CHROMAPHONE 3

USER MANUAL

Applied Acoustics Systems

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Contents

1 Introduction

Chromaphone is a synthesizer based on acoustic resonators. Combinations of resonators are used to create drums, percussion, string and hybrid synth-like instruments. Membranes, bars, marimbas, plates, strings, and tubes form pairs that are excited by a mallet and a flexible noise source. Access to different parameters such as the material of the resonators, their tuning and hit position allow for the creation of a vast range of realistic and creative instruments and sonic colors.

Chromaphone is entirely based on Applied Acoustics Systems (AAS) physical modeling technology and uses no sampling nor wave tables. Sound is produced by solving, on the fly, mathematical equations modeling the different types of resonators and how they interact. This elaborate synthesis engine responds dynamically to the control signals it receives while you play reproducing the richness and responsiveness of real instruments. *Chromaphone* features an innovative coupling technology allowing an accurate description of the exchange of energy between the resonators resulting in rich and natural sounding tones.

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 *Chromaphone 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 *Chromaphone 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 *Chromaphone 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 the authorization procedure 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 *Chromaphone 3* in Standalone Mode

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

- **Windows** - Select *Chromaphone 3* from the **Start** menu.
- **Mac OS** - Double-click on the *Chromaphone 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 **Audio MIDI Setup** button located in the *Settings* view which is accessed by clicking on the *Settings* tab at the top of the 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

Chromaphone 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 *Chromaphone 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 sounds 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* tab 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 *Chromaphone 3* as a Plug-in

Chromaphone 3 integrates seamlessly into the industry's most popular multi-track recording and sequencing environments as a virtual instrument plug-in. *Chromaphone 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 *Chromaphone 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 *Chromaphone 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 *Chromaphone 3*

Chromaphone is synthesizer built around the combination of acoustic resonators. The resulting instruments are played using a mallet or the signal from a noise source. It is very simple yet the range of sounds it is capable of is surprisingly wide, from realistic reproductions of acoustic percussion instruments to creative and innovative tones simply not possible with traditional synthesizers.

2.1 Signal Flow of the *Chromaphone* Engine

Available resonator types are: string, open and closed tube, plate, drumhead, membrane, bar, marimba bar and a manual mode. Resonators can be configured to be in parallel or coupled mode as shown in Figures 1 and 2.

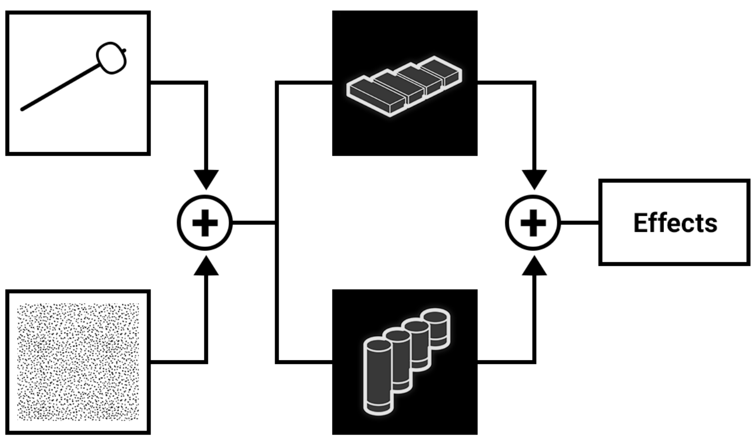


Figure 1: Signal flow of *Chromaphone*. Resonators in parallel mode.

In parallel mode, both resonators are excited by the sources and the output signal from the resonators is a simple mix between the output of both resonators, the balance between the sources being determined by the position of the *Balance* slider. In coupled mode, resonator A is excited and energy is transmitted to the second resonator at their junction point. At first sight this configuration could appear like a simple series configuration in which the signal from Resonator A is sent to Resonator B but *Chromaphone* really takes into account the bidirectional nature of the energy flow that occurs in real life when two objects are coupled. In other words, once energy is received by Resonator B, it starts to vibrate which in return influences the motion of Resonator A. The modeling of these complex interactions results in tones and timbres that reproduce the richness of sounds from real acoustic instruments. The amount of coupling between the two resonators is controlled with the help of the *Balance* slider.

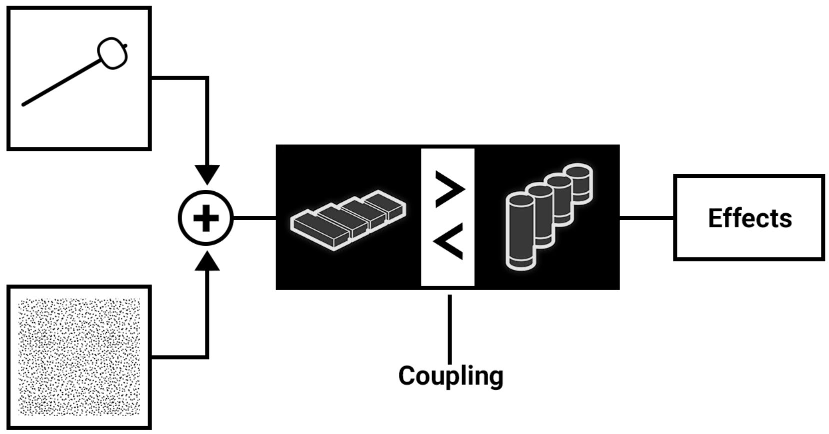


Figure 2: Signal flow of *Chromaphone*. Resonators in coupling mode.

2.2 Multimbral Architecture

Chromaphone 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 3 and consists of a MIDI routing module, two independent *Chromaphone* engines in parallel, a mixer, and a multi effects module.

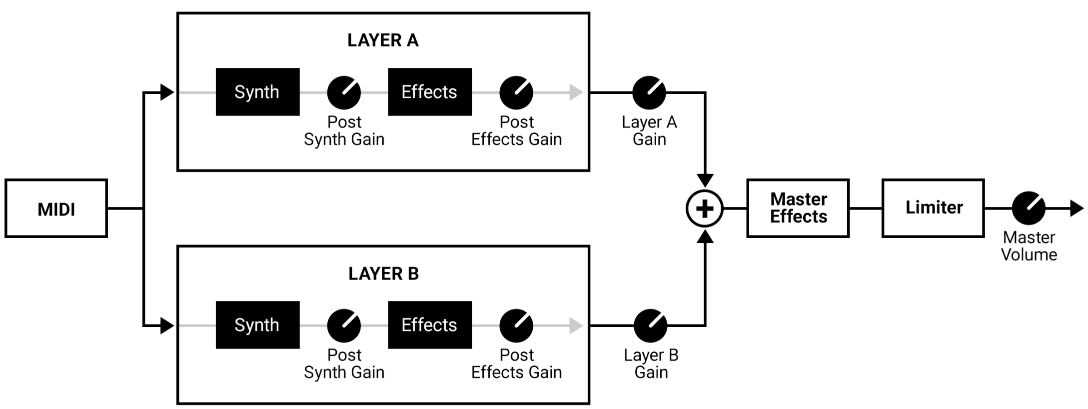


Figure 3: Multitimbral architecture of *Chromaphone 3*.

2.3 Interface

The top part of the interface is called the *Utility* section and is shown in Figure 4. 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 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 includes a MIDI LED located on the left of this section. This LED is is activated when the synthesizer receives MIDI signal. On the right of this section, one finds the master volume knob and a VU meter allowing one to monitor the level of the output signal from the synthesizer. Just below are 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 tabs 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.

The size of the interface can be adjusted by click-dragging the lower right corner of the interface or by choosing specific size ratios in the *Settings* view. This is useful to find the optimal size of the interface depending on your display and its resolution.

In the lower part of each view is a clickable seven octave virtual keyboard allowing one to trigger sounds directly from the interface which is useful when no MIDI keyboard is connected to the computer. For sounds using the split-keyboard feature, the range of the different layers is indicated by a colored line just above the keys of the keyboard.

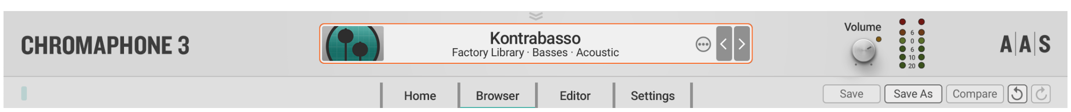


Figure 4: The *Utility* section.

2.3.1 The Home View

This is the main view for auditioning sounds and playing. The principal elements on this view 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

2.3.2 The Browser View

Clicking on the *Browser* tab reveals the sound browser. This view gives complete information on the sound library and this is where management tasks on sounds and sound packs can be carried

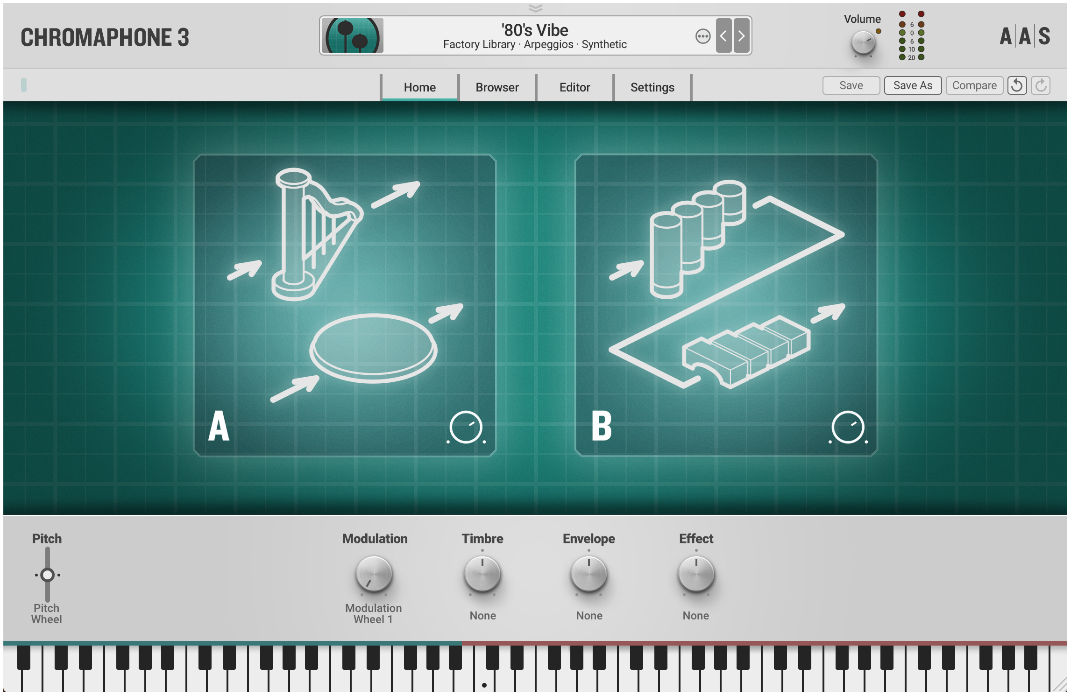


Figure 5: The *Home* view.

out. A complete description of the sound browser is the subject of Chapter 4.

2.3.3 The Editor View

Clicking on the *Editor* tab gives access to the synthesis engine itself. The multitimbral architecture of *Chromaphone 3* includes two identical instances of the synthesis engine in parallel, each represented by a different color and corresponding to one layer or sound. The graphical user interface of each synth engine is identical and has been organized around three different sections as shown in Figures 9, 10 and 11. The *Editor* view also includes a layer mixer which is used to adjust the contribution of each layer in the resulting sound.

The first section, called *Modes*, gives access to different performance parameters as well as to a step sequencer. The second and third sections, called *Synth* and *Effects* respectively, are used for in-depth editing of the synthesis and effect parameters. One can switch from one section to the other by using the *Modes*, *Synth* and *Effects* tabs located just below the layer mixer. Note that the two layers are followed by a *Master Effects* section which is identical to the *Effects* section of the individual layers.

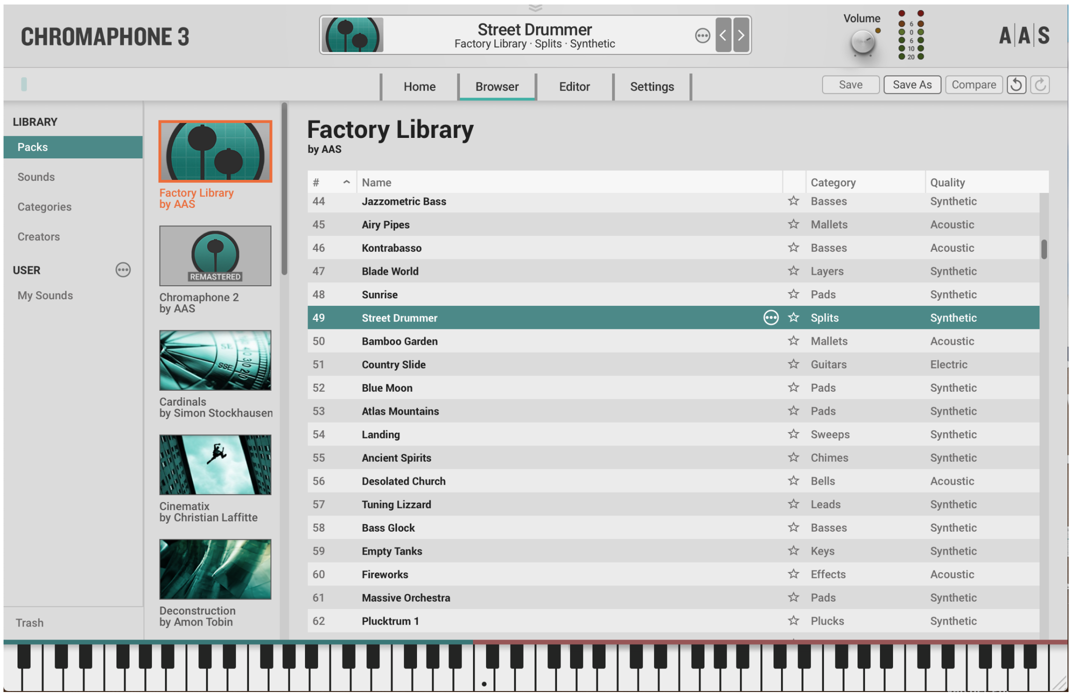


Figure 6: The *Browser* view.

Figure 7: The *Editor* view.

The Layer Mixer

The layer mixer, shown in Figure 8, includes general controls which can be applied to each layer. It is where the output level of each layer can be monitored and adjusted using the different level meters and *Gain* knobs.

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 below 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* knob. Note that it is possible to move the gain knobs of both layers by the same amount. In order to move the knobs together shift-click on a knob 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* view. 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.

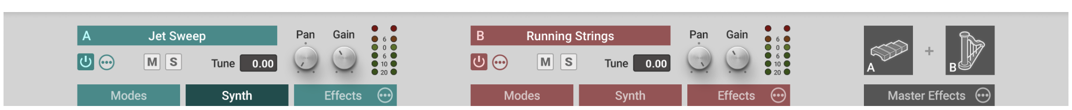


Figure 8: The layer mixer.

The Modes Section

This section includes a master clock, keyboard, unison, vibrato and arpeggiator modules as well as macro parameters which will be described in more details in Chapter 5.

The Synth Section

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

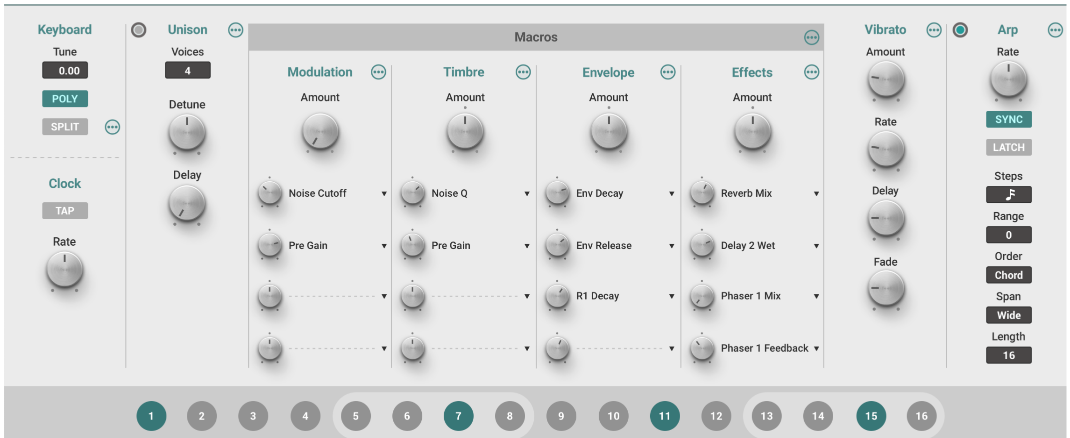


Figure 9: The *Modes* section.

Figure 10: The *Synth* section.

The Effects Section

The *Effects* section proposes a multi-effects module comprising five effects in series. The effect list includes an **EQ**, **Compressor**, **Reverb**, **Delay**, **Distortion**, **Chorus**, **Flanger**, **Phaser**, **Wah Wah**, **Auto Wah**, **Guitar Amplifier**, **Tremolo**, and a **Notch** filter. The position of a module in the signal path can be changed by click-holding on the handle to the left of the module and moving it to the desired position. The functioning of the different effect modules is described in details in Chapter 5.

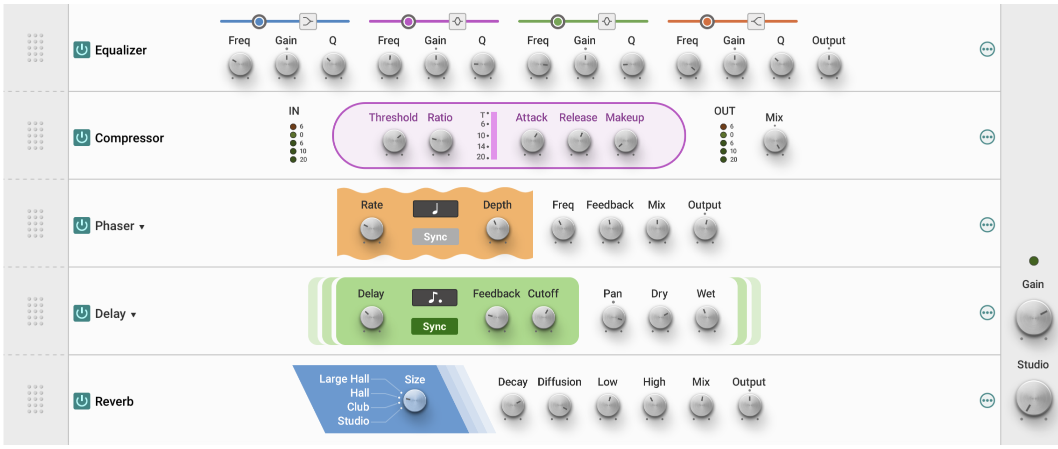


Figure 11: The *Effects* section.

2.3.4 The Settings View

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

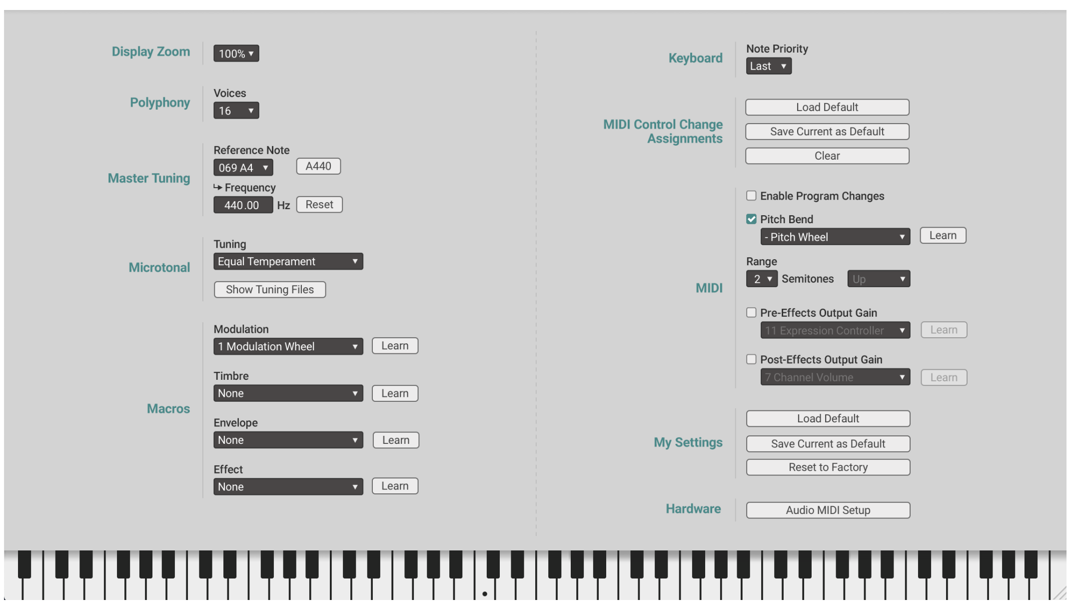


Figure 12: The *Settings* view.

3 The Home View

This is the main view for auditioning sounds and playing. The first elements of the view are displays with a representation of the resonator combination used in each layer of the current sound. The output level for each layer can be adjusted using the knob located in the lower right corner of the displays.

In the lower part of this view are four macro 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. To the left of these controls one also finds a pitch bend wheel.

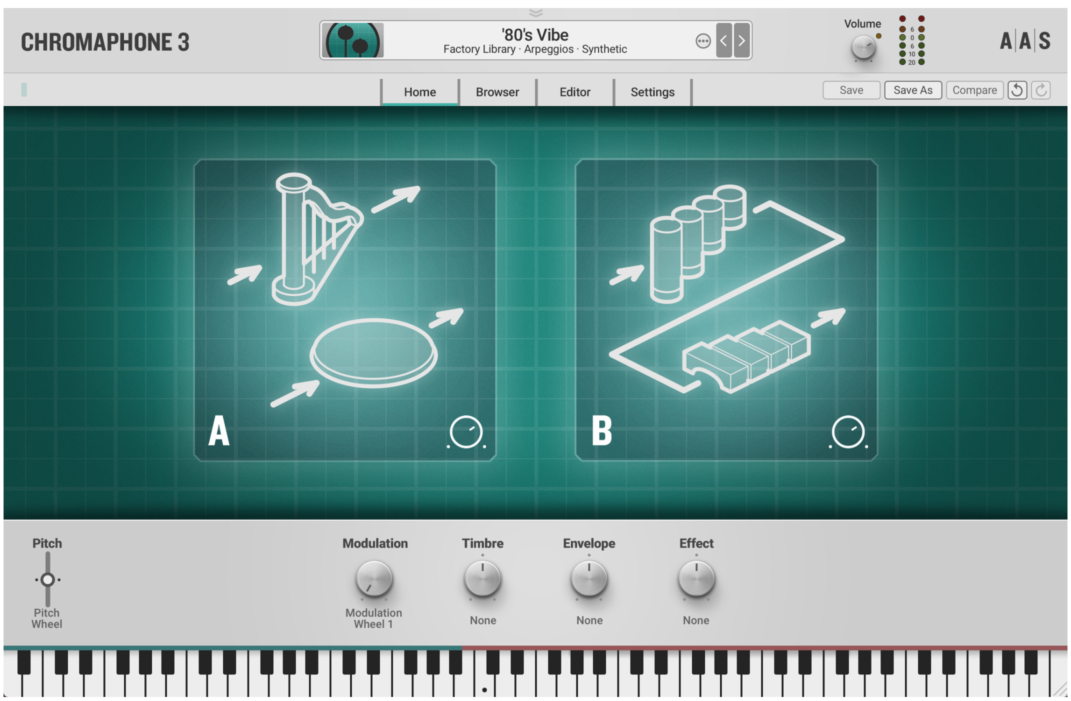


Figure 13: The *Home* view.

The range of the macro 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 a filter, 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 in its zero position, the macro has no effect and the value of the mapped parameter is kept fixed. 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 a macro and an external controller is carried out in the *Settings* view as described in Chapter 7 or by clicking on the label located just below the macro knob. 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 indicate 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.

4 The Browser View

In the context of *Chromaphone 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 *Chromaphone 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 14. 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.

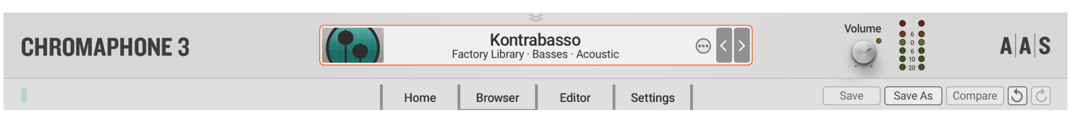


Figure 14: Currently loaded sound and sound pack.

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 14). 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 and a *edited* label also appears to the right of the sound name. Clicking on **Save** button 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 on 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* tab in the top part of the interface (see Figure 14). 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 15. Packs are divided into two categories, AAS packs and user packs. AAS packs comprise the new *Chromaphone 3* factory sounds, remastered *Chromaphone 2* 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 16 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 17 and Figure 18, 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

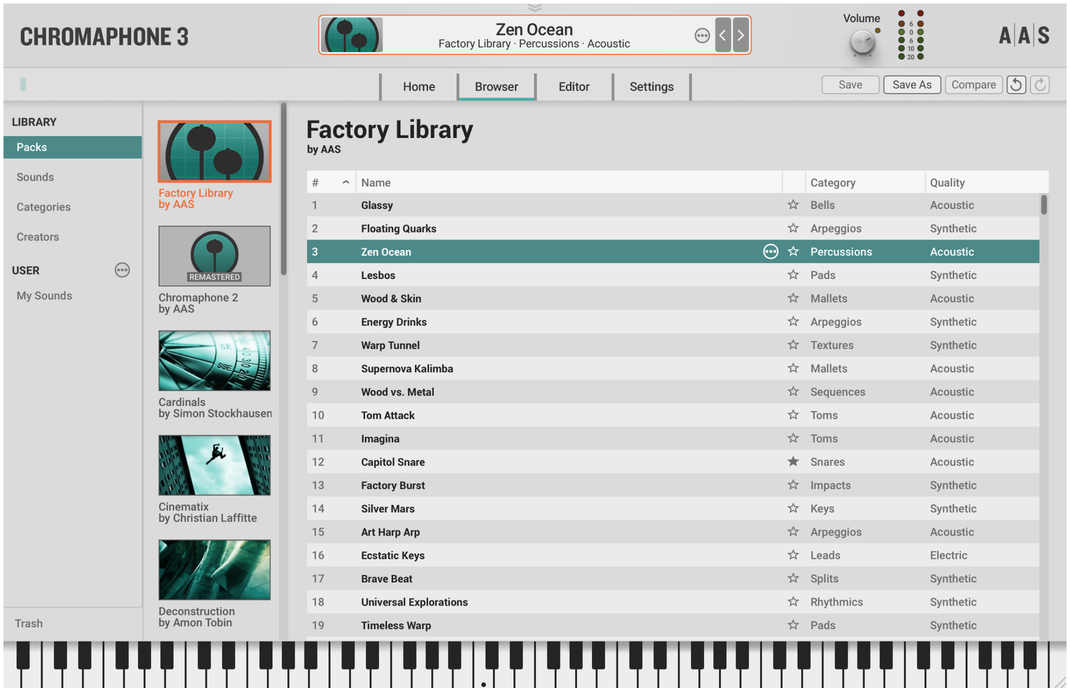


Figure 15: Browsing by sound pack.

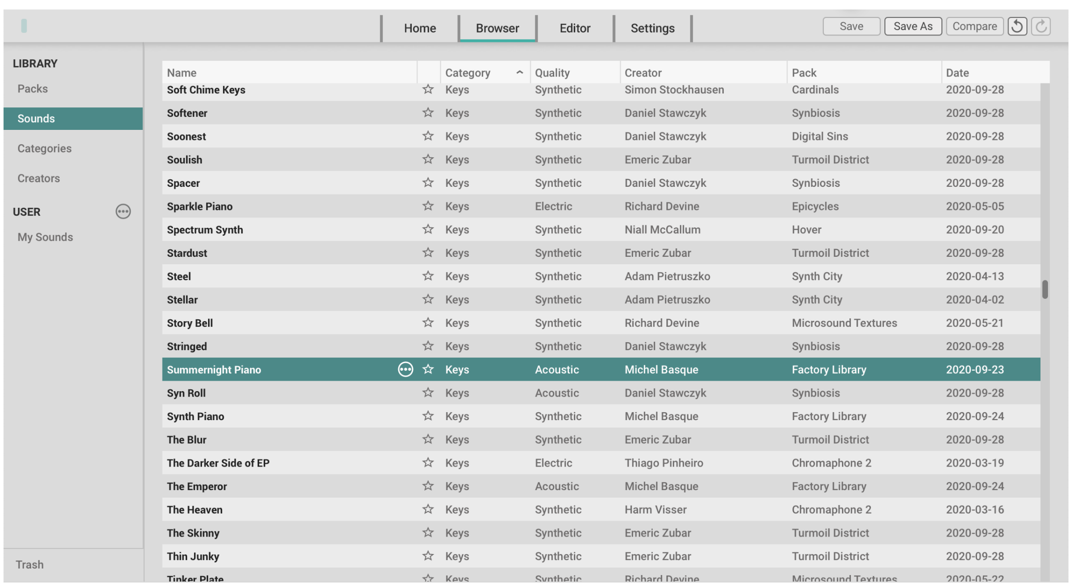


Figure 16: Browsing the entire library by sounds.

enters a name for the pack and clicks on the **Create** button. This creates an empty pack in the user

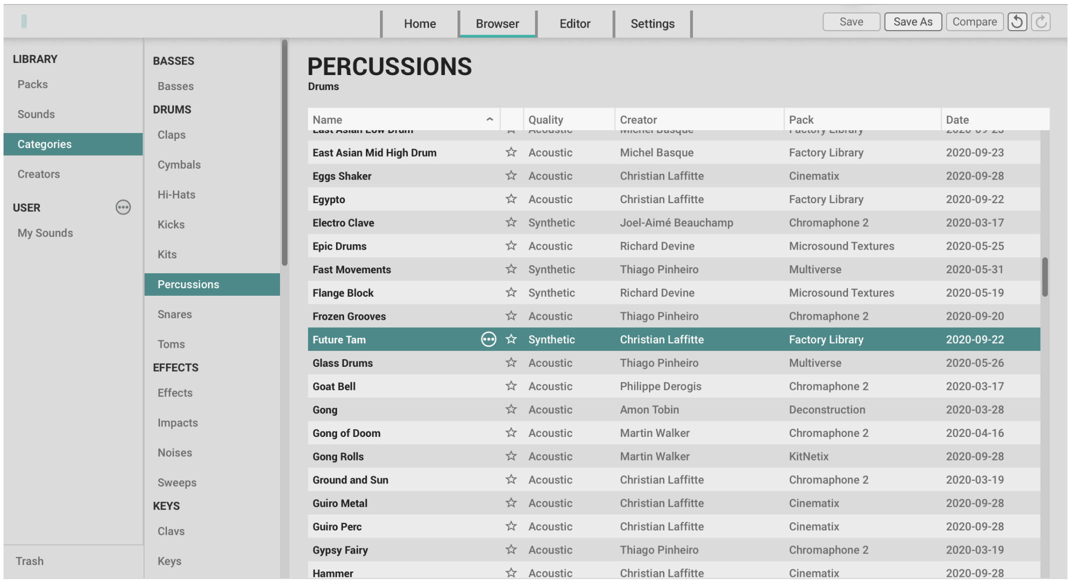


Figure 17: Browsing the library by sound category.

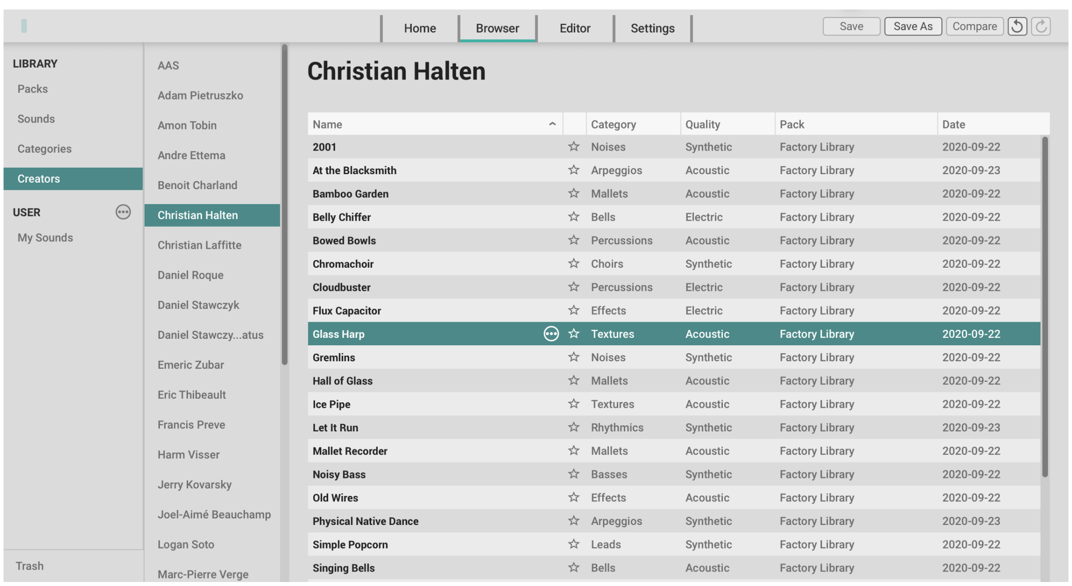


Figure 18: Browsing the library by creator.

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 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 *CP-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 *Chromaphone 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 *Chromaphone 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 *Chromaphone 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 *Chromaphone 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 *Chromaphone 3* consist of one or two layers, each layer corresponding to one instance of the *Chromaphone 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 19. It is opened by clicking by selecting the **Browse** command in the layer command menu revealed by clicking on the ellipsis icon located on the right of the power button of each layer.

Figure 19: The layer browser.

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 pack list. Layers can have a label but this is optional. When a layer has no label 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 below the layer label 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 of the *Editor* view 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 power switch. 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 *Chromaphone*

Chromaphone 3 includes a converter that allows one to import sounds from *Chromaphone 2*. The conversion operation simply involves copying an *Chromaphone 2* pack file into the *Chromaphone 3* sound pack folder. The conversion operation is then triggered automatically when *Chromaphone* detects a pack from a previous version.

Chromaphone 2 sound packs, which were then actually called banks, can be found by opening *Chromaphone 2*, clicking on the *Manage* button at the top of the *Chromaphone 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 *Chromaphone 2* sound pack files are stored while on Mac OSX, the command opens a Finder window

Once you have the *Chromaphone 2* files you wish to import, go back to *Chromaphone 3*, click on the **Browser** tab 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 *Chromaphone 2* files to be converted to this location and they will be automatically converted to *Chromaphone 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 *Chromaphone* 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 *Chromaphone* which were installed on your computer prior to the installation of *Chromaphone 3* should not be converted in this way. The *Chromaphone 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 *Chromaphone*.

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 *Chromaphone 3*. These modules are accessed from the *Editor* view which is displayed by clicking on the *Editor* tab 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* section 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

Some parameters of the synthesizer can be modulated by different signals. The modulation controls appear as colored dots or lines below or next to their corresponding parameter. Modulations sources include the MIDI pitch and velocity, signals (Key and Vel labels), the signal from the **Noise Envelope** and **LFO** modules (Env and LFO labels), as well as a random signal (RDM label).

A modulation can be viewed as the variation of a parameter around its current value controlled by a modulation signal. The different modulation controls act as gain parameters which multiply the modulation signal by a certain factor. The amount of modulation is adjusted by click-holding on a modulation dot or line (or its label) and and moving the mouse (or the finger on a track pad) either upwards and downwards or leftwards and rightwards. The amount of modulation is indicated by colored rings or lines that appear around or along the parameter control, the length of the ring or line being proportional to the amount of gain applied to the modulation signal.

Note that the colored rings (or line in the case of the *Balance* control) appear in a bold and light shade. A bold segment indicates a variation of the parameter when the value of the modulation signal is positive while a light shade indicates the direction of the change when the modulation signal is negative.

The *Key* modulation are used to modulate a parameter depending on the note played on the keyboard. When there is no modulation (no color ring), the value of the corresponding parameter is equal across the whole range of the keyboard.

The variations are applied relative to the middle C (C4, MIDI note 60) for which the parameter value is always that corresponding to the actual parameter knob. The value of the parameter then varies up or down linearly with ascending or descending pitch depending on the direction of the modulation. A bold blue ring segment indicates the direction of the parameter value change when playing high notes while a light blue segment indicates the direction of the change when playing low notes.

The *Vel* modulations are used to modulate the value of a parameter depending on the MIDI velocity signal received from the keyboard so that the value of a parameter increases or decreases as notes are played harder on the keyboard. The direction of the change is indicated by a red ring segment. In the case of the MIDI velocity modulation, the zero position corresponds to a MIDI velocity value of 64. Values from 63 to 0 will therefore follow a light colored segment while the values from 65 to 127 will follow bold segments.

Modulations using the signal from the **LFO** and **Env** modules are controlled using the *LFO* and *Env* dots and are displayed by green and orange rings respectively around the modulated parameter. The amplitude of the *LFO* modulation is proportional to the length of the green ring and it can be positive or negative depending on the orientation of the bold and light colors on the ring. In the case of the *Env* modulation, the amplitude of the modulation is proportional to the length of the orange ring segment and its direction follows its orientation.

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 *Modes* section.

5.1.6 Module Commands

All modules in the *Modes*, *Synth*, and *Effects* sections 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 20. 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.

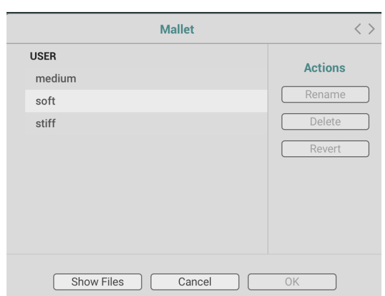


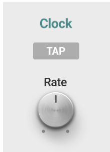
Figure 20: The module preset window.

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 Section

The *Modes* section is where the main performance oriented modules are located. It is accessed by clicking on the *Modes* tab of each layer.

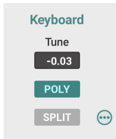
5.2.1 The Clock Module



This module is used to control the tempo of the different effects of the *Effects* section as well as that of the **LFO** and **Arpeggiator** modules when their respective *sync* button is switched *on*. When *Chromaphone 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 *Chromaphone 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

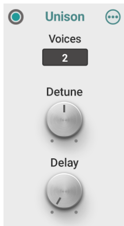


The **Keyboard** module controls how the synthesizer voices respond to the events coming from an external MIDI keyboard or from a MIDI sequencer.

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 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 *Effects*, 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 envelope modules of the synthesizer. Finally, the *Effects* 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 to the right of the four *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 the 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 a filter, 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.

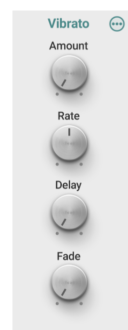
A good way to adjust these parameters is to set the *Amount* knob to its maximum value by turning it fully clockwise and then adjusting the value of the range knobs to get the maximum desired amount of modulation effect. The *Amount* knob is then set back to its zero position.

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* view and *Modes* section of the *Editor* view. The reason for this is that the *Amount* knobs on the *Home* view affect both layers of the synthesizer at the same time while those in the *Mode* sections only affect the individual layers. The knobs in the *Modes* section 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.

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.

5.2.5 The Vibrato Module



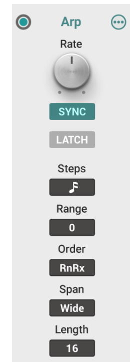
The vibrato effect is equivalent to a periodic low frequency pitch modulation. This effect is generally obtained by using an LFO to modulate the pitch signal of an oscillator. In *Chromaphone 3*, a dedicated module is provided for this effect. The *Rate* knob sets the frequency of the vibrato effect 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.

5.2.6 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.

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 (corresponding button in its *on* position). Notes are played regularly when all the steps of the display are turned *on* and rhythmic patterns are created by selecting only certain steps.

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

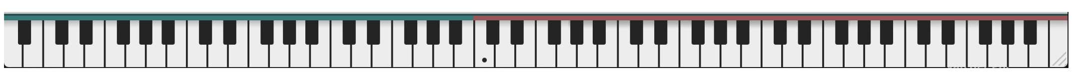
The **Arpeggiator** module is toggled in latch mode by clicking the *Latch* button to its *on* position. In this mode, the **Arpeggiator** keeps playing its pattern when the notes on the keyboard are released and until a new chord is played.

5.2.7 Virtual Keyboard

The lower part of each view includes a virtual keyboard. The keyboard covers seven octaves and notes are activated by clicking on the keyboard. The keyboard is useful to test sounds when no MIDI keyboard is connected to your computer.

5.2.8 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 of the **Keyboard** module. 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 *Chromaphone 3*.



5.3 The Synth Section

The **Synth** section is accessed by clicking on the *Synth* tab of a layer. This section allows one to go 'under the hood' and tweak the core synthesis modules of *Chromaphone 3* and therefore to customize its tone and behavior.

5.3.1 General Notions of Acoustics

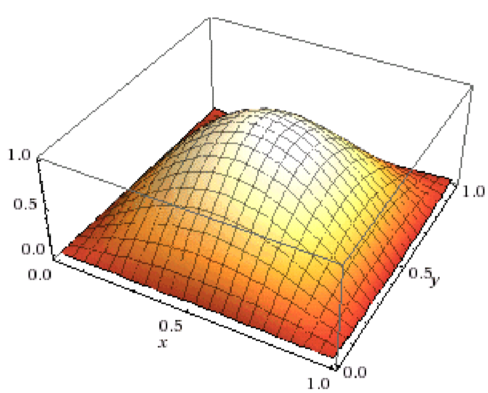
Normal Modes

Exciting an object such as the skin of a drum by hitting it with a mallet results in a complex vibrational motion. It is this vibration of the object that will create pressure waves in the surrounding air which will propagate to our ears as sound waves.

Mathematically, a complex vibrational motion can be decomposed into elementary motion patterns called the *normal modes* of the object. Under a normal mode, all the parts of the structure move in phase and at the same frequency in a sinusoidal motion. In other words, this complex motion results from the fact that objects naturally oscillate at many different frequencies at once, each frequency being related to a normal mode of vibration. These frequencies are called *partials*; the lowest partial is called the *fundamental* and the higher ones are referred to as *overtones*. When relating to music, the fundamental corresponds to the *note* played and the overtones are called *harmonics* as in most musical instruments their frequency is a multiple integer (or almost) of the fundamental.

As an example, the vibration motion associated with two normal modes of a rectangular plate is illustrated in Figures 21 and 22. In the first figure, one can see the vibration motion associated with two different normal modes of the plate (modes [1,1] and [3,2]). Over one period of oscillation, all the points go up and down in phase. The principle remains the same for all mode, the motion pattern only becoming more and more complex as the order of the mode increases. The full motion of a plate, however complicated, will always correspond to a combination of all its normal modes. Figure 22 is a top view of the plate and shows contour lines corresponding to the same normal modes. A contour line groups points that oscillate with the same amplitude. In particular, the straight lines in the second graph of this figure, corresponds to so-called *nodal lines* where the amplitude of the motion is zero and therefore where the plate is still.

The relative frequencies or ratio of the frequency of the overtones to the fundamental frequency is specific to the type of the object and its boundary conditions (whether its boundaries are free to vibrate or are fixed). In other words this distribution of partials is characteristic of the type of object and could be viewed as its tonal signature; it allows us to distinguish, for example, a vibrating plate from a drumhead. The specific frequency of the partials, related to the sensation of pitch, is determined by the dimensions of the object, for example a small plate will have a higher pitch than an larger one.



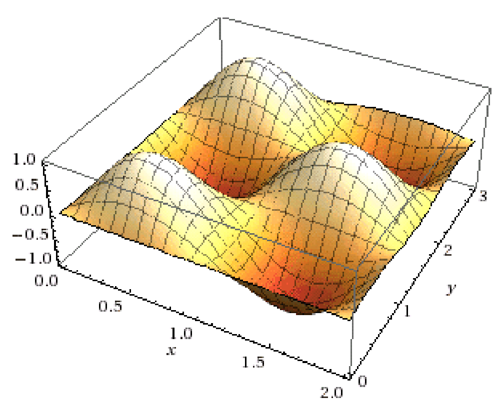


Figure 21: Motion corresponding to normal mode [1,1] and [3,2] of a plate.

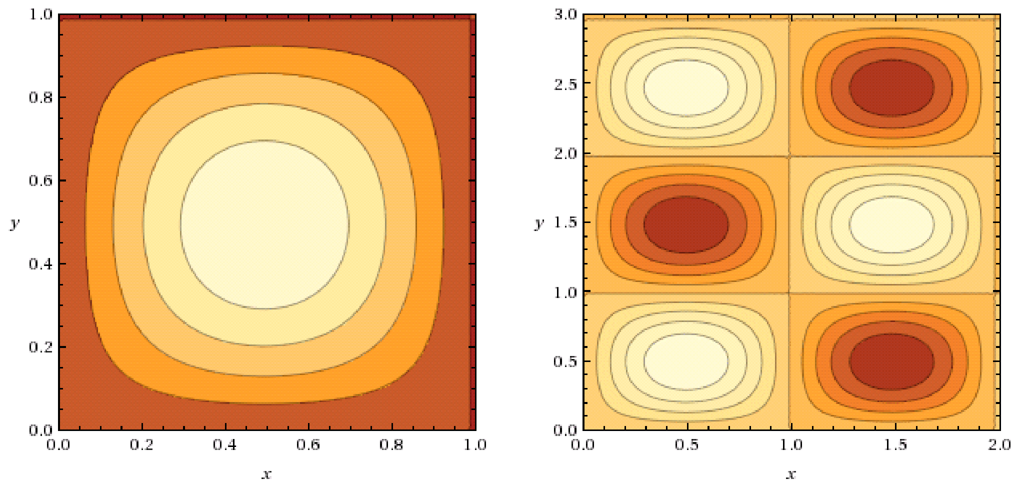


Figure 22: Contour plot corresponding to normal mode [1,1] and [3,2] of a plate.

But this is not all, we can distinguish different types of objects, such as a vibrating plate and a beam, but also two objects of the same type but made out of different material. For example a metal plate will sound brighter and have a longer decay than a wooden plate. This is due to the fact that the physical properties of an object depend on its material which determine the relative *amplitude* and *phase* of the different partials as well as their damping, a measure of how fast they will decay once excited. The specific amplitude, phase and damping of each partial therefore determine the specific tone of the object as well as how it evolves with time.

There is finally one more parameter which affects how an object sounds, it is the point of excitation. Indeed, a drumhead does not sound the same if it is hit in the middle or near the rim of the drum. This can be understood by the fact that exciting an object on a point located on a nodal line of a mode (a line where amplitude of the motion associated with a mode is zero) does not allow the transfer energy to that specific mode and its corresponding partial will not be excited. The effect will not be as pronounced but will still exist as the excitation point is moved around the nodal lines which explains how the excitation point influences the relative amplitude of the partials and therefore the tone.

Coupling of Resonators

One of the key features of *Chromaphone* is that it allows one to couple objects together, in other words to take into account the interaction between objects as opposed to simply feeding the signal from one object to the other. This is very interesting because this interaction between components results into a new object which, while being related to its original elements, behaves and sounds differently. In fact, musical instruments are based on combinations of objects such as a string and a soundboard for a guitar, a bar and a tube in the case of a vibraphone or a skin and a column of air in a drum.

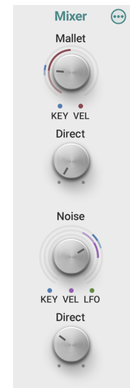
The coupling of objects results in a bidirectional transfer of energy between the objects. In physical terms, the amount of exchange is determined by the relative value of the mechanical impedance of the different objects. The impedance is a notion which measures how much an object opposes motion when subjected to a force. It is a frequency domain function as the response of an object can vary greatly with frequency. For example the amplitude of the motion of an object will be much greater when excited at a resonance frequency.

In simpler terms, the effect of coupling can be understood by considering how rigid one object is compared to the other which determines how much energy can be transferred from the first object to the second one. Let's imagine a string attached to a very stiff sound board. While some energy will be transmitted to the sound board through the bridge, it will not greatly affect the motion of the string; most of the energy will be reflected back into the string at the bridge resulting in a standing wave in the string and a long decay. Now let's imagine that the soundboard becomes much less rigid. The string can now set it into motion more easily at the bridge. This implies that more energy will be able to flow from the string to the soundboard resulting in a shorter decay as less energy is reflected back into the string. But the soundboard also moves according to its own vibration modes which are different from that of the string. This motion interacts with that of the string which modifies the tone that we hear. One could say that we now hear more the soundboard in the resulting sound. The amount of coupling between the resonators therefore affects both the resulting tone and its decay time.

The material of the objects is not the only thing to consider. Their respective tuning, which can be related to their geometry, also greatly influences the response of the combined objects. For example if the objects are tuned at the same fundamental frequency, their respective motion will be synchronized and result in a sound having a large amplitude. For example, in a vibraphone, the tubes are tuned to the fundamental of the bar above them in order to amplify the fundamental. But there is also another effect which might seem contradictory at first. The fact that energy is well transmitted from the bar to the tube also implies a faster decay of the oscillations. Hence, the overall effect of the combination of the bar and the tube is to amplify the fundamental while decreasing the decay time of the note.

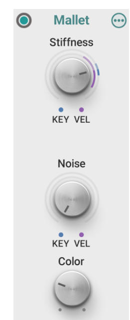
As we can see, the overall effect of coupling can be quite complex as many factors must be taken into account. As a rule of thumb, in traditional musical instruments, a first resonating object with a long decay is usually coupled to a second resonator having a very short decay time (try knocking on the sound board of a guitar) in order to avoid unpleasant resonance effects.

5.3.2 The Mixer Module



The two Chromaphone resonators can be excited by a mallet and a noise source. The **Mixer** module is used to adjust the amplitude of both of these sources. The *Mallet* knob is used to adjust the amplitude of the force impact from the mallet while the *Noise* knob controls the amplitude of the noise source. Both of these parameter can be modulated with pitch and MIDI velocity. The noise source can also be modulated with the **LFO** module. The two *Direct* knobs are used to add signal from the mallet or noise source to the output signal from the resonators. When in their leftmost position, there is no extra source sound added to the output signal and the source component that is present in the output sound is the original sound from the sources filtered by the resonator(s). Turning these knobs clockwise adds an increasing amount of direct source signal in the output sound.

5.3.3 The Mallet Module



The **Mallet** module is used to simulate the force impact produced by a mallet striking an object. The force of the impact is adjusted with the *Mallet* knob from the **Mixer** module as described above while the stiffness of the mallet (related to its material) is varied with the *Stiffness* knob. Figure 23 shows the effect of the adjustment of the stiffness on the output signal. As the stiffness is increased the excitation signal becomes narrower. The effect of the amplitude of the force impact is also shown in the same figure. The *Stiffness* parameter can be modulated with the MIDI velocity and the note played. These modulation, combined with a corresponding modulation of the *Mallet* parameter from the **Mixer** module are usually used to get a stronger impact with increasing keyboard velocity and to make the mallet softer as the impact velocity increases, a behavior one observes, for example, on piano hammer heads due to the non-linearity of the felt.

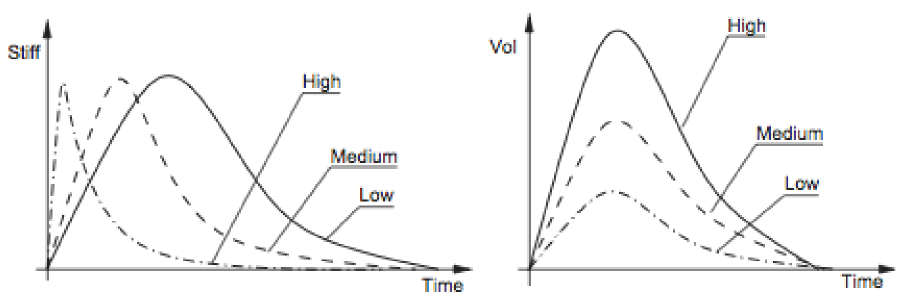


Figure 23: Effect of the *Stiffness* and amplitude of the force impact (*Mallet* knob from the **Mixer** module) on the output from the **Mallet** module.

Noise can also be added to the impact sound allowing for some interesting effects. The amount of noise is controlled with the *Noise* control. In its leftmost position there is no noise added to the signal and one only hears the impact noise. Turning this knob clockwise gradually increases the amount of noise. The frequency content of the noise can be adjusted with the help of the *Color* control. Turning this knob clockwise increases the cut-off frequency of a high-pass filter.

5.3.4 The Noise Module

The **Noise module** is an alternate way to excite the resonator. This module can be used to add noise to the impact signals from the **Mallet** module but, with its associated envelope generator, it also allows one to produce long excitation signals, very different from the impact-like signals from the **Mallet** module, and add sustain to the sound.

The source of this module is a white noise generator whose output can be filtered using the different filters available from the *Filter* drop-down list at the top of the module. Available filter types are: resonant low-pass, resonant high-pass, band-pass, and low-pass and high-pass in cascade allowing for a flat response in the pass band. There is also a graphic mode allowing for precise multi-band shaping of the noise source.

The amplitude of the noise source is controlled using the *Noise* knob from the **Mixer** module and the envelope signal from the **Envelope** module. This parameter can further be modulated with the pitch or velocity signal from the keyboard or with the output from the **LFO** module.

The *Frequency* control is used to adjust the cut-off or center frequency depending on the type of filter used to shape the noise source. This parameter can be modulated with the the pitch or velocity signal from the keyboard or with the output from the **Noise Envelope** or **LFO** module. The third control for this module has different values depending on the type of filtering applied to the noise source. When a resonant filter is chosen, the label for this parameter is *Q* and the parameter controls the resonance or quality factor of the filter. In the case where a combination of low-pass and high-pass filter is chosen, the label is *Width* and the parameter controls the width of the pass-band of the resulting filter.

In the case when the *Graphic* option is chosen in *Filter* list, the noise source is shaped by a filter bank. The *Frequency* and *Q* knobs are replaced by ten sliders each one being associated with a specific frequency band. The different bands are controlled by a band-pass filter except for the first and last bands which are controlled by a low and high pass filter respectively. The amplitude of each band can be adjusted from $-\infty$ to zero dB. When all the sliders are in their rightmost or 0 dB position, the spectrum of the noise source is flat. Moving any slider to the left decreases the amplitude of the noise source in the corresponding frequency band until it is completely removed when the slider reaches its leftmost position. Another way to work with these filters is to put all the sliders in their leftmost position, equivalent to switching off the noise source, and then adding noise in the desired frequency bands.

The last parameter of the module is called *Density* and it is used to control the rate at which random samples are fired by the module. When this control is in its left position, the density is low and one can clearly hear individual random noise samples which may sound like individual particles hitting the surface of the resonators. Increasing the noise density by turning the knob clockwise increases the number of clicks generated in a given interval of time until the output starts to become continuous. This parameter can be used to produce interesting effects by exciting the resonators randomly. This parameter can be modulated with the the pitch or velocity signal from the keyboard or with the output from the **Noise Envelope** or **LFO** module. The density parameter also has a sample and hold feature which is turned *On* using the *sh* switch to the right of the *Density* knob. When activated, a noise sample is held until a new one is triggered. This features affects the color of the noise but is mainly there for compatibility reason with presets from version 1.

5.3.5 The Resonator Module

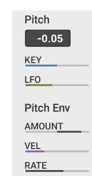
In *Chromaphone*, instruments are created by forming pairs of acoustic resonators. The excitation signal from the **Mallet** and/or **Noise** source modules is sent to the resonators which can be arranged in a series or parallel configuration. Resonator A and B can be turned *On* or *Off* by clicking on the green led in the top-left corner of each module.

The *Resonator* selector allows one to choose the type of resonator used. The resonator type can be changed by clicking on the resonator icons or by using the drop-down menu at the top of the icon display. The list of resonators include the main type of objects used in the making of musical instruments. Available types are:

- **String**: a perfectly elastic string,
- **Beam**: a rectangular beam with constant cross-section,
- **Marimba**: a beam with variable section allowing one to obtain partials having a quasi-harmonic ratio,
- **Plate**: a rectangular plate,

- **Drumhead**: circular membrane,
- **Membrane**: rectangular membrane,
- **Open Tube**: a cylindrical tube with both ends open allowing one to obtain the complete harmonic series (even and odd harmonics),
- **Closed Tube**: a cylindrical tube with one end closed allowing one to obtain only odd harmonics,
- **Manual**: In this mode, one can create a custom resonator by selecting up to four partials (see *Quality* control). The rank of each partial is fixed using the *Partial 1* to *Partial 4* selectors.

The *Mode Density* control is located just below the resonator selector and it allows one to adjust the number of modes taken into account in the synthesis and therefore the richness and complexity of the sound. This control has four positions, *Low*, *Medium*, *High*, *Full*, corresponding to 4, 16, 30 and 70 modes respectively. When the resonator is a **Tube**, this control is deactivated and all modes are taken into account. Note that the CPU time required by a resonator is proportional to the number of modes calculated; the higher the number of modes used, the higher the CPU load. In the particular case where the **Manual** resonator type is selected, this control is used to determine how many of the four available partials will be used to form the resonator.



The reference pitch of a resonator, or in other words the frequency of its first partial, is adjusted using the *Pitch* parameter. 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). When the semi-tone and cent controls have a value of zero, the reference pitch of the object is the middle C of the piano (C4 = 261.62 Hz). The value of the reference pitch can be adjusted by click-dragging on the semi-tone and cent controls. Double clicking on these controls brings back their value to zero.

The *Key* control determines how the pitch varies as a function of the note played on the keyboard. When this parameter is zero, the pitch does not vary and therefore it is the same whatever the note played on the keyboard. When this control has a value of 1.00:1 (one semi-tone for each semi-tone on the keyboard), the pitch of the object follows the pitch of the note played on the keyboard or in other words, the pitch variation is tempered. Using values smaller or higher than 1.00:1 results in intervals smaller or greater than a semi-tone when adjacent notes are played on the keyboard. The pitch can also be modulated using the signal from the **LFO** module. The *LFO* control is used to adjust the amount of gain applied to the signal from the **LFO**.

The *Level* and *Rate* controls are used to obtain a modulation of the pitch when a note is played. The *Level* control is used to determine the amount by which a note is detuned when it is triggered. The *Rate* control sets the amount of time before the note reaches its normal pitch. Note that the value of the *Level* control can be positive or negative allowing the note to start above or below its real pitch. It can also be modulated by the MIDI keyboard velocity. This adjustment is obtained using the *Vel* control.

The decay time of the partials of the object is determined by the *Decay* control. This parameter can be modulated as a function of the note played on the keyboard and its MIDI velocity using the *Key* and *Vel* modulation parameters respectively. Note that in the case of a **Tube** object, the decay time of the sound is also affected by the *Radius* parameter. In that case, the total decay time will be determined by the cumulative effect of the *Decay* and *Radius* parameters. Note that the decay time of instruments with coupled resonators also depends on the amount of coupling.

The *Rel* parameter is used to simulate the effect of dampers on the object when a note is released. The release time is calculated as a percentage of the total decay time of the object as set by the *Decay* parameter.

The *Material* control allows one to fix the decay time of partials as a function of frequency with respect to that of the fundamental. This is a parameter characteristic of the material of the object. When this parameter is set to a value of zero, all partials decay at the same rate, that fixed by the position of the *Decay* control. Adjusting the *Material* control to a negative value favors low frequencies by decreasing more and more the decay time of partials as their frequency increases. When this control is set to a value of -1, the decay time will be inversely proportional to the frequency of the partial. Thus a partial with a frequency twice as great as that of the fundamental will have a decay twice as short as that of the fundamental, a partial with a frequency three time as great will have a decay time three times shorter and so on. Using a positive value for this parameter has an opposite effect as the low partials then decay more rapidly than the higher ones. When this parameter is set to a value of 1, the decay time is proportional to the frequency of the partial. For example, the decay time of a partial with a frequency twice as great as that of the fundamental will have a decay twice as long as that of the fundamental and so on.

The *Tone* control is used to adjust the amplitude of the partials as a function of frequency with respect to that of the fundamental. When this control is adjusted to a value of zero, all partials have the same amplitude. When this control is set to a negative value, the high partials have a smaller amplitude than the low ones. For example, a value of -6dB/octave results in the amplitude of the partials being inversely proportional to their frequency. Thus a partial having a frequency twice as great as that of the fundamental will have an amplitude twice as small (-6 dB), a partial with a frequency four times that of the frequency will have an amplitude 4 times smaller (-12 dB) and so on. When this control has a positive value, the effect is inverted. The low frequency partials then have a smaller amplitude than the higher ones. For example, when this parameter is set to a value of +6 dB/octave, the amplitude of the partial is proportional to its frequency. Thus a partial with a frequency twice that of the fundamental will have an amplitude twice as great (+6 dB) as that of the fundamental and so on. Note that these amplitude values can further be modulated by the excitation position (see *Hit Position* control) which is a parameter affecting the relative amplitude of the partials.

The *Low Cut* parameter gives additional control on the low frequency response of the resonator by applying a -24 dB per octave low-cut filter. This control is useful when clearer sounds are desired. The *Low Cut* knob is used to adjust the cut-off frequency of the filter. In its leftmost position, the low cut filter is inactive and the sound is not affected. Turning the knob clockwise displaces the cut-off frequency towards higher frequencies following steps corresponding to harmonics numbers thereby removing more and more low frequency content in the sound.

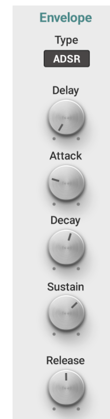
The *Radius* parameter replaces the *Material* control when a **Tube** object is selected. In fact, standing waves in a tube do not result from the vibrations of the walls of the tube but rather by vibrations of the air column inside the tube. The material of the tube is therefore not a relevant parameter in that case. The effect of the *Radius* parameter can be viewed as that of a high-pass filter with the cut-off frequency of the filter increasing as the radius of the tube is decreased. In other words, the smaller the radius, the brighter the sound. The radius of the tube also affects the total decay time of the object, the decay time being shorter for large radii as a result of larger radiation losses at the open ends of the tube. The *Radius* control on the interface has been adjusted to follow the same behavior as that of the *Decay* one, in other words to obtain longer decay time as it is turned clockwise. Even if this may seem contradictory at first, this implies that the actual radius of the object decreases has the value of the parameter is increased.

The *Hit Position* controls where the excitation signal is applied on a resonator. This is an important parameter as it affects the relative amplitude of the different partials of the resonator and therefore the spectrum of the sound it radiates as explained in Section 5.3.1. This position is indicated as a percentage of the total size of the object. The minimum value of the control corresponds to an excitation applied on the border of the object while the maximum value corresponds to an excitation applied on its center. In the case where both resonators are coupled, the *Hit Position* setting of resonator A represents the location where the excitation signal is applied while this setting on resonator B represents the point where the extremity of Resonator A is coupled to resonator B. As the tone of the resonator varies with the excitation position, it is interesting to modulate this position while playing. This is possible using the *Vel*, *Key* controls which are used to adjust the amount of modulation from the keyboard velocity, pitch signal respectively and the *Rnd* control which applies a random modulation.

The *Coupling* selector is used to determine if the two resonators are coupled or not. In the *Off* position, the resonators are not coupled and excited simultaneously. They are, in other words, in a parallel configuration. The output signal is then a mix of the signals from the two resonators in a proportion determined by the setting of the *Balance* slider. When in its center position, an equal amount of signal from resonator A and B is present in the mix. More signal from resonator A or B is obtained by adjusting the balance slider up or down.

The two resonators are coupled when the *Coupling* control is in the *On* position. In this case, resonator A receives the excitation signal and energy is exchanged between the two resonators through coupling which creates a new object whose characteristics depend on the parameters of the two objects. In coupling mode, the *Balance* slider is used to adjust the impedance ratio, in other words how easy it is to set one object into motion compared to the other. In the A position, the impedance of resonator A is lower than that of resonator B implying that resonator B is very stiff compared to resonator A. As a result, most of the energy is reflected back into a at the junction point and resonator A is not much affected by resonator B; one mostly hears resonator A. Increasing this parameter decreases the impedance of resonator B with respect to that of resonator A affecting more and more the functioning of the first resonator. Below the center position, the impedance of resonator B is lower than that of resonator A resulting in a change in the limit conditions of resonator A and hence the frequency of its fundamental and partials depending on the settings of resonator B. In other words, one starts to hear resonator B more and more in the final sound. The amount of coupling or balance (in the case where they are in parallel mode) between the resonators, can be modulated with the pitch of the note played with the *Key* control.

5.3.6 The Noise Envelope Module



This module is an envelope generator used to modulate the amplitude of the noise source as well as its *Frequency* and *Density* controls. The envelope generator can be operated in ADSR or AHD mode. The *Type* drop-down control is used to select between these options.

In ADSR mode, the envelope is divided in four phases: *Attack*, *Decay*, *Sustain* and *Release* as illustrated in Figure 24. During the attack phase, the envelope signal goes from a value of zero to a value of 1 in a laps of time controlled by the *A* knob. The decay phase then begins and the signal goes from 1 to the sustain value of the signal in a laps of time controlled by the *D* knob. The level of the sustain portion of the modulation signal is adjusted using the *S* knob. This value is held as long as a note is depressed. Upon release of the note, the signal then decreases from its sustain value to zero in a laps of time controlled by the *R* knob. If the note is released during the attack or decay phase, it will switch to the release phase and decay to zero. The *Delay* knob of this module is used to add a delay between the triggering of a note and the start of the envelope. This is useful to add noise to the excitation signal following the initial impact noise from the **Mallet** module.

The AHD mode is used to create envelopes for short attack sounds such as in one-shots. In this mode, the envelope is divided in three phases: *Attack*, *Hold*, and *Decay* as illustrated in Figure 25. Once triggered, the complete envelope signal is generated even if the note is released before the end of the envelope itself. During the attack phase, the envelope signal goes from a value of zero to a value of 1 in a time interval controlled by the *A* knob. The envelope signal then remains at this peak value during a time determined by the *H* knob. The signal then decreases from this value to zero in a lapse of time controlled by the *D* knob.

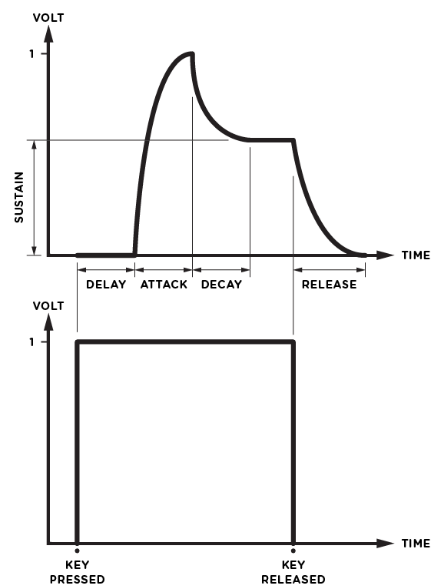


Figure 24: ADSR Response curve.

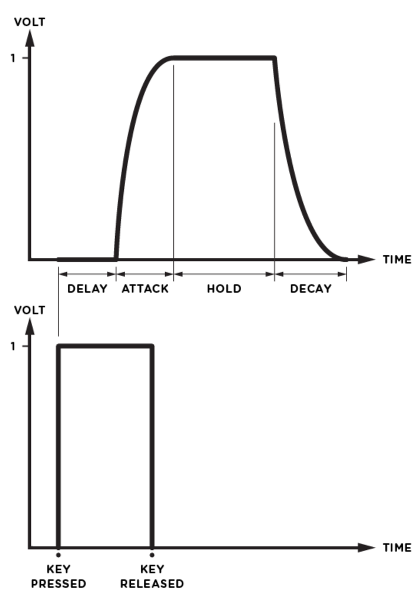
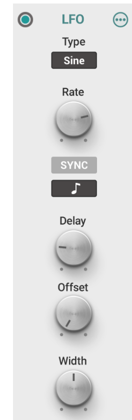


Figure 25: AHD envelope response curve.

5.3.7 The LFO Module



The **LFO** module is used as a modulation source for the **Noise** source module. The waveform of the **LFO** is selected with the *Shape* drop-down menu on the top of the module. The possible values are *Sine*, *Triangular*, *Square*, *Random* and *Random Ramp*. The shape of the triangular and square waveform can be varied using the *Width* parameter. In the case of the triangular wave, the waveform is thus varied gradually from a triangular shape in the middle position to a sawtooth shape starting at its lowest value and going up when the knob is turned to its leftmost position to a sawtooth starting from its maximum point and going down when the knob is fully turned to the right. In the case where the square wave is selected, the waveform is square when the knob is in its center position and is transformed gradually to a smaller and smaller pulse as the knob is moved anti-clockwise and to a an increasingly rectangular wave when moving the knob clockwise from its center position. When the waveform is set to *Random*, the **LFO** module outputs random values at the rate determined by the *Sync* control or the *Rate* knob. In this case, the output value from the **LFO** module remains constant until a new random value is introduced. The *Random Ramp* 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.

There are two ways to adjust the rate, or frequency, of the output of the **LFO** module. If the *Sync* control is in its *off* position, the rate is fixed with the *Rate* knob. When the *Sync* control is *on*, the frequency of the oscillator is fixed relative to the frequency (tempo) of the host sequencer and the value set by the *Sync* control. Sync values range from 16 quarter notes (4 whole notes) to 1/8 of a quarter note (a thirty-second note) where the duration of the whole note is determined by the host sequencer. The **LFO** module can also be synced to a triplet (t) or a dotted note (d).

The *Delay* control allows one to insert a delay between the moment a note is played and the triggering of the **LFO** module. Finally the *Offset* parameter determines the point in the waveform from which the **LFO** module is triggered. In its left position, there is no offset and the waveform starts with with a zero phase. Increasing the *Offset* parameter moves the starting point later in the waveform. For example, if a sine wave is selected and the offset adjusted to a value of 25%, the starting point will correspond to a quarter of a period and therefore to a positive peak of the waveform and the signal will start decreasing. A value of 75% would correspond to three quarter of a period and therefore a negative peak and the signal value would then start increasing.

5.4 The Effects Section

The **Effects** section 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 (labelled *Master Effects)* located after the layer mixer in signal flow. The individual effects modules are identical in each of these multi-effects modules.

The Effects modules allows one to process and shape the signal. The type of each module can be set by using the drop-down menu located to the left of each module. They are turned *on* or *off* by using the power button located just before the name of the effect. The effect list includes an **EQ**, **Compressor**, **Reverb**, **Delay**, **Distortion**, **Chorus**, **Flanger**, **Phaser**, **Wah Wah**, **Auto Wah**, **Guitar Amplifier**, **Tremolo**, and a **Notch** filter. The order of these effect modules in the signal flow is adjusted by click-holding the handles located on the left of the module and dragging the module to the desired position.

5.4.1 Equalizer

The **Equalizer** module provides equalization over the different frequency bands or a signal. It is composed of four filters in series each offering a low shelf, high shelf, low pass, high pass and peak filter option.

The functioning of the low shelf filter is depicted in Figure 26. 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 26. 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.

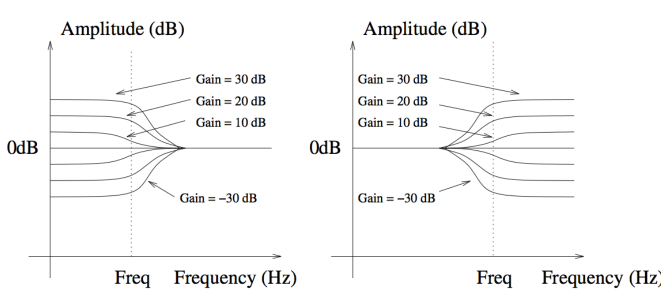


Figure 26: Low and high shelf filters.

The **Equalizer** module features two peak filters, labeled **LMF** and **HMF**, allowing to shape the signal in two frequency bands as illustrated in Figure 27. 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 ± 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.

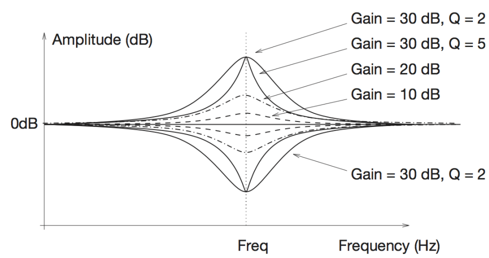
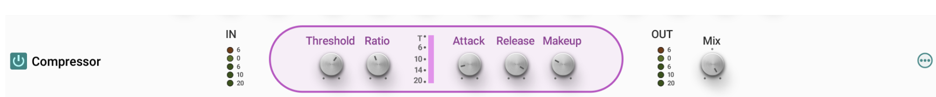


Figure 27: Peak filter.

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.

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).

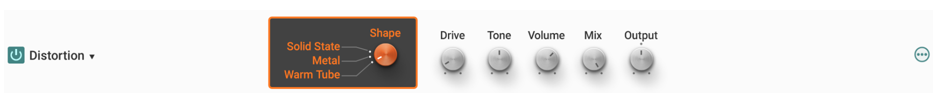
The *Delay knob* is used to adjust the amount of delay, in seconds, introduced by the effect. Turning this knob clockwise increases the delay. The *Feedback* parameter is a gain factor, varying in the range between 0 and 1, applied to the signal at the end of the delay lines. It controls the amount of 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.

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.

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 28. 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.

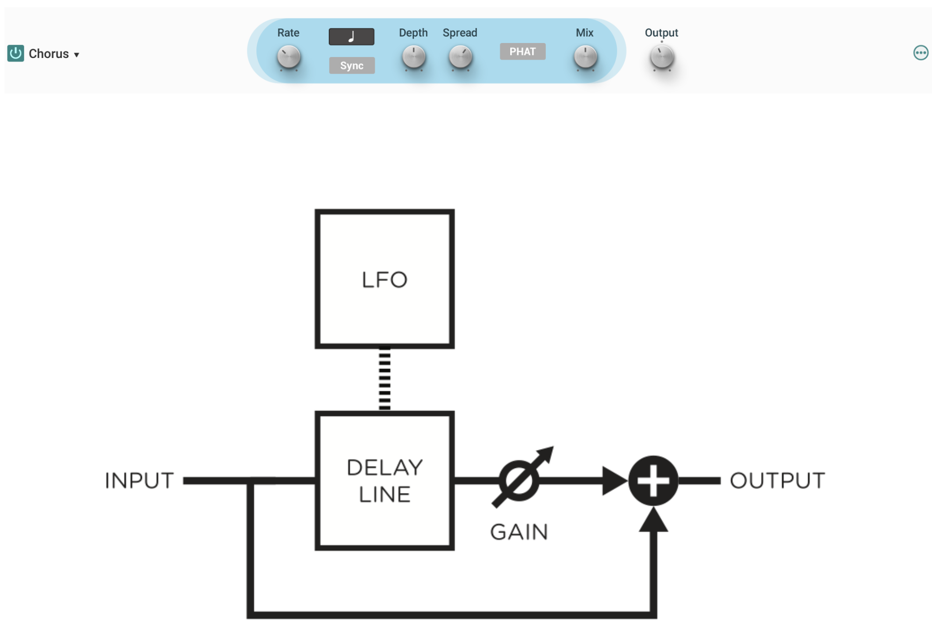


Figure 28: **Chorus** module.

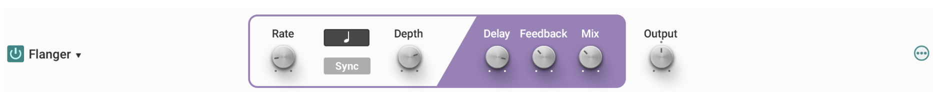
Tuning

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 29. 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.

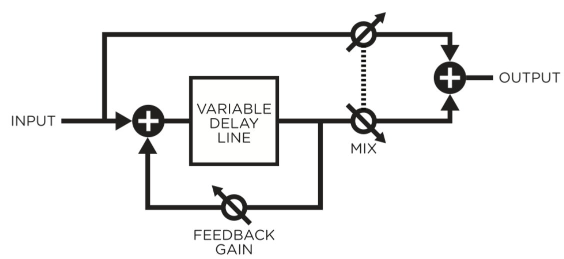


Figure 29: **Flanger** algorithm.

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 30. 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/(2delay)), 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.

The amount of effect is determined by the ratio of wet and dry signal mixed together as shown in Figure 31. 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

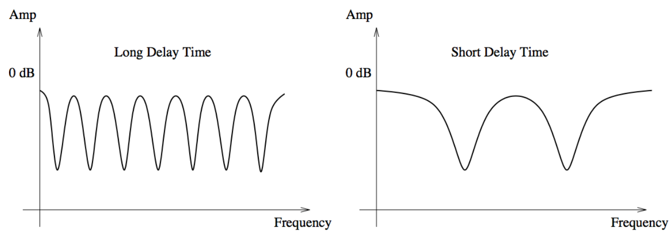


Figure 30: Frequency response of a **Flanger** module. Effect of the length of the delay line.

by the amount of wet signal re-injected into the feedback loop as shown in Figure 32. 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.

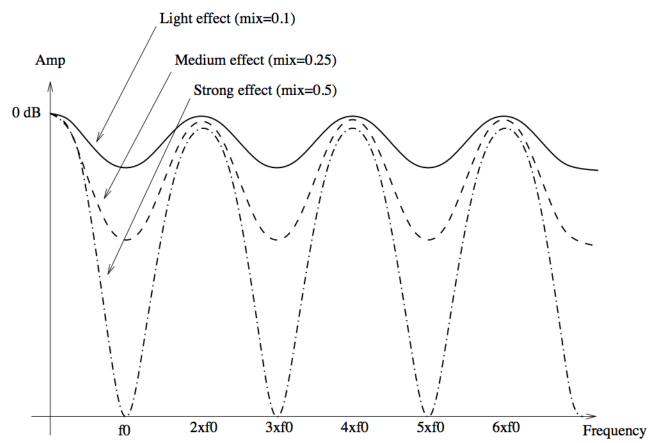


Figure 31: Effect of the mix between wet and dry signal on the frequency response of a **Flanger** module

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, 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

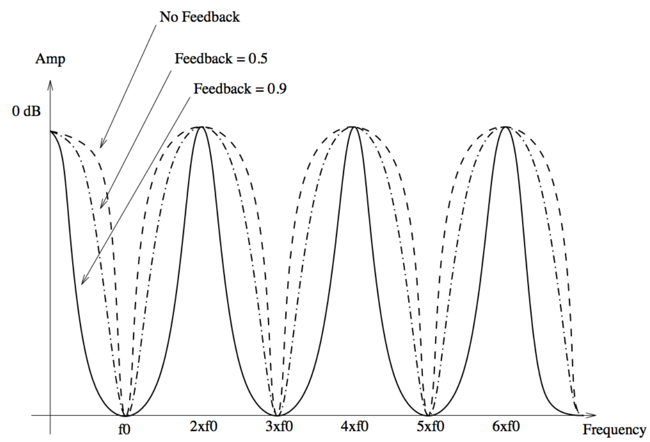


Figure 32: Effect of the amount of feedback on the frequency response of a **Flanger** module.

sent to the output.

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 33. 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 34. The rejection is maximum when the phase delay is equal to 180 degrees and a given component is out 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 34. 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.

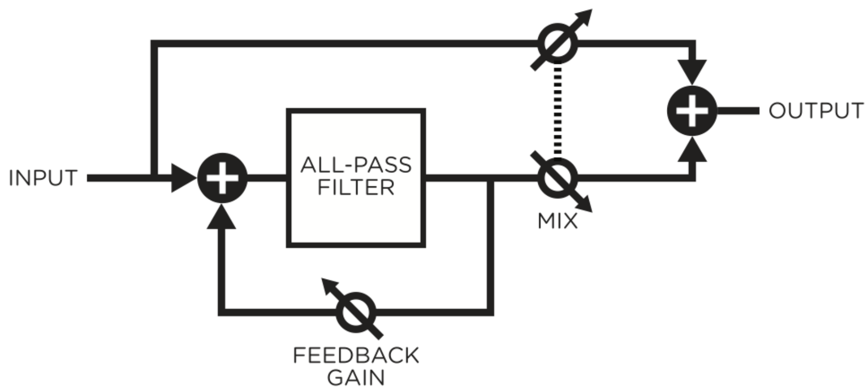


Figure 33: **Phaser** algorithm.

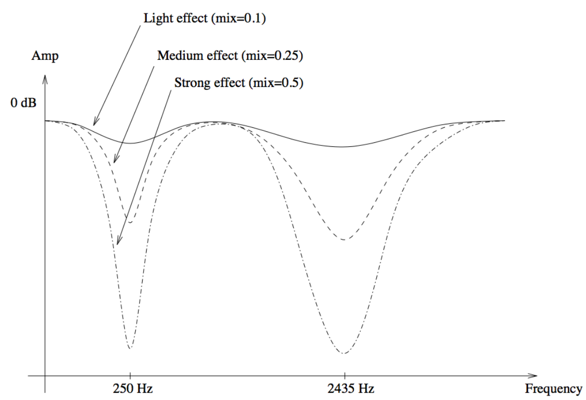


Figure 34: Frequency response of a **Phaser** module. Effect of the mix between wet and dry signal on the frequency response.

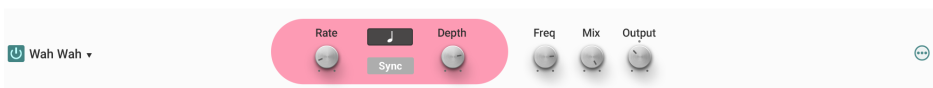
Tuning

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.

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.

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 35. 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.

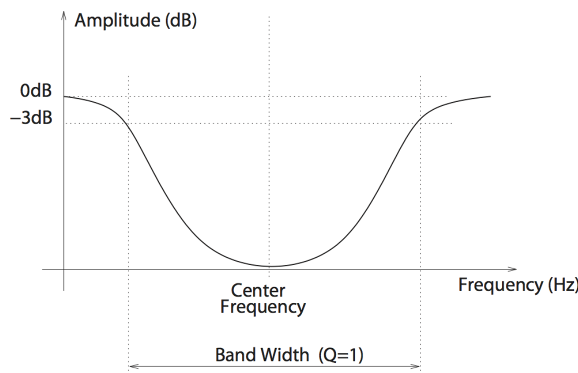


Figure 35: Frequency response of a notch filter.

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.

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.

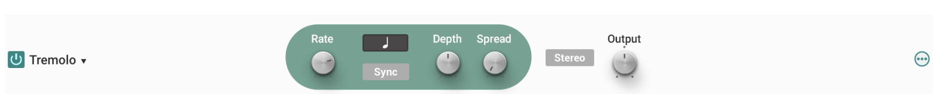
The *Low* and *High* parameters are used to boost or cut low and high frequencies respectively by up to pm 18dB by turning the knob from its center position. These controls have a similar behavior for both channels. Additional control on the frequency response is obtained by using the *Bite* control 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 36 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).

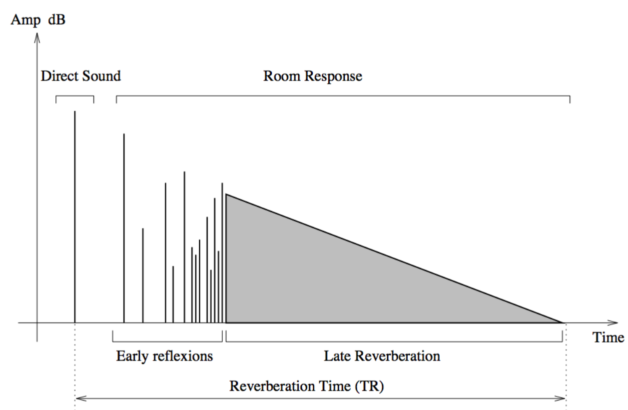


Figure 36: Impulse response of a room.

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.

5.4.13 Studio Mix

The *Studio* knob on the right of the multi-effects 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* 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.

5.4.14 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, around 0 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 +6 dBr zone. It will turn to yellow and then red as the output level increases.

6 The Settings View

6.1 Settings

Clicking on the *Settings* tab in the top of the interface opens the *Settings* view, shown in Figure 37, 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.

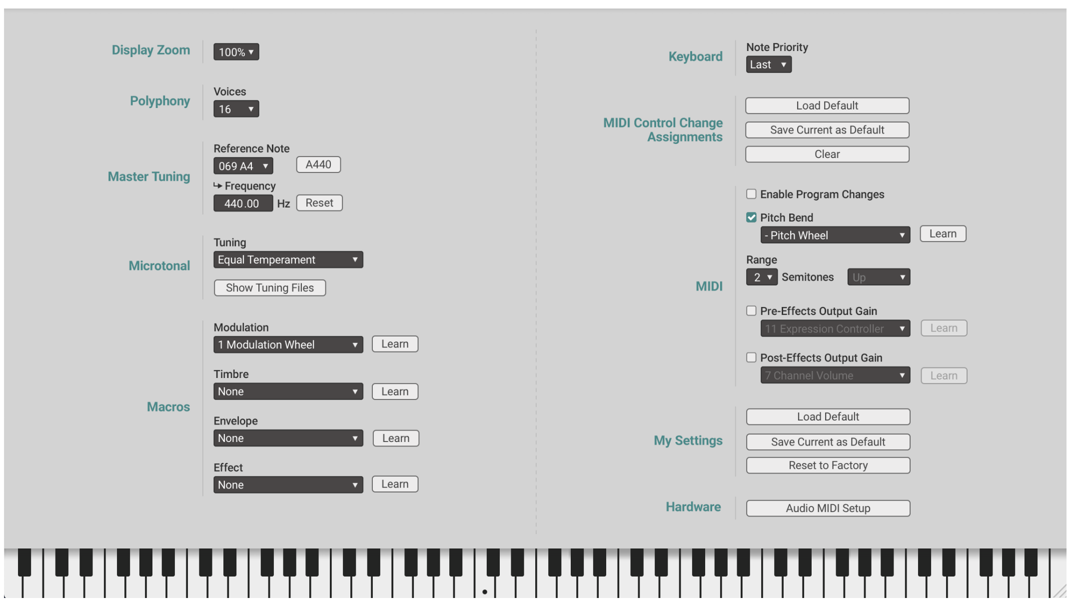


Figure 37: The *Settings* window.

6.1.1 Display Zoom

The size of the interface can be adjusted using the *Display Zoom* drop-down menu. Different size ratio options are presented from small to very large. Note that the size of the interface can also be adjusted by click-dragging the lower right corner of the interface.

6.1.2 Polyphony

The *Voices* control located at the top of the *Settings* window allows one to adjust the number of polyphony voices used by *Chromaphone 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.

6.1.3 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, *Chromaphone 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.

6.1.4 Microtonal Tuning

An interesting feature of *Chromaphone 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.

6.1.5 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 5.2.4.

6.1.6 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.

6.1.7 MIDI

Control Change Assignments

Any control on the *Chromaphone 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.

MIDI Program Changes

Chromaphone 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 *Chromaphone 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

Chromaphone 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, *Chromaphone 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.

Volume and Expression Controller

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 to 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** 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 Hardware

Clicking on *Audio MIDI Setup* buttons opens the *Setup* dialgo which allows on to select an audio and MIDI device when *Chromaphone* is used in standalone mode.

The dialog first allows one to choose an 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.

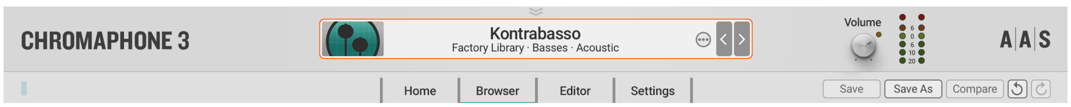
The list of available MIDI inputs appears at the bottom of the dialog. Click on the checkbox corresponding to any of the inputs you wish to use.

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

The utility section is located at the top of the *Chromaphone* interface and it includes important parameters and monitoring tools. For information on the selection of sounds and sound packs, as well as the **Save** and **Save As** commands, please refer to Chapter 4



7.1 The MIDI LED

The MIDI LED is located on the left of the utility section just below the *Chromaphone* logo. 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 *Chromaphone*. 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 Section 8.

7.2 History and Compare

The *History* control allows one to go back through all the modifications that were made to programs since the application was started. In order to travel back and forth in time, use the left and right-pointing arrows respectively. The application will switch between different program states and indicate the time at which they were modified.

The *Compare* button, located above the *Program* display, is used to switch between **Edit** and **Compare** mode. This button is visible only once a modification is applied to a given program. It allows one to revert to the original version of a program in order to compare it with the current version. 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 by clicking on the *Compare* button in order to resume edition.

7.3 Volume

The *Volume* knob is the master volume of the application. It is used to adjust the overall level of the output signal from the synthesizer. General level is increased by turning the knob clockwise.

7.4 Level Meter

The level meter allows one to monitor 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 *Chromaphone*, 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 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 0 dB level mark should typically correspond to playing at mezzo forte (moderately loud) level. 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.

The top LED of the VU-meter (red) is activated when the maximum allowed value of the signal is reached. The signal is then clipped. Before this limit is reached (17 dB), a limiter is triggered in order to ensure a smooth transition before clipping occurs. When the limiter is activated, the yellow LED next to the *Volume* knob is 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 or on the product or company logo. The box is closed by clicking again on the chevrons or outside the box. Useful information is displayed in this box such as the program's version number, the serial number that was used for the authorization and the the email address that was used for registration. 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 *Chromaphone 3*. Audio and MIDI configuration tools are accessed from the *Settings* view which is accessed by clicking on the *Settings* tab in the top 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 accessed by clicking on the *Audio MIDI Setup* button located in the lower right corner of the *Settings* view. 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. *Chromaphone 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 MIDI Setup* button located in the lower right corner of the *Settings* view 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 *Chromaphone 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* section of the *Settings* view which is opened by clicking on the *Settings* tab 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 *Chromaphone 3*.

8.2.3 Creating MIDI Control Assignments

Every control on the *Chromaphone 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 *Chromaphone 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 *Chromaphone 3* to the MIDI controller you just moved.

To deactivate a MIDI assignment, simply right-click/Control-click on the corresponding control on the *Chromaphone 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 *Chromaphone 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* view by clicking on the *Setting* tab located in the top of the interface and use the **Save Current as Default** command in the *MIDI* section under *Control Change 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* view. The *Clear* command is used to disable all MIDI control assignments at once.

8.2.5 MIDI Program Change

Chromaphone 3 responds to MIDI program change messages. In order to enable MIDI program changes, open the the *Settings* view by clicking on the *Settings* tab in the top of the interface and check the **Enable Program Changes** option in the *MIDI* section. When this is the case, *Chromaphone 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 *Chromaphone 3* to respond to MIDI program changes, deselect the **Enable Program Changes** option.

8.2.6 Pitch bend

Chromaphone 3 reacts to the MIDI pitch bend signal received by *Chromaphone 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* view by clicking on the *Settings* tab in the top 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, *Chromaphone 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

Chromaphone 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 5.2.4.

8.2.8 Sustain Pedal

By default, *Chromaphone 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 *Chromaphone 3* as a Plug-In

Chromaphone 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. *Chromaphone 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 *Chromaphone 3*.

9.1 Audio and MIDI Configuration

When *Chromaphone 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

Chromaphone 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 *Home* and *Editor* views.

9.3 Multiple Instances

Multiple instances of *Chromaphone 3* can be launched simultaneously in a host sequencer.

9.4 MIDI Program Change

MIDI program changes are supported in *Chromaphone 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* view 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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