pack, as the program number in the message. In order to view and modify the index of sounds in a pack, please refer to section 4.3.2.

Pitch Bend

Ultra Analog VA-3 reacts to the MIDI pitch bend signal when this option is activated. By default, the pitch bend is controlled by the pitch bend wheel but this can be changed by using the drop down menu appearing in this section. A **Learn** command is available to assign a controller automatically.

In order to adjust the range of the pitch bend, use the *Pitch Bend Range* drop-down menu. The different options are listed in number of semi-tones. Note that by choosing a value of zero semi-tone, *Ultra Analog VA-3* will stop responding to MIDI pitch bend signal. The direction of the bend can also be adjusted using the drop-down menu next to the range menu.

For more information on the channel volume and expression controller, please refer to section 8.2.6.

6.1.7 MIDI Control Assignments

By default the MIDI channel volume and expression controller (MIDI CC number 7 and 11 respectively) are mapped to gain parameters controlling the output volume of the synthesizer. In order activate the expression controller, check the *Pre-Effects Output Gain* option. In order to activate the channel volume, check the *Pre-Effects Output Gain* option. Note that the pre and post-effects output gains can be assigned to other MIDI continuous controllers by using the corresponding drop-down menus. The **Learn** next command can also be used to map these gain parameters to a controller. When this command is selected, the corresponding gain parameter is assigned to the first MIDI continuous controller from which a message is received.

For more information on the channel volume and expression controller, please refer to section 8.2.2.

6.1.8 Control Change Assignments

Any control on the *Ultra Analog VA-3* interface can be manipulated by an external MIDI controller through MIDI control change assignments or MIDI links. In order to save the current configuration of assignments as the default map, use the **Save Current as Default**. This default map will be loaded the next time the program is started in standalone mode. In order to revert to the default map, after making some modifications for example, use the **Load Default** command. The *Clear* command is used to disable all MIDI control assignments.

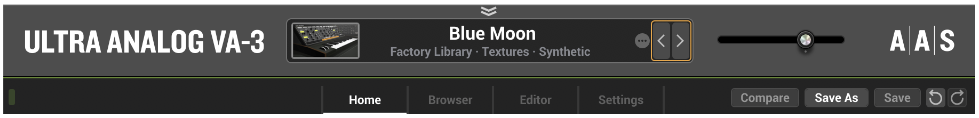
For more information on MIDI Control Assignments, please refer to section 8.2.3.

6.1.9 Saving Settings

Once changes are made to any parameter in the settings window, they are applied to the synthesizer regardless of the sound played. In plug-in mode, the value of these settings are automatically saved with the DAW project. In order to save the settings, one needs to use the *Save Current as Default* button, under *Settings*. The settings values last saved will be used as the default values when a new instance of the program is started in plugin mode. The *Load Default* button is used to load this default map which may be useful when making changes to the settings and wanting to revert to the original configuration. It is also possible to revert to factory settings by clicking on the *Reset to Factory* button. Note that in standalone mode, the settings values used when starting the program are the same as those when the program was last closed.

7 Utility Section and Layer Mixer

The utility section is located at the top of the *Ultra Analog VA-3* interface and it includes important parameters and monitoring tools.



7.1 Sound Display

At the top of the utility section one finds the sound display with information on the currently loaded sound and its associated sound pack. Information includes the name of the sound and sound pack, the sound category, and sound quality.

Clicking on the ellipsis icon next on the right of the sound name opens a menu of commads which can be applied to the sound. The **Compare** command allows one to compare the current version of the sound with the original version saved on disk. This command is only active when the sound has been modified. When in **Compare** mode, edition is blocked and it is therefore not possible to modify any parameter. The **Compare** mode must then be switched *off* in order to resume edition. Note that the *Compare* button in the lower right part of the utility section is a shortcut for this command. The **Save** and **Save As** commands are used to save a new version of a sound. Note that the **Save** command is only available when the sound is modified and is always inactive for read-only AAS factory sounds and expansion packs. These two commands also have shortcut buttons next to the *Compare* button. The **Reload** command is used to reload the original version of a modified sound. This may be useful when a sound has been modified but one wants to discard the changes. The **Back** and **Forward** commands are used to go through the history of modifications applied to a sound backwards or forwards in time. The two rounded arrow buttons next to the *Save* button, are shortcuts for these commands. The **Locate** command allows one to locate the current sound in the sound library. Different options are available in order to locate the sound (by pack, creator, category, or general list of sounds). The browser is automatically opened at the corresponding location once an option has been chosen. Finally, the **Show Sound Details** command is used to open the *Sound Details* window which includes general information about the sound including its name, category, quality, creator, and a general note field. This information can be edited and saved in this pop-up window.

It is possible to change the currently loaded sound by clicking on the name of the current sound which opens a drop-down menu with the list of sounds included in the currently loaded sound pack. It is also possible to navigate through this list using the left and right-pointing arrows to the right of the sound name. Note that when these arrows are surrounded by an orange line, it is possible to change the sound using the arrows on the computer keyboard. This feature is de-activated by hitting the *Escape* key on the computer keyboard or clicking outside the sound display on the interface.

For more information on sounds, sound packs, and the sound manager, please refer to Chapter 4.

7.2 The MIDI LED

The MIDI LED is located on the left of the utility section just below the product name. The LED blinks when the synthesizer receives MIDI signal. If the application is not receiving MIDI signal, make sure that the host sequencer is sending MIDI to *Ultra Analog VA-3*. If you are running in standalone mode, make sure that the MIDI controller you wish to use is well connected to your computer and that it is selected as explained in Chapter 8.

7.3 View Selector

The utility section includes, just below the sound display, tabs labelled *Home*, *Browser*, *Editor*, and *Settings* which are used to switch between the main views of *Ultra Analog VA-3*.

For a general description of these views, please refer to chapter 2.

7.4 The Layer Mixer

The layer mixer, shown in Figure 43, includes general controls which can be applied to each layer. It is also where the output level of each layer and the master output is monitored and adjusted using the different level meters and *Gain* sliders.

Each layer can be named using a label. In order to edit a label, click on it and type on the computer keyboard. Once a name has been entered, hit the *Return* key or click outside the label in order to deselect this region.

A layer can be switched *on* or *off* by clicking on the power switch icon located just before its label. Switching *off* a layer not only mutes its output but completely deactivates the synthesis engine of the layer. Note that in this case, the split layer feature of the keyboard is also deactivated. Each layer can also be muted or soloed by using the *M* and *S* buttons respectively. The level of a layer is adjusted using the corresponding *Gain* slider. The volume is adjusted by click-holding the mouse on the cursor and moving it. A specific level can also be reached at once by clicking directly on the slider rail. Note that it is possible to move the cursor of both layer by the same amount. In order to move the sliders together shift-click on a slider and move it. Finally the *Master* level slider is also duplicated on the utility section so that it is always available even when the layer mixer is hidden.

The *Pan* knob is used to position the output of a layer in stereo space by adjusting the relative amplitude of signals sent to the left and right channels. When in its leftmost position, signal is only sent to the left channel while in its rightmost position signal is only sent to the right channel. When in its center position, an equal amount of signal is sent to both channels.

Each layer can also be transposed independently using the *Tune* controls. The adjustments are relative to the general tuning of the synthesizer which is specified in the *Settings* window as explained above. This control is composed of two numbers separated by a dot. The first number indicates a value in semi-tones while the second one indicates a value in cents (one hundredth of a semi- tone). The amount of transposition can be adjusted by click-dragging upward or downward on the semi-tone and cent controls. Double clicking on these controls brings back their value to zero.

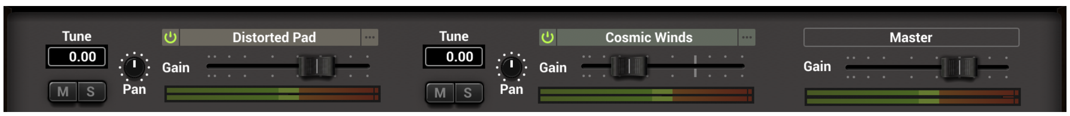


Figure 43: The layer mixer.

Clicking on the ellipsis icon to the right of each layer label opens a menu with a list of commands which can be applied to a layer.

The **Copy** and **Paste** commands are used to copy the settings of an entire layer and apply it to the other layer of the same sound or any other sound.

The **Load** command is used to open the layer preset window which displays a list of user saved layer presets. These can be layers which are used often in the creation of new sounds. By default, this list includes a list of AAS factory basic layers which can be used as init presets or a basis for new layers. The **Save** command is used to add new layers to this list of user saved presets. Note that layer presets can be renamed and deleted in the layer preset window.

The **Browse** command is also used to load layer parameters into a layer slot but from layers used in other sounds rather than from the list of user saved layer presets. This command opens the layer browser allowing one to select layers from sounds in the different available sound packs. For more information on the layer browser, please refer to section 4.6.

The **Swap** command is used swap the content of Layer A and B of a sound which can be useful for testing different mix and panning settings.

Finally, the **Layer Settings** command is used to activate or deactivate the effect of pitch bend and sustain pedal on a given layer as well as that of the different macro controls. By default, all these control signals are active for a given layer. When one is filtered-out, the box with the ellipsis icon to the right of the layer label becomes yellow.

7.4.1 Level Meter

The level meters allow one to monitor peak and RMS (root means square) level of the left (L) and right (R) output channels from the synthesizer. As a limiter is located at the output of *Ultra Analog VA-3*, it is important to make sure that the amplitude of the signal remains within values that ensure that no distortion is introduced in the signal at the output.

The optimal level for the signal lies in the light green zone of the level-meter (0 and +4 dBr) and should typically be reached when playing at mezzo forte (moderately loud) level. The 0 dB mark on the level meter has been adjusted to correspond to -20 dBFS (full scale). This means that at that level, the signal is -20 dB below the maximum allowed value. This ensures a headroom of 20 dB which should be more than enough to cover the dynamics of most playing situations and therefore guarantee that no additional distortion is added in the output signal.

A peak value mark allows one to follow the maximum level values reached by the output signal. The limiter is triggered when this mark enters the red zone of the level meter (17 dB) and it remains active while the side vertical bars at the top of the lever meter are switched *on*.

7.5 The About Box

The **About** box is open by clicking on the chevrons located at the very top of the interface. The box is closed by clicking again on the chevrons or outside the box. Useful information is displayed in this box such as the version number of the program, and the serial number and the email address associated with the program license. The box also includes a link to the pdf version of this manual.

8 Audio and MIDI Settings

This chapter explains how to select and configure Audio and MIDI devices used by *Ultra Analog VA-3*. Audio and MIDI configuration tools are accessed by clicking on the *Audio Setup* button located in the lower left corner of the *Ultra Analog VA-3* interface and the *MIDI* button located just below the MIDI led in upper part of the interface.

Note that in plug-in mode the audio and MIDI inputs, sampling rate, and buffer size are set by the host sequencer.

8.1 Audio Configuration

8.1.1 Selecting an Audio Device

Audio configuration tools are available by clicking on the *Audio Setup* button located in the lower left corner of the *Ultra Analog VA-3* interface. The **Audio Setup** dialog first allows you to select an audio output device from those available on your computer. Multi-channel interfaces will have their outputs listed as stereo pairs.

On Windows, the audio output list is organized by driver type. The device type is first selected from the **Audio Device Type** drop-down list. If you have ASIO drivers available, these should be selected for optimum performance. The *Configure Audio Device* button allows you to open the manufacturer's setup program for your audio interface when available.

Once the audio input has been selected, you can then select a sampling rate and a buffer size from those offered by your audio interface.

8.1.2 Latency

The latency is the time delay between the moment you send a control signal to your computer (for example when you hit a key on your MIDI keyboard) and the moment when you hear the effect. Roughly, the latency will be equal to the duration of the buffers used by the application and the sound card to play audio and MIDI. To calculate the total time required to play a buffer, just divide the number of samples per buffer by the sampling frequency. For example, 256 samples played at 48 kHz represent a time of 5.3 ms. Doubling the number of samples and keeping the sampling frequency constant will double this time while changing the sampling frequency to 96 kHz and keeping the buffer size constant will reduce the latency to 2.7 ms.

It is of course desirable to have as little latency as possible. *Ultra Analog VA-3* however requires a certain amount of time to be able to calculate sound samples in a continuous manner. This time depends on the power of the computer used, the preset played, the sampling rate, and the number of voices of polyphony used. Note that it will literally take twice as much CPU power to process audio at a sampling rate of 96 kHz as it would to process the same data at 48 kHz, simply because it is necessary to calculate twice as many samples in the same amount of time.

Depending on your machine you should choose, for a given sampling frequency, the smallest buffer size that allows you to keep real-time for a reasonable number of voices of polyphony.

8.2 MIDI Configuration

8.2.1 Selecting a MIDI Device

The list of available MIDI inputs appears at the bottom of the **Audio Setup** dialog. Click on the *Audio Setup* button located in the lower left corner of the *Ultra Analog VA-3* interface and then click on the checkbox corresponding to any of the inputs you wish to use.

8.2.2 MIDI Channel Volume and Expression Controller

The MIDI channel volume and expression controller (MIDI CC number 7 and 11 respectively) messages received by *Ultra Analog VA-3* can be used to control gain parameters, and therefore the output volume, just before the multi-effects processor of each layer and after the global multi-effects processor. These controllers are enabled in the *MIDI Control* section of the *Settings* window which is opened by clicking on the *Settings* button located in the right of the utility section at the top of the interface.

The expression controller is set by default to control the gain parameter located before the multi-effects processor of each layer. This could be used, for example, to control the output volume of the synthesizer, but without loosing the tail signal from a reverb. In order to enable this controller, click on the *Pre-Effects Output Gain* option.

The channel volume controller is set by default to control the gain parameter located after the global multi-effects processor. In order to enable this controller, click on the *Post-Effects Output Gain* option.

Both the Pre and Post-effects gains can be assigned to other controllers. In order to change the assignment, use the drop-down menu corresponding to these gains to choose another MIDI continuous controller. Alternatively one can click on one of the *Learn* buttons which will link the corresponding gain parameter to the controller sending the next MIDI CC message received by *Ultra Analog VA-3*.

8.2.3 Creating MIDI Control Assignments

Every control on the *Ultra Analog VA-3* interface can be manipulated by an external MIDI controller through MIDI control change assignments or MIDI links.

In order to create a MIDI control assignment:

- On the *Ultra Analog VA-3* interface, right-click/Control-click on a control (knob, button) and select the **Learn MIDI Assignment** command.

- Move a knob or slider on your MIDI controller (this can be a keyboard, a knob box, or any device that sends MIDI). This will link the control of the *Ultra Analog VA-3* to the MIDI controller you just moved.

To deactivate a MIDI assignment, simply right-click/Control-click on the corresponding control on the *Ultra Analog VA-3* interface and select the **Forget MIDI Assignment** command.

Note that MIDI assignments are not saved with sound presets. They are global parameters which apply to all sounds. When *Ultra Analog VA-3* is used in plug-in mode, these assignments are saved with the DAW project and are therefore loaded when the project is opened. It is therefore possible to use different assignments in different projects.

8.2.4 Creating a default MIDI Assigment Map

In order to save the current configuration of assignments as the default map, open the *Settings* window by clicking on the *Setting* button located in the top right of the interface and use the **Save Current as Default** command in the *MIDI Control Assigment* section. This default map will be loaded the next time the program is started as a plug-in. In order to revert to the default map, after making some modifications for example, use the **Load Default** command in the *Settings* window. The *Clear* command is used to disable all MIDI control assignments at once.

