

CHROMAPHONE



USER MANUAL



Applied Acoustics Systems

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1 Introduction

Chromaphone is a synthesizer dedicated to the creation of acoustic instruments. It is based on the combination of acoustic resonators 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 acoustic instruments. *Chromaphone* features a brand-new 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*:

Mac OS

- Mac OS X 10.7 or later
- Intel Core processor or later
- 512 MB of RAM
- 70 MB of free hard drive space
- 1024 x 768 screen resolution
- Built-in audio interface

Windows

- Windows 7 or later 32-bit/64-bit
- Intel Core or equivalent processor

- 512 MB of RAM
- 70 MB of free hard drive space
- 1024 x 768 screen resolution
- Windows-compatible audio interface
- Windows-compatible MIDI interface/keyboard

Keep in mind that the computational power required by *Chromaphone* 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

Simply double-click on the installer file that you have downloaded and follow the instructions of the installer.

1.3 Authorization and Registration

Chromaphone uses a proprietary challenge/response copy protection system which requires authorization of the product. A *challenge code* is a long string of capital letters and numbers that is generated uniquely for each machine during the registration process. The *response code* is another unique string of capital letters and numbers generated from the data encrypted in the challenge code. As the keys are unique to each machine, it is necessary to go through this procedure every time the program is installed on a new computer.

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.

1.3.1 Your Computer is Online

The authorization process is very simple if your music computer is connected to the internet since the *Chromaphone* program will connect to the AAS server and take care of the key exchange automatically.

After starting the application, a message will appear telling you that the application needs to be authorized as shown in Figure 1. Enter your serial number and click on the *Authorize* button. The program will then connect to the AAS server and complete the authorization process.

If this is the first AAS product that you authorize on your computer, or if no registration information can be related to your serial number by our server, you will be asked to provide your name

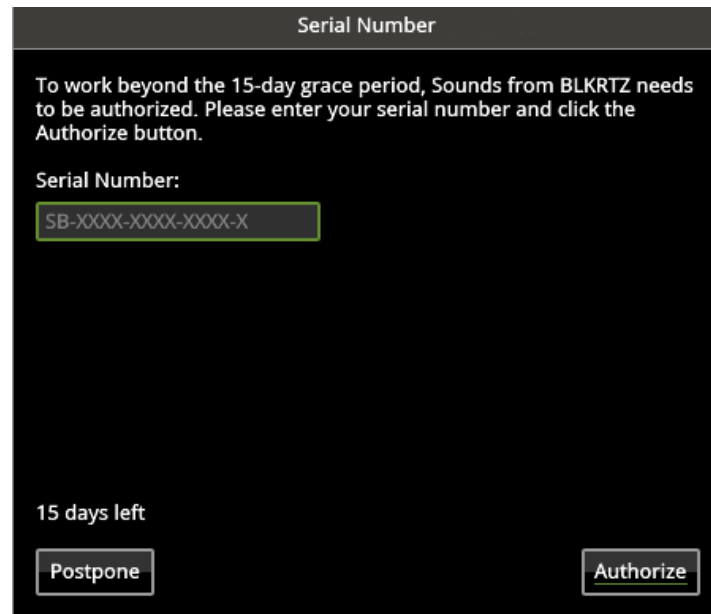


Figure 1: Online Authorization.

and email address for registration purposes. Note that only a valid email address is required to register your product. Registration of your product will entitle you to receive support and download updates when available, as well as take advantage of special upgrade prices offered from time to time to registered AAS users.

1.3.2 Your Computer is Offline

If your music computer is not connected to the internet you will need to obtain the *response code* from an internet connected computer or by contacting AAS.

After starting the application, a message will appear telling you that the application needs to be authorized. After clicking on the *Authorize* button, a pop-up window will appear as shown in Figure 1. Enter your serial number and click on the *Authorize* button. The program will then inform you that your computer is not connected to the internet, click on the *Offline Options* button and a new pop-up window will appear as shown in Figure 2.

Your serial number as well as the automatically generated *challenge code* are displayed but you need to obtain the *response code*. To do so, take note of your *serial number* and *challenge code* and proceed to an internet connected computer. Launch your browser and go to the unlock page of the AAS website located at:

www.applied-acoustics.com/unlock/

Enter your serial number and *challenge code* in the form, follow the instructions, and the *response code* will appear on screen. Write it down, go back to your music computer, and enter the

Offline Options

Forward us your serial number and challenge code via email, visit the Unlock page from another computer connected to the internet, or give us a call to get your response code. Once you have your response code enter it below.

support@applied-acoustics.com
www.applied-acoustics.com/unlock/
Tel. 1 514 871-8100 (Worldwide)
Tel. 1 888 441-8277 (North America)

Serial Number:
CL-3991-01EA-2323-5

Challenge Code:
1109-86E2-0122

Response Code:
XXXX-XXXX-XXXX-XXXX-XXXX

Postpone Go Back Authorize

Figure 2: Offline Authorization.

response code in the authorization pop-up window. This will complete the authorization procedure.

If you prefer, you can also contact us by email at support@applied-acoustics.com with your *serial number* and *challenge code* and we will send you back your *response code*.

Should you not have access to the internet, AAS support representatives are available to assist you in the unlock and registration process Monday to Friday, 9am to 6pm EST. You may contact us by phone at:

- North America Toll-free number: 1-888-441-8277
- Outside North America: 1-514-871-8100

1.4 Getting Started

1.4.1 Using *Chromaphone* in Standalone Mode

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

- **Windows** - Double-click on the *Chromaphone* icon located on your desktop or select *Chromaphone* from the **Start > All Programs >** menu.
- **Mac OS** - Double-click on the *Chromaphone* 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 Setup** button located in the lower left corner of the *Chromaphone* interface. The **Audio Setup** dialog first allows you to select an audio output device from those available on your computer. Multi-channel interfaces will have their outputs listed as stereo pairs.

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

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

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.

1.4.2 Exploring the Factory Sounds

Chromaphone comes with a wide range of factory programs right out of the box 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* offers is simply to go through the programs one at a time.

Chromaphone uses the notions of *Banks* and *Programs* to organize and classify sounds. A program or preset is a stored set of parameters corresponding to a given sound. The programs are grouped and organized in banks.

The name of the currently loaded bank and program are displayed at the top of the interface. One navigates among the different banks and programs by using the arrows in each of the corresponding boxes or by opening the associated drop-down menu by clicking inside these boxes. Banks and programs are managed using the *Bank Manager* which is revealed by clicking on the *Manage* button appearing above the right-top corner of the *Bank* box. Playing programs and organizing them is pretty straightforward, please refer to Chapter 3 for a complete description of the bank and program management operations.

1.4.3 Using *Chromaphone* as a Plug-in

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

1.5 Getting Help

AAS technical support representatives are on hand from Monday to Friday, 9am to 6pm EST. Whether you have a question on *Chromaphone*, 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. The support pages are located at:

1.6 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*

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 General Organization and Signal Flow

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 3 and 4.

