ULTRA ANALOG SESSION

USER MANUAL

A|A|S

Applied Acoustics Systems

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Contents

1 Introduction

Ultra Analog Session is a virtual analog synthesizer. This special edition is based on modules from the well-known AAS *Ultra Analog-VA* synthesizer. With this instrument, we have not attempted to emulate a specific vintage analog synthesizer but rather to combine into a modern instrument, different features of legendary synthesizers. The design of this instrument allows one to get a wide range of tone from vintage analog to that of modern virtual analog synthesizers. Key controls and effects have been included resulting in a simple yet creative and versatile instrument.

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

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

1.1 System Requirements

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

Mac OS

- Mac OS X 10.11 (El Capitan) or later
- Intel Core i5 (circa 2012), Apple M1 processor or later
- 64-bit DAW

Windows

- Windows 10 64-bit or later
- Intel Core i5 (circa 2015) or later
- 64-bit DAW

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

1.2 Installation and Authorization

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

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

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

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

1.3 Getting Started

1.3.1 Using *Ultra Analog VA* in Standalone Mode

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

- **Windows** - Double-click on the *Ultra Analog VA* icon located on your desktop or select *Ultra Analog VA* from the **Start** > **All Programs** > menu.
- **Mac OS** - Double-click on the *Ultra Analog VA* 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 *Ultra Analog VA* 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.

1.3.2 Exploring the Factory Sounds

Ultra Analog VA 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 *Ultra Analog VA* offers is simply to go through the programs one at a time.

Ultra Analog VA 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.

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

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

1.4 Getting Help

AAS technical support representatives are on hand from Monday to Friday, 9am to 6pm EST. Whether you have a question on *Ultra Analog VA*, 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:

1.5 About this Manual

Throughout this manual, the following conventions are used:

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

2 Architecture of the *Ultra Analog Session*

Ultra Analog Session is a synthesizer based on the AAS modeling technology. It is a simple synthesizer yet its clever design results in a surprisingly versatile instrument that is really fun to play with.

The sound source is a rectangular wave **VCO** with hard synchronisation possibilities. This wave generator is also modulated by a triangular waveform **LFO** allowing for a wind range of sounds. This source is sent to a resonant low pass filter and an effect processor for further sound shaping. *Ultra Analog Session* includes an unison mode and features an arpeggiator for the creation of rhythmic sequences. The functioning and different parameters of these modules are discussed in more details in the next section.

Figure 1: Graphical user interface of *Ultra Analog Session* Synthesizer.

3 Bank and Program Management

Ultra Analog Session 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 contaning so-called *programs*. The name of the currently selected bank is shown in the *Bank* drop-down display located at the top of the *Ultra Analog Session* 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.

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.

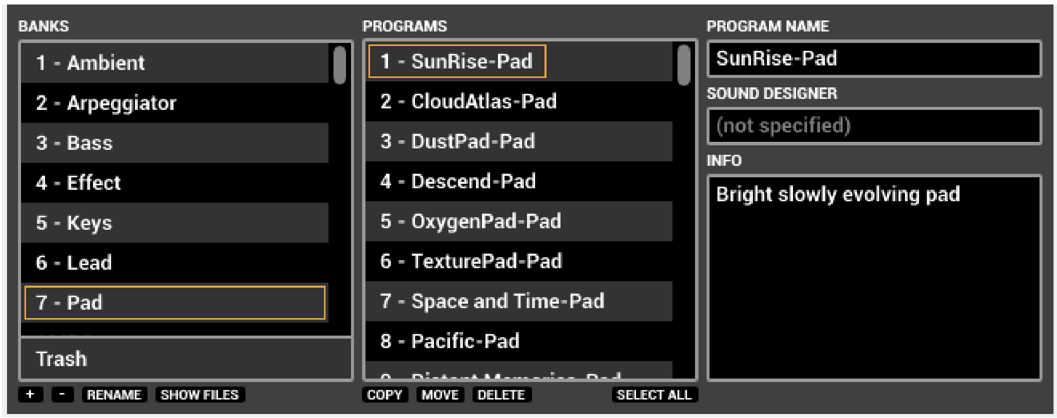


Figure 2: Bank and program manager window.

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/Ultra Analog Session 2/Banks

On **Windows**:
%AppData%\Applied Acoustics Systems\Ultra Analog Session 2\Banks

The bank files saved by *Ultra Analog Session* are named using the following convention:

[name of bank].UAS2 Bank

These file contain all the information corresponding to the programs they include. These files can be displayed directly from *Ultra Analog Session* 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 *Ultra Analog Session* 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 *Ultra Analog Session*.