8.2.5 MIDI Program Change

Ultra Analog VA-3 responds to MIDI program change messages. In order to enable MIDI program changes, open the the *Settings* window by clicking on the *Settings* button in the top right of the interface and check the **Enable Program Changes** option in the *MIDI Program* section. When this is the case, *Ultra Analog VA-3* loads the sound in the currently selected sound pack whose index number corresponds to the one received in the program change message. If you do not wish *Ultra Analog VA-3* to respond to MIDI program changes, deselect the **Enable Program Changes** option.

8.2.6 Pitch bend

Ultra Analog VA-3 reacts to the MIDI pitch bend signal received by *Ultra Analog VA-3*. The pitch bend wheel on the interface, located on the left of the *Home* view, can also be used to modify the pitch of a sound. The range of the pitch bend is 2 semi-tones up or down by default but can be changed. To adjust the range of the pitch bend, open the the *Settings* window by clicking on the *Settings* button in the top right of the interface and use the *Pitch Bend Range* drop-down menu. The different options are listed in number of in semi-tones. Note that by choosing a value of zero semi-tone, *Ultra Analog VA-3* will stop responding to MIDI pitch bend signal. The pitch bend signal can be assigned to another MIDI controller than the pitch bend wheel. This is adjusted in the *MIDI* section of the *Settings* view.

Pitch bend can quickly be disabled or enabled for each layer from the *Layer Settings* window which is opened by clicking on the ellipsis icon next to a layer label and choosing the *Layer Settings* command.

8.2.7 Modulation Wheel

Ultra Analog VA-3 responds to the signal from the modulation wheel of MIDI keyboards (continuous controller number 1). The first macro module is usually mapped to this controller. For more information on the **Macros** modules, please refer to section 5.2.5.

8.2.8 Sustain Pedal

By default, *Ultra Analog VA-3* responds to MIDI sustain pedal messages (MIDI cc number 64). The sustain pedal can however be turned *on* or *off* independently for each layer from the *Layer Settings* window which is opened by clicking on the ellipsis icon next to a layer label and choosing the *Layer Settings* command.

9 Using *Ultra Analog VA-3* as a Plug-In

Ultra Analog VA-3 is available in VST2, VST3, AAX and Audio Units formats and integrates seamlessly into the industry's most popular multi-track recording and sequencing environments as a virtual instrument plug-in. *Ultra Analog VA-3* works as any other plug-in in these environments so we recommend that you refer to your sequencer documentation in case you have problems running it as a plug-in. We review here some general points to keep in mind when using a plug-in version of *Ultra Analog VA-3*.

9.1 Audio and MIDI Configuration

When *Ultra Analog VA-3* is used as a plug-in, the audio and MIDI ports, sampling rate, buffer size, and audio format are determined by the host sequencer.

9.2 Automation

Ultra Analog VA-3 supports automation functions of host sequencers. All parameters visible on the interface below the *Utility* section and related to the synthesis engine can be automatized, in other words parameters from the *Layer Mixer*, *Home*, and *Editor* views.

9.3 Multiple Instances

Multiple instances of *Ultra Analog VA-3* can be launched simultaneously in a host sequencer.

9.4 MIDI Program Change

MIDI program changes are supported in *Ultra Analog VA-3*. When a MIDI program change is received by the application, the current sound used by the synthesis engine is changed to that having the same index, in the currently loaded pack, as that of the MIDI program change message.

In order to use MIDI program changes, make sure the *Enable Program Changes* option is selected in the Settings window as explained in section 6.1.

9.5 Saving Projects

When saving a project in a host sequencer, the currently loaded sound is saved with the project in order to make sure that the instrument will be in the same state as when you saved the project when you re-open it. Note that packs are not saved with the project which implies that if you are using MIDI program changes in your project, you must make sure that the pack you are using in your project still exists on your disk when you reload the project. The sounds must also exist and be in the same order as when the project was saved.

9.6 Performance

Using a plug-in in a host sequencer requires CPU processing for both applications. The load on the CPU is even higher when multiple instances of a plug-in or numerous different plug-ins are used. To decrease CPU usage, remember that you can use the **freeze** or **bounce to track** functions of the host sequencer in order to render to audio the part played by a plug-in instead of recalculating it every time it is played.

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