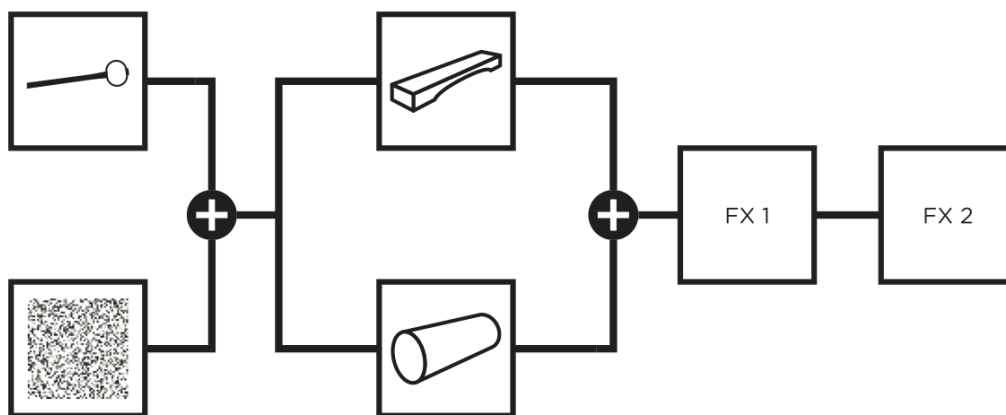


Figure 3: Signal flow of *Chromaphone*. Resonators in parallel mode.

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.

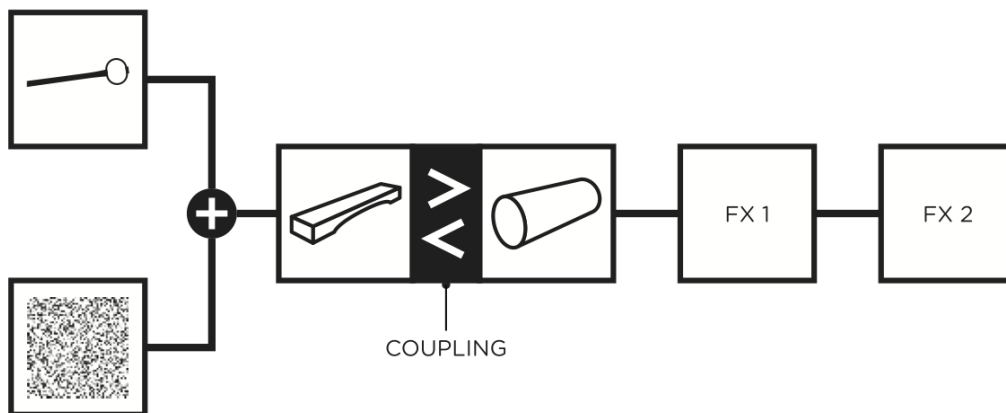


Figure 4: Signal flow of *Chromaphone*. Resonators in coupling mode.

2.2 Interface

The graphical user interface has been organized around three different views as shown in Figures 5, 6 and 7.

The first view, called the *Play* view of the instrument, gives access to different performance parameters as well as to a step sequencer. The second and third views, called the *Edit* and *FX* views respectively, are used for in-depth editing of the synthesis and effect parameters.

One can switch from one view to the other by using the *Play*, *Edit* and *FX* buttons located in the *Utility* section at the top of the interface. This section of the interface is common to all the views and includes the bank manager, used to access and manage sounds, as well as general settings and indicators. These tools are described in details in Chapter 3 and Chapter 5 respectively.

2.2.1 The Play View

The lower section of this view includes a master clock, unisson, vibrato and arpeggiator modules which will be described in more details in Chapter 4. This part also includes a resonator display giving information on the type of resonators used in the instrument currently being played and its configuration.

On the left of these parameters, one finds a pitch bend wheel and a modulation wheel. The modulation wheel is normally used to control the amount of vibrato in the sound but it can also be used to adjust any other parameter through MIDI links which will be described in Chapter 6. Just below is a clickable eight octave ribbon allowing one to play different notes on the range of the piano which can be useful when no MIDI keyboard is connected to the computer.

The middle section of this view allows one to turn the effects from the multi-effects module, compression and equalizer *on* and *off* and to rapidly adjust their main parameters.



Figure 5: The Play view.

2.2.2 The Edit View

The *Edit* view gives access to the synthesis parameters described in details in Chapter 4 and allows one to really go under the hood. In this view, one can choose the sound source, the type of resonators used and how they are configured. All module parameters and modulations are adjusted in this view.

2.2.3 The FX view

The *FX* view includes an equalizer, a compressor a multi-effects, and a reverb module. The multi-effects module consists in two effects in series. The effect list includes a delay, distortion, chorus, flanger, phaser, wah wah, auto wah, tremolo, and a notch filter. The functioning of the effect modules is described in details in Chapter 4.



Figure 6: The Edit view.



Figure 7: The FX view.

3 Bank and Program Management

Chromaphone comes with several factory presets, called *programs*, covering a wide range of sounds. This collection of programs lets you play and familiarize yourself with this synthesizer without having to tweak a single knob. Soon, however, you will be experimenting and creating your own sounds and projects that you will need to archive or exchange with other users. In this section, we review the management of programs.

3.1 Banks and Programs

Sounds are stored in banks containing so-called *programs*. The name of the currently selected bank is shown in the *Bank* drop-down display located at the top of the *Chromaphone* interface. The list of available banks is viewed by clicking on the *Bank* display. A bank can be selected by navigating in the list of banks using the left and right-pointing arrows in the display or by clicking on its name when the list of banks is open. Clicking on the bank display brings focus on this section of the interface, the display is then outlined by an orange line, and one can then navigate through the list of banks using the up, down, left, or right arrows of the computer keyboard.

The list of programs included in the currently selected bank can be viewed by clicking on the *Program* display located below the *Bank* display. A program is selected by using the left and right-pointing arrows or by clicking directly on its name. Once a program is selected, the value of the different parameters of the synthesizer are updated and it can then be played. As for the bank list, one can navigate through the program list using the computer arrows after clicking on the *Program* display.

3.2 Saving Programs

Programs are saved by clicking on the **Save** button located on the top of the *Program* display. When a program has just been loaded, this command is greyed and therefore inactive. It is activated as soon as a parameter of the interface is modified. Clicking on this command replaces the stored version of the program with the new one.

The **Save As** command is activated by clicking on the corresponding button which opens the **Save Program** pop-up window. It is then possible to save the program under a new name or its current one in any of the available program banks. Note that if the original name of the program is used, a new program with the same name will be created at the end of the program list meaning that the original program is not erased. This also implies that it is possible to have many programs with the same name in the same bank.

3.3 The Bank Manager

Banks and Programs can be edited using the **Bank Manager**. The manager window is displayed by clicking on the *Manager* button located above the *Bank* display. It is closed by clicking again

on the same button. On the left of the window, one finds the list of banks. Clicking on a bank name fills the list of programs located in the center of the window with the name of these included in the selected bank.

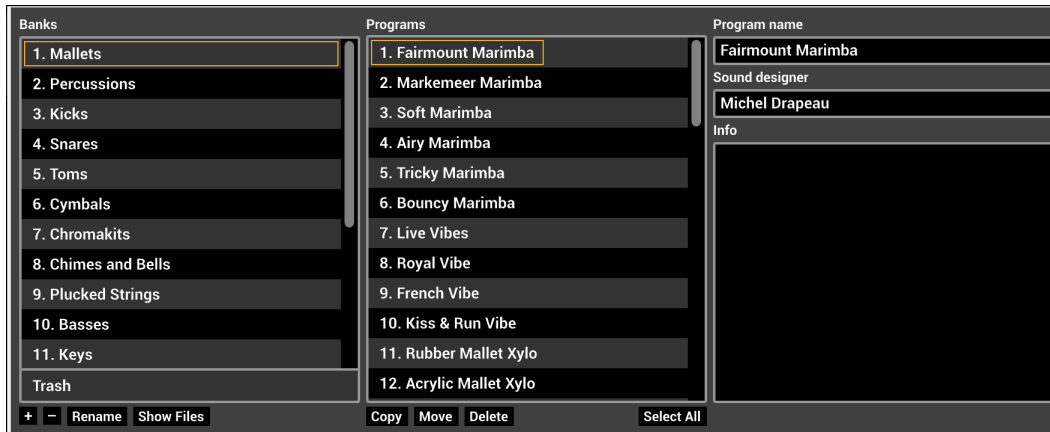


Figure 8: Bank and program manager window.

A new bank can be created by clicking on the + button below the bank list. This opens the **Create New Bank** window in which the name of the new bank can be entered. A bank can be deleted by first selecting it in the bank list and then clicking on the - button. Be careful, this command erases a bank and all the programs it contains; this operation is permanent and can not be undone. In order to rename a bank, simply click on the *Rename* button and enter a new name.

Banks and the information corresponding to each of its programs is stored in a simple text file on your computer hard disk. In order to view these bank files, click on the *Show Files* button under the bank list. On Windows, this command will open an Explorer window at the location where the files are stored. On Mac OSX, the command has a similar effect and opens a Finder window. All the bank file names follow the same format and begin with the bank name. These files can be used for backups or to exchange presets with other users.

The list of programs included in the selected bank is displayed in the program list in the center of the manager window. Presets are selected by clicking on their name which updates the program information appearing on the right of the preset list. Program information includes the name of the preset, its author and comments. This information can be updated by clicking on the corresponding box which opens an edition window. Note that multiple presets can be updated simultaneously by selecting more than one preset at once and clicking on a preset information box.

A multiple selection consisting of adjacent programs is obtained by holding down the *Shift* key on the computer keyboard and then clicking on the name of the first program to be copied and then 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 programs to be copied. It is also possible to select all programs at once by clicking on the *Select All* button at the bottom of the program list.

Programs can be copied to another bank by clicking on the *Copy* button. A program must first

be selected by clicking on its name on the program list; it is then copied by moving the mouse to a given bank in the *Bank* list on the right and clicking on the bank name. The **Move** command is activated by clicking on the *Move* button; it copies a preset to a new bank but also erases it in the original bank. A multiple selection of programs can be used with the **Copy** and **Move** commands

Programs can be deleted from a bank by first selecting them and then clicking on the *Delete* button. This will move the programs to a special bank called *Trash* which is located below the regular list of banks. This means that deleted programs can always be recuperated as long as they are not deleted from the *Trash* bank. The content of the *Trash* bank is viewed by clicking on its name; the different programs can then be moved to the other banks as explained above. The *Trash* bank can be emptied by clicking on the *Empty Trash* button which appears below the program list when the *Trash* bank is selected. Be careful as this command can not be undone.

3.4 Using MIDI Bank and Program Changes

Banks and programs can be changed using MIDI bank and program change commands. For more information on how to use these commands, please refer to sections 6.2.4 and 6.2.5.

3.5 Backups of Banks and Programs

User banks are stored on disk as simple text files located in the following folders:

On **Mac OS**:

/Users/[user name]/Library/Application Support/Applied Acoustics Systems/*Chromaphone 2*/Banks

On **Windows**:

%AppData%\Applied Acoustics Systems*Chromaphone 2*\Banks

The bank files saved by *Chromaphone* are named using the following convention:

[name of bank].*Chromaphone 2* Bank

These files contain all the information corresponding to the programs they include. These files can be displayed directly from *Chromaphone* by opening the *Bank* manager and clicking on the *Show Files* button. This will open an Explorer or Finder window on Windows or Mac OS respectively at the right location.

The simplest way to create a backup of banks and programs is to make a copy on an external media of the above mentioned folders. Individual banks can be backed-up by making copies of individual bank files.

3.6 Exchanging Banks and Programs

Banks and programs can easily be shared with other *Chromaphone* users. This operation simply involves the exchange of the above mentioned user bank files. When a new bank file is copied to the bank folder, it is automatically available to *Chromaphone*.

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

3.7 Restoring the Factory Library

If necessary, it is possible to restore the original factory library of banks and programs. The original factory bank files are located in the following folders:

On **Windows 64-bit**:

C:\Program Files (x86)\Applied Acoustics Systems*Chromaphone 2*\Factory Library

On **Windows 32-bit**:

C:\Program Files\Applied Acoustics Systems*Chromaphone 2*\Factory Library

On **Mac OS** startup disk:

/Library/Application Support/Applied Acoustics Systems/*Chromaphone 2*/Factory Library

Restoring the factory library simply involves copying the files contained in these folders and pasting them in the user bank folders listed in Section 3.5. The user bank folders can be opened directly in an Explorer or Finder window, on Windows and Mac OS respectively, or by using the *Show Files* command directly from the *Chromaphone* bank manager.

Note that if you have bank files with the original factory bank names in your user bank folder, they will be replaced by the original factory files. This means that you will lose programs that you would have modified or created in these banks. This operation must therefore be done with caution and it is recommended that you make copies or rename your user banks before proceeding with the restore.

3.8 Importing Programs from Chromaphone 1

Chromaphone 2 includes a converter allowing one to import programs from version 1 to version 2. The conversion itself is automatic but first involves to copy program bank files from the folder where version 1 banks are stored to the folder where version 2 banks are stored.

Banks are stored in the folders mentioned in section 3.5. The simplest way to access them, consists in using the *Show Files* button in the bank manager of each product version which will open a Finder or Explorer window on Mac OS X or Windows respectively at the right location. Bank files that are to be converted then simply need to be copied from the version 1 bank folder to that corresponding to version 2.

4 Parameters

This section can be used as a reference for the different controls appearing on the *Chromaphone* graphical interface. 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.

4.1 General Functioning of the Interface

4.1.1 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. Precise values can also be entered manually by clicking on the parameter label and typing the value on the computer keyboard.

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

4.1.2 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.

4.1.3 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.

4.1.4 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, velocity, and Modulation Wheel signals (Key, Vel and MW 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 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.

4.1.5 Synchronisation

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

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

4.2 General Notions of Acoustics

4.2.1 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 9 and 10. 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 10 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.

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

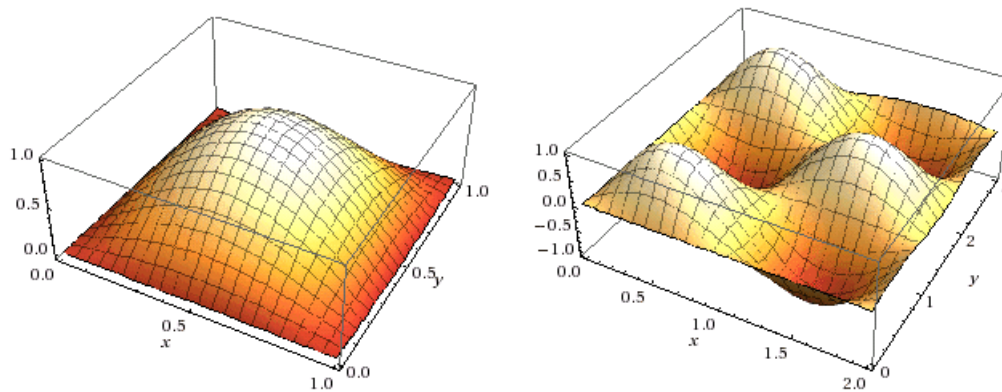


Figure 9: Motion corresponding to normal mode $[1,1]$ and $[3,2]$ of a plate.

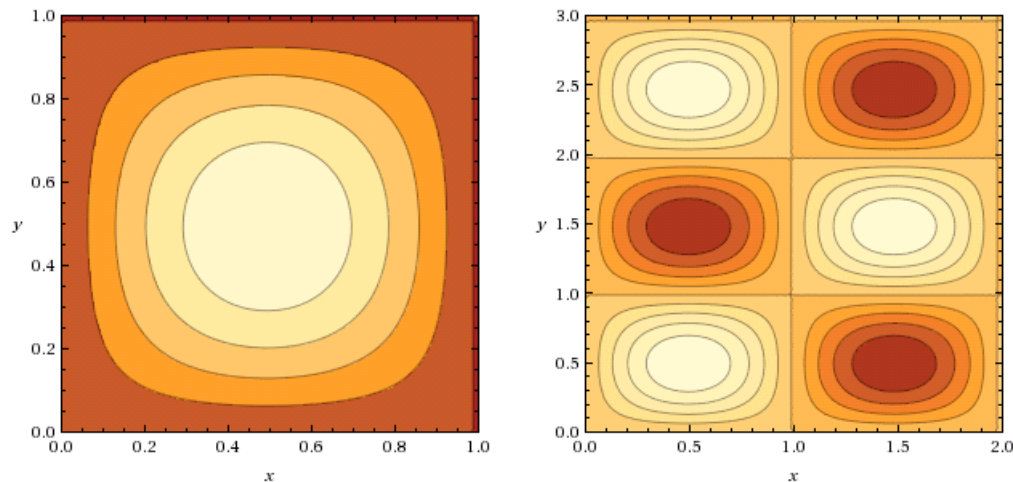


Figure 10: Contour plot corresponding to normal mode $[1,1]$ and $[3,2]$ of a plate.

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.

4.2.2 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.

4.3 The Edit View

4.4 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.



4.5 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 11 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.

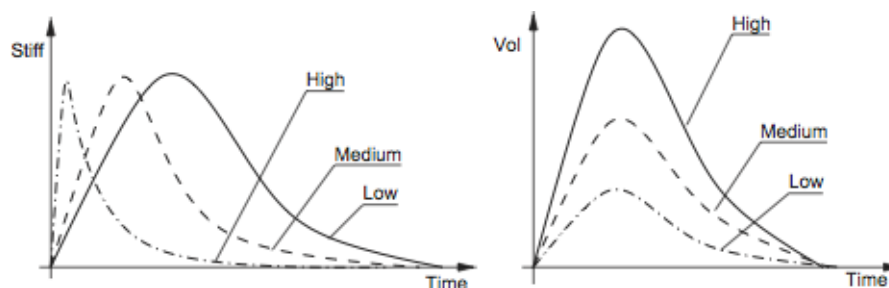


Figure 11: Effect of the *Stiffness* and amplitude of the force impact (*Mallet* knob from the **Mixer** module) on the output from the **Mallet** module.

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.

4.6 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.

4.7 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.



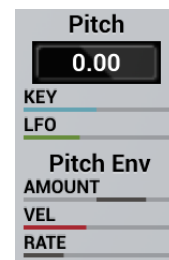
The *Resonator* selector allows one to choose the type of resonator used. The resonator type can be changed by clicking on the resonator icons or by using the drop-down menu at the top of the icon display. The list of resonators include the main type of objects used in the making of musical instruments. 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 *Quality* control is located just below the resonator selector and is represented by big dots. 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 corresponding to 4, 16, 30 and 70 modes. 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. The *Key* modulation parameter associated with this control allows one to adjust this parameter as a function of the note played on the keyboard. 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 times 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 low-pass filter with the cut-off frequency of the filter increasing as the radius 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 as 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 4.2.1. 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 at 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.

4.8 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 12. 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 13. 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

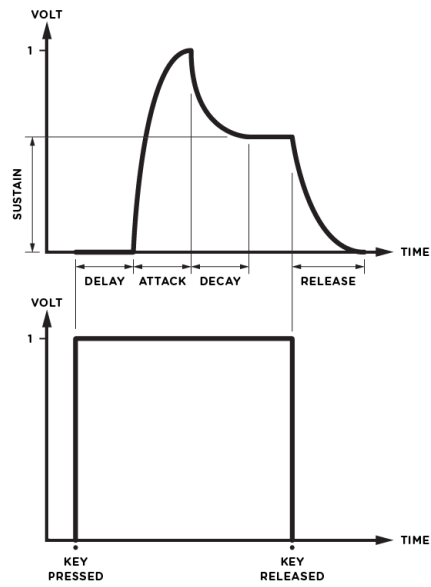


Figure 12: ADSR Response curve.

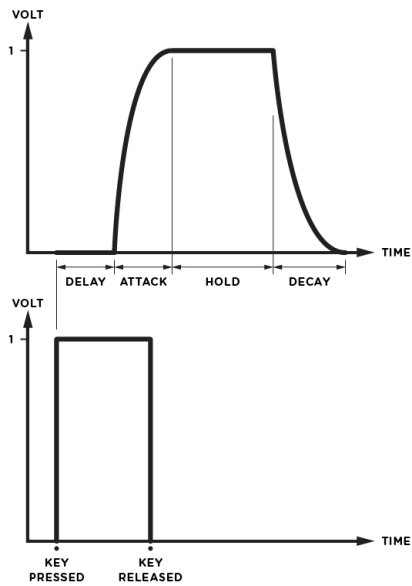


Figure 13: AHD envelope response curve.

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.

4.9 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 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 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.

4.10 The FX View

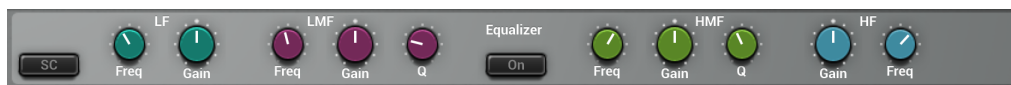
The **FX** view is displayed by clicking on the *FX* button in the utility section at the top of the interface and is based around a **Multi-effects** module.

The **Multi-Effects** module allows one to process and shape the signal from the piano before sending it to the output. This module comprises an **EQ** and a **Compressor** in series with two configurable effect processors and a **Reverb**. The configuration of the **EQ** and the **Compressor** module depends on the position of the *SC* and *Pre* buttons of these modules as will be explained below. The two effect processors can be set to a different type by using the drop-down menu located at the center of each module for a wide range of possibilities. The effect list includes a **Delay**, **Distortion**, **Chorus**, **Flanger**, **Phaser**, **Wah Wah**, **Auto Wah** and a **Notch** filter.

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

4.10.1 EQ

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



The functioning of the low shelf filter is depicted in Figure 14. 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 ± 15 dB range. In its center position there is no attenuation (0 dB). Turning it clockwise boosts the amplitude of low frequencies while turning it anti-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 14. 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 ± 15 dB 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.

The **EQ** module features two peak filters, labeled **LMF** and **HMF**, allowing to shape the signal in two frequency bands as illustrated in Figure 15. The filters apply a gain factor to frequency components in a band located around the cutoff frequency of the filters. This cutoff frequency is

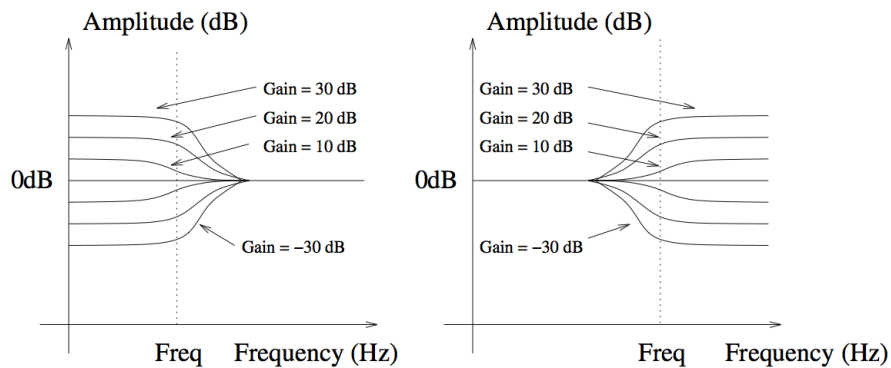


Figure 14: Low and high shelf filters.

adjusted using the *Freq* knob and can vary between 100 Hz and 10 kHz. The gain factor applied at 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.

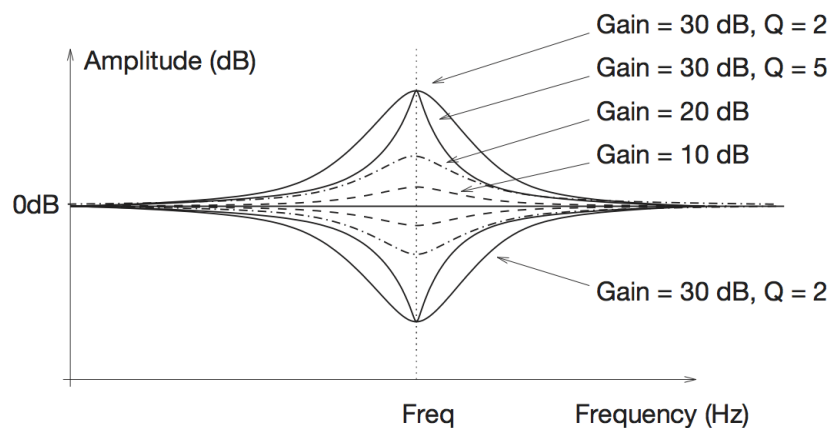
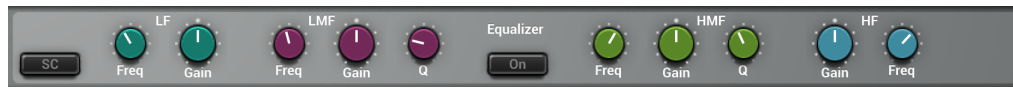


Figure 15: Peak filter.

The *SC* button (side-chain) is used to determine if the output from the **EQ** module is to be used as the control signal of the **Compressor** module as described in Section 4.10.2. Finally, note that all the gain knobs from this module can be accessed directly from the **Play** view.



4.10.2 Compressor

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

Tuning

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

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

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

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

The location of the **Compressor** in the signal path depends on the setting of the *Pre* button. When this knob is *on*, the **Compressor** is located at the output from the piano, just before the **EQ** module. In this position, the input signal of the **Compressor** and its control signal are both the output signal from the piano. When the **Pre** button is *off*, the **Compressor** is located after the **EQ** module. In this configuration, the control signal of the **Compressor** is then the output signal from the **EQ** module. The input signal to the compressor is determined by the position of the **SC** button. When it is *on*, the **Compressor** is in a *side-chain* configuration. The input of the **Compressor** is

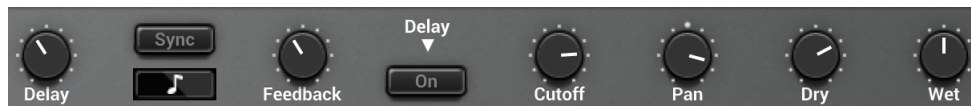
then the output signal from the piano. When it is *off*, the input of the **Compressor** is the output signal from the **EQ** module.

Using the compressor in side chain configuration is useful when one wants to trigger the compressor using other criteria than the general level of the signal to be compressed. For example, a sound with a lot of bass would easily trigger the **Compressor** when playing low notes. In order to avoid that, the **EQ** module would be set to filter out low frequency components. This signal would then be used to control the **Compressor** while the input signal to the **Compressor** would still include these low frequency components.

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.

4.10.3 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 less extreme behavior can be obtained by choosing an intermediate position. In particular when the *Pan* knob is in its center position, an equal amount of signal is sent in both channels.

The output signal from the **Delay** module can include a mix of input signal (dry) and delayed signal (wet). The *Wet* and *Dry* knobs are used to adjust the amplitude of each component in the

final output. The amplitude of each component is increased by turning the corresponding knob clockwise from no signal to an amplitude of +6dB. Note that the *Wet* parameter is also adjustable from the **Play** view.

4.10.4 Distortion

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



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

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

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

4.10.5 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 16. 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.

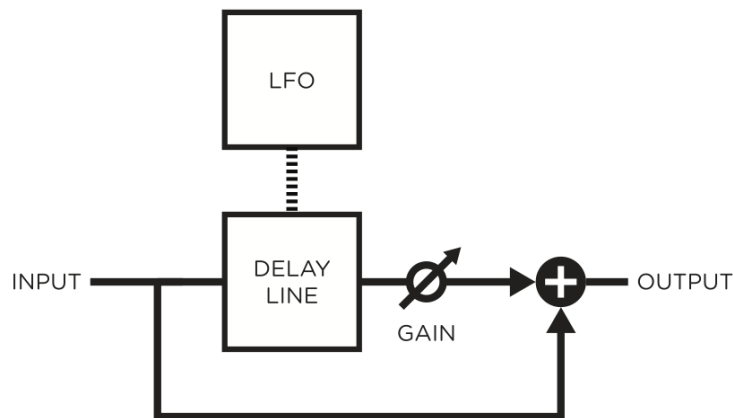
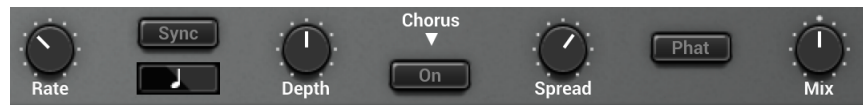


Figure 16: **Chorus** module.

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 hears the dry input signals. In its rightmost position, one only hears 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.

4.10.6 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 17. 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.

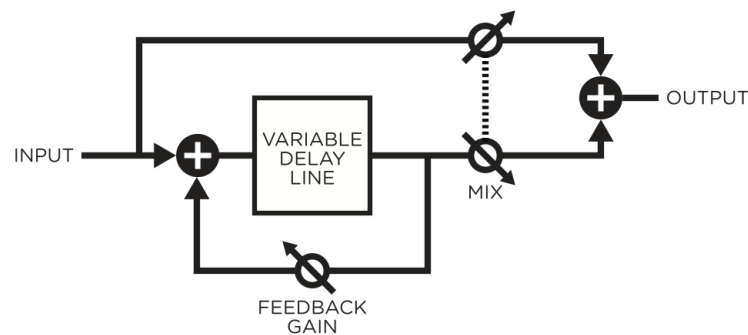


Figure 17: **Flanger** algorithm.

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 18. The location of the fundamental frequency f_0 and the spacing between the valleys and peaks of the frequency response is determined by the length of the delay line ($f_0 = 1/(2\text{delay})$), the longer the delay, the lower is f_0 and the smaller the spacing between the harmonics while decreasing the delay increases f_0 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 19. As the amount of wet signal sent to the output is increased, the amount of rejection increases. Finally, the shape of the frequency response of the **Flanger** module is also influenced by the amount of wet signal re-injected into the feedback loop as shown in Figure 20. 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.

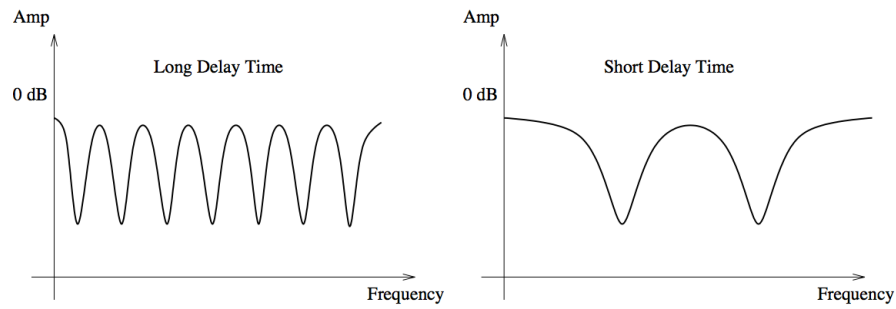


Figure 18: Frequency response of a **Flanger** module. Effect of the length of the delay line.

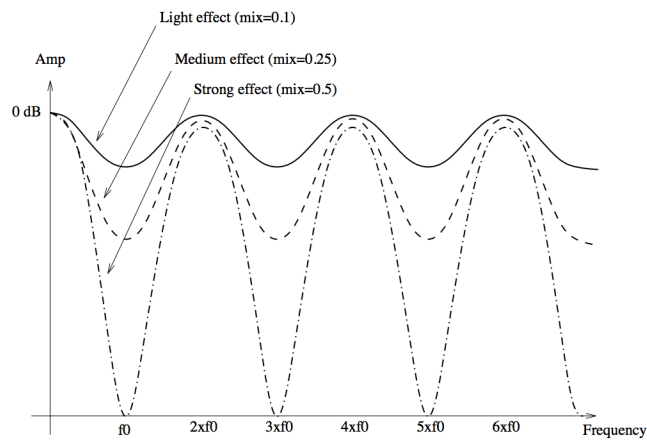


Figure 19: Effect of the mix between wet and dry signal on the frequency response of a **Flanger** module

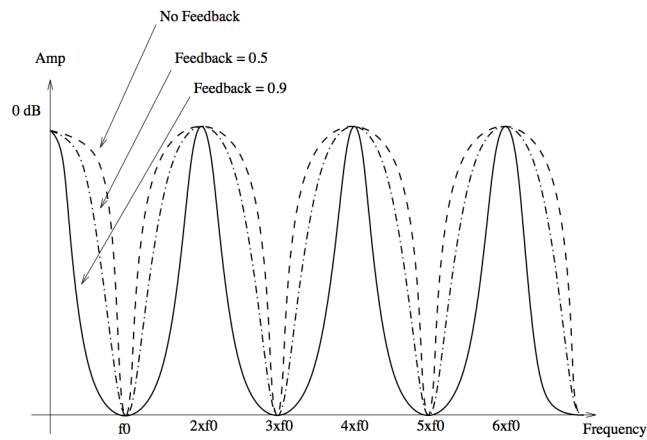


Figure 20: Effect of the amount of feedback on the frequency response of a **Flanger** module.

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 sent to the output. Note that the *Depth* parameter is also adjustable from the **Play** view.

4.10.7 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 21. 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 22. 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 22. 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.

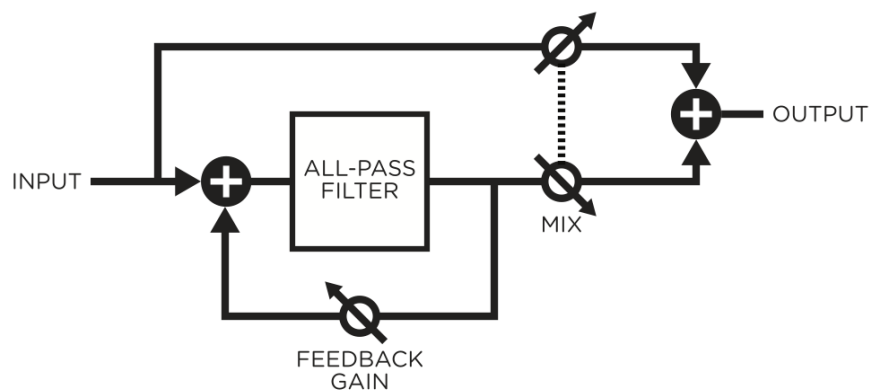


Figure 21: **Phaser** algorithm.

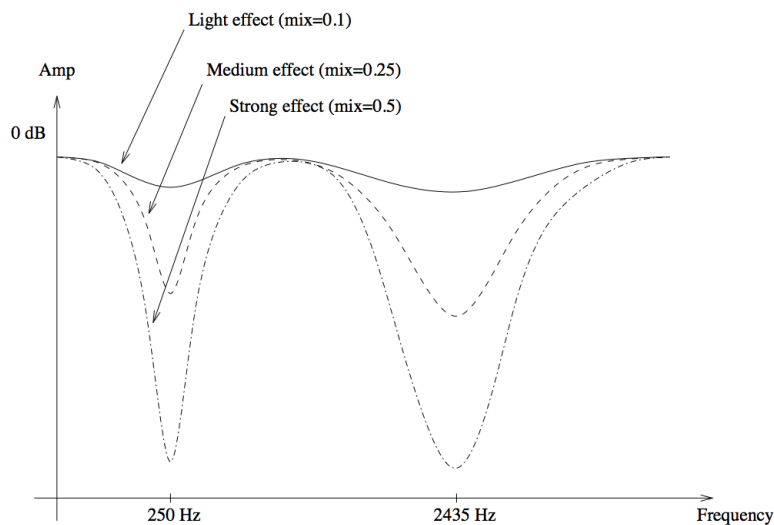


Figure 22: Frequency response of a **Phaser** module. Effect of the mix between wet and dry signal on the frequency response.

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.

4.10.8 Wah

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



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

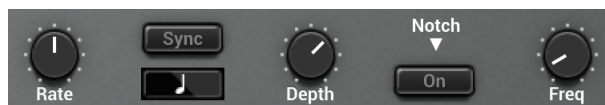


The *Depth* knob controls the excursion of the center frequency of the filter. In the case of the *Wah Wah* effect, this excursion is applied around the value fixed by the *Freq* knob while in *Auto Wah* effect the value of the center frequency increases from the value fixed by the *Freq* knob. Turning this knob clockwise increases the excursion of the center frequency. Note that the *Depth* parameter is also adjustable from the **Play** view.

Finally, 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.

4.10.9 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 23. As was the case for the *Wah Wah* effect, the filter can be modulated.



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

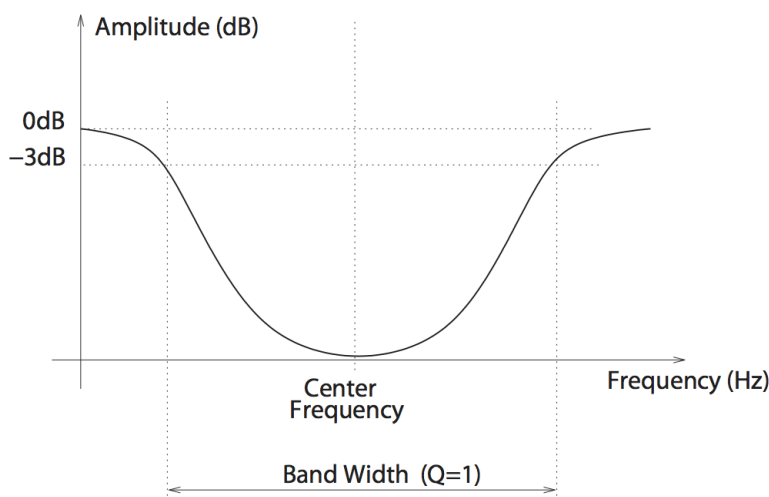


Figure 23: Frequency response of a notch filter.

4.10.10 Reverb

The *Reverb* effect 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.

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 24 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).

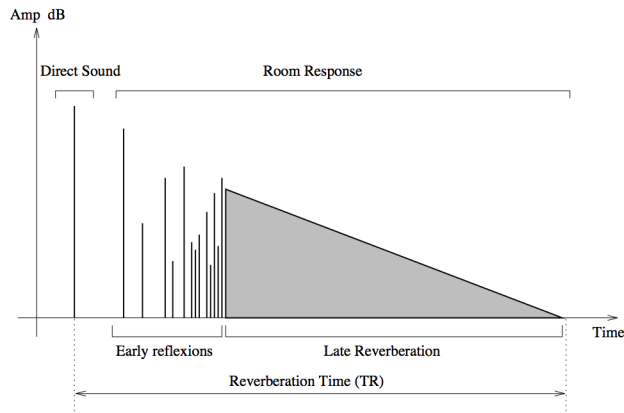


Figure 24: Impulse response of a room.

Adjusting the room effect

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

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

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



between the sound source and the listener. While the room response is quite constant regardless of

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

4.11 The Play View

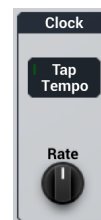
The **Play** view is where the main performance oriented modules are located. Key parameters from the **Edit** and **FX** view are also included for quick access. This view is loaded when starting the instrument and can be accessed from another view by clicking on the *Play* button on the top part of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

The middle section of this view allows one to switch *on* and *off* the **EQ**, **Compressor** and **Reverb** as well as the active effect modules. Key effect parameters are also adjustable as presented in the description of the different effect modules in section 4.10



4.11.1 The Clock Module

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

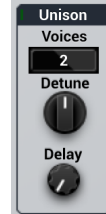


When using *Chromaphone* in plugin mode, the *Tap Tempo* pad is replaced by a *Sync To Host* switch. In its *on* position, the rate is synchronized with that of the host sequencer. When switched *off*, the tempo is determined by the value of the *Rate* knob.

4.11.2 Unison

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.

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.



4.11.3 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*, a dedicated module is provided for this effect. The vibrato module is hard wired and affects the pitch of both oscillators.

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 *MW* gain knob is used to determine the effect of the keyboard modulation wheel on the depth of the vibrato. When this knob is fully turned to the left, the modulation wheel has no effect but as it is turned clockwise the depth of the vibrato increases when the modulation wheel is used. The increase is always relative to the position of the *Amount* knob and becomes greater as the *MW* knob is turned clockwise.



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.

4.11.4 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 module 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 (red button *on*). Notes are played regularly when all the steps of the display are turned *on* and rhythmic patterns are created by selecting only certain steps. The arrow button below each step is used to fix looping points from which the rhythmic pattern starts being played again from the beginning.



Rate and Synchronization

The rate at which the arpeggiator pattern is scanned is set by the *Rate* knob of the **Arpeggiator** module or can be synced to the master clock of the *Clock* module. The *Rate* knob is only effective when the *Sync* control is set to *off*. When the *Sync* control is *on*, the rate (tempo) is fixed by the master **Clock** module (see 4.11.1) 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.

4.11.5 Pitch Wheel

The MIDI pitch wheel allows one to vary the pitch of the note played. The pitch wheel can be moved with the mouse but it is also automatically connected to the pitch wheel signal received from your MIDI keyboard.

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 MIDI configuration window by clicking on the **MIDI** button located just below the MIDI led in the top part of the interface and use the **Pitch Bend Range** drop-down menu to select the range in semi-tones.

4.11.6 Modulation Wheel

The modulation wheel is linked to the *Amount* parameter of the **Vibrato** module. It can be activated on screen or from the modulation wheel of your MIDI controller (MIDI controller number 1). The *MW* gain knob of the **Vibrato** module is used to control the sensitivity of the vibrato amplitude to the modulation wheel. Note that other parameters can be linked to the modulation wheel using MIDI links as explained in Section 6.

4.11.7 Ribbon

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



5 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 *Banks* and *Programs* please refer to Chapter 3



5.1 The MIDI LED

The MIDI LED is located on the left of the level-meter. 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 6.

5.2 Polyphony

The *Voices* control located in the upper left corner of this section allows one to adjust the number of polyphony voices used by *Chromaphone*. 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.

5.3 Tuning

The *Tune* control, located to the right of the MIDI LED, 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.

An interesting feature of *Chromaphone* is that it can be tuned using different temperaments using Scala micro-tuning files. Temperament files are loaded by clicking on the *Tune* button which opens the Tuning pop-up window and displays the list of available tuning temperament files available.

By default, *Chromaphone* is set to equal temperament. Other files can be added to the list by copying them to the following folders:

On **Mac OS**:

/Users/[user name]/Library/Application Support/Applied Acoustics Systems/Scala Tunings/

On **Windows**:

%AppData%\Applied Acoustics Systems\Scala Tunings\

These folders can be displayed directly from *Chromaphone* by clicking on the *Show Tuning Files* button at the bottom of the *Tuning* pop-up window.

Selecting a Scala file in the list automatically triggers the loading of the corresponding temperament. The reference note that will be used as the base note for the scale described in the Scala file can be set using the Reference Note control appearing at the bottom of the *Tuning* window. The frequency of this reference note is calculated relative to the settings of the *Tune* control. Please note that the reference note does not appear in the window when the default temperament is chosen, it only appears once a Scala file is loaded.

5.4 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.

5.5 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.

5.6 Level Meter

The level meter allows one to monitor peak and RMS (root means square) level of the left (L) and right (R) output channels from the synthesizer. As a limiter is located at the output of *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.

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

5.7 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 email address that was used for registration. The box also includes a link to the pdf version of this manual.

6 Audio and MIDI Settings

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

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

6.1 Audio Configuration

6.1.1 Selecting an Audio Device

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

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

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

6.1.2 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* 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.

6.2 MIDI Configuration

6.2.1 Selecting a MIDI Device

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

6.2.2 Creating MIDI Links

Every control on the *Chromaphone* interface can be manipulated by an external MIDI controller through MIDI control change assignments. In most cases this is much more convenient than using the mouse, especially if you want to move many controllers at once. For example, you can map the motion of a knob on the interface to a real knob on a knob box or to the modulation wheel from your keyboard. As you use the specified MIDI controllers, you will see the controls move on the *Chromaphone* interface just as if you had used the mouse.

In order to assign a MIDI link to a controller:

- On the *Chromaphone* interface, right-click/Control-click on a control (knob, button) and select the **MIDI Learn** 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* to the MIDI controller you just moved.

To deactivate a MIDI link, simply right-click/Control-click on the corresponding control on the *Chromaphone* interface and select the **MIDI Forget** command.

6.2.3 Creating a default MIDI Map

It is possible to define a set of MIDI links, called a MIDI map, that will be loaded automatically when *Chromaphone* is launched. Once you have defined a set of MIDI links that you wish to save, click on *MIDI* button to open the *MIDI* configuration window and click on the **Save Current as Default** button.

If you make changes to MIDI links after opening the program and wish to revert to the default MIDI map click on *MIDI* button to open the *MIDI* configuration window and click on the **Load Default** button.

If you wish to deactivate all the MIDI links at once open the *MIDI* configuration window and click on the **Clear MIDI Map** button.

6.2.4 MIDI Program Changes

Chromaphone responds to MIDI program changes. When a program change is received, the current program is changed to the program having the same number as that of the program change message in the currently loaded bank.

If you do not wish *Chromaphone* to respond to MIDI program changes, open the *MIDI* configuration window by clicking on the *MIDI* button and uncheck the **Enable Program Changes** option.

6.2.5 MIDI Bank Changes

In general, MIDI bank numbers are coded using two signals: the MSB (most significant byte) and LSB (least significant byte) transmitted using MIDI CC (continuous controller) number 0 and 32 respectively. The way these signals are used differs with different manufacturers.

In the case of *Chromaphone*, the value of the MSB signal is expected to be zero while the value of the LSB signal represents the bank number. Banks are therefore numbered from 0 to 127 with this number corresponding to the position of a bank within the list of banks as displayed by the Bank manager (see Section 3.3). For example, an LSB value of 0 corresponds to the first bank in the bank list while an LSB value of 10 corresponds to the eleventh bank in the list. Note that a bank change only becomes effective after the reception of a new MIDI program change signal.

If you do not wish *Chromaphone* to respond to MIDI bank changes, open the *MIDI* configuration window by clicking on the *MIDI* button and uncheck the **Enable Bank Changes** option.

6.2.6 Pitch bend

The MIDI pitch wheel allows one to vary the pitch of *Chromaphone*. The pitch wheel can be moved with the mouse but it is also automatically connected to the pitch wheel signal received from your MIDI keyboard.

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 MIDI configuration window by clicking on the **MIDI** button located just below the MIDI led in the top part of the interface and use the **Pitch Bend Range** drop-down list to select the range in semi-tones.

6.2.7 Modulation wheel

Chromaphone responds to MIDI modulation (MIDI controller number 1). For more details, please refer to Section 4.11.6.

7 Using *Chromaphone* as a Plug-In

Chromaphone is available in VST, RTAS and AudioUnit formats and integrates seamlessly into the industry most popular multi-track recording and sequencing environments as a virtual instrument plug-in. *Chromaphone* 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*.

7.1 Audio and MIDI Configuration

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

7.2 Automation

Chromaphone supports automation functions of host sequencers. All parameters visible on the interface can be automatized except for the **Polyphony**, **Bank**, **Program** and **History** commands.

7.3 Multiple Instances

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

7.4 MIDI Program Change

MIDI program changes are supported in *Chromaphone*. When a MIDI program change is received by the application, the current program used by the synthesis engine is changed to that having the same number, in the currently loaded bank, as that of the MIDI program change message.

7.5 Saving Projects

When saving a project in a host sequencer, the currently loaded program 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 banks of programs 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 bank you are using in your project still exists on your disk when you reload the project. The programs must also exist and be in the same order as when the project was saved.

7.6 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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