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 **Mac OS** startup disk:
/Library/Application Support/Applied Acoustics Systems/Ultra Analog Session 2/Factory Library

On **Windows 64-bit**:
C:\Program Files (x86)\Applied Acoustics Systems\Ultra Analog Session 2\Factory Library

On **Windows 32-bit**:
C:\Program Files\Applied Acoustics Systems\Ultra Analog Session 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 *Ultra Analog Session* 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.

4 Parameters

This section can be used as a reference for the different controls appearing on the *Ultra Analog Session* 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

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

Fine adjustment of a control is obtained by holding down a modifier key of the computer keyboard (Shift, Ctrl, Command or Alt key) while adjusting the parameter. Double clicking on a knob brings it back to its default value when available.

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 the **Arpeggiator** module.

4.1.3 Drop-down Menus

Clicking on black display controls reveals a drop-down menu with a set of possible settings for the control. Adjustment of the parameter is obtained by clicking on a selection.

4.1.4 Modulation Signals

Different parameters can be modulated by the signal from the envelope generators and the **LFO** modules. Modulation signals are controlled with small gain knobs located on the left of the corresponding modulated parameters. The amplitude of the modulation is zero when the knob is centred. It is increased by moving it from its middle position clockwise or anti-clockwise. When turning it anti-clockwise, the phase of the modulating signal is inverted while it is preserved when moving it clockwise.

4.1.5 Scaling

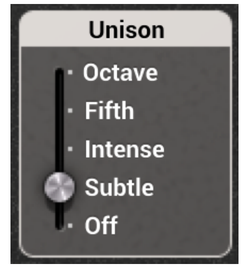
The size of the graphical interface can be adjusted by click-dragging the handle in its lower right corner. While resizing the interface the zoom factor is displayed in the upper left corner of the interface. This zoom factor can always be displayed by positioning the mouse on this corner. When clicking on the zoom factor, a drop-down menu with specific size ratios is displayed. Selecting a value resizes the interface automatically to that ratio. Note that this resize feature is only available for 64-bit versions of the program.

4.2 The Mode Module

The **Mode** module is used to control how the synthesizer voices respond to the events coming from an external MIDI keyboard or from a MIDI sequencer. When the switch is in the *Mono* position, the synthesizer is in monophonic mode allowing one to play only one note at a time. In the *Mono Glide* position, the synthesizer is still in monophonic mode but the pitch slides between notes rather than changing immediately from note to note. In the *Poly* position, the synthesizer is in polyphonic mode, allowing one the play chords. In the *Arp* position, the **Arpeggiator** module is switched *on* and notes are played sequentially depending on the settings of the **Arpeggiator** module. In the *Arp Glide*, the arpeggiator is active and the pitch slides between notes as in the *Mono Glide* case.

4.3 Unison

The unison mode allows one to play two voices for each note played on the keyboard. This mode creates the impression that more than one instrument are playing the same note together, adding depth to the sound.



In the *Fifth* and *Octave* positions, the second voice is one fifth or one octave above the note played respectively. In the *Subtle* position, the second voice plays the same note as the first one but it is slightly detuned resulting in a slow beating effect. In the *intense* position, the second note also plays the same note as the first one but the detuning between both voices is more important resulting in a richer sound with a rapid beating effect.

4.4 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 modules allows one to produce a wide range of arpeggios and rhythmic patterns and to sync the effects to the tempo of an external sequencer.

Arpeggio Patterns

The arpeggio pattern is set by the combination of the value of the *Range*, 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 -2 and 2, the notes played are transposed and played sequentially, over a range of one or two octaves. The direction of the transposition depends on the sign of the *Range* value with downwards transposition for negative values and upwards transposition for positive ones. 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, *RnRx* (Rock and Roll exclusive) and *RnRi* (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.

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

When *Ultra Analog Session* is launched in standalone mode the tempo of the arpeggiator, in bpm, is set by using the Rate knob. The tempo can also be adjusted by clicking at the desired tempo on the *Tap* tempo pad of the module. Once the new tempo is detected, the value of the *Rate* knob is automatically adjusted.

When using *Ultra Analog Session* in plugin mode, the *Tap* tempo pad is replaced by a *Sync* 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.

The rhythmic value of each step is set using the *Steps* parameter. Values 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.

4.5 The Oscillator Module

The **Oscillator** module is the sound source of *Ultra Analog Session*. It is based on a square wave generator with pulse width adjustment and hard sync feature allowing for a rich spectrum of sounds. The **Oscillator** module is implemented using precise modeling algorithms rather than wave tables providing alias free waveforms.

The pulse width of the rectangular wave is controlled using the *PW* knob. The width of the pulse varies from 4% in its leftmost position to 50% in its rightmost position or in other words to a perfectly square wave and therefore a spectrum containing only odd harmonics.

Synchronization

The **Oscillator** module features a hard synchronization mode. Synchronization involves a master and a slave oscillator. In this mode, the signal from the slave **Oscillator** is reset, or in other words

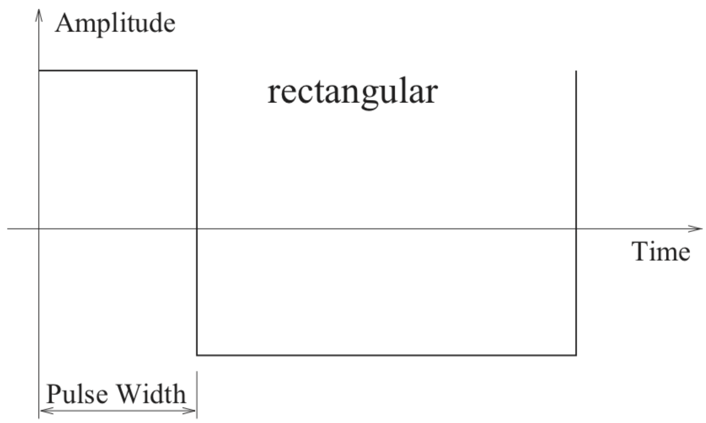


Figure 3: Choice of wave forms provided by the **Oscillator** module.

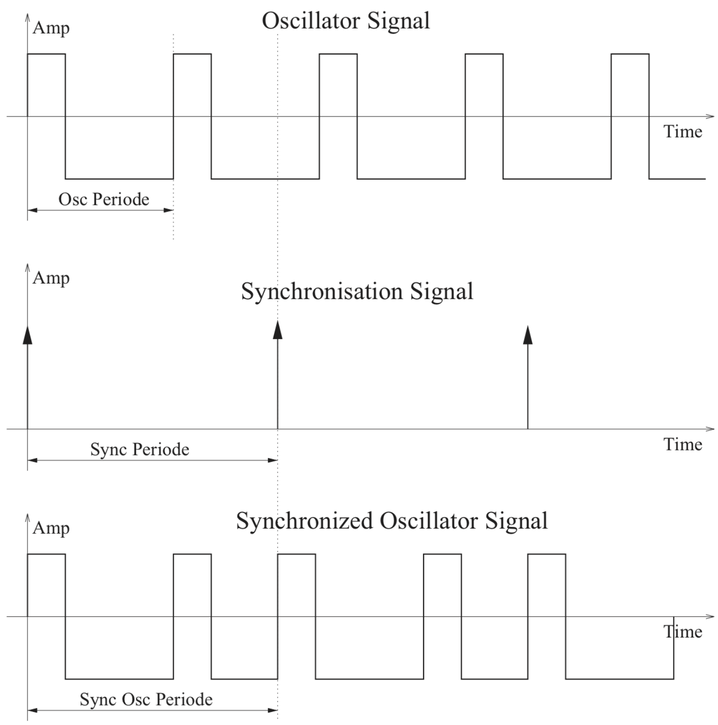


Figure 4: Synchronization of the oscillator.

restarted, at the beginning of each period of the waveform of the master oscillator which therefore acts as a master clock as shown in Figure 4.

The perceived pitch of the final output of the oscillator module is the same as that of the master oscillator while the frequency of the slave or synced oscillator only affects the harmonic content and therefore the timbre of the resulting signal. The frequency of this slave oscillator is adjusted using the *Sync* knob and its value is presented as a ratio between the frequency of the slave and that of the master oscillator. In its leftmost position, this ratio has a value of 1 which is equivalent to deactivating the synchronization mode. The ratio, or frequency of slave oscillator is increased by turning this knob clockwise.

The ratio value can further be modulated by an **LFO** with a triangular waveform. The amplitude of the modulation is controlled using the *LFO* knob. When in its center position, the amplitude is zero and there is no modulation. The depth of the modulation is increased by turning the knob to the left or right. When turned anti-clockwise, the slope of the modulation is negative while it is positive when turned clockwise. The frequency of the modulation is adjusted using the *Rate* slider. A light modulation results in an effect similar to a vibrato or a pulse width modulation while a deep modulation is close to an aggressive filter sweep effect. Note that the **LFO** waveform is automatically re-trigged when a new note is played except when the playing mode is set to *Arp* or *Arp Glide*. When using the *Mono Glide* mode, the signal from the **LFO** module is only re-trigged if a note is not played legato.

4.6 The Filter Module

Ultra Analog Session is equipped with a 12dB/oct low-pass resonant filter. A low-pass filter is used to remove the higher spectral components of the signal while leaving the lower components unchanged. The frequency at which attenuation begins to take effect is called the *cutoff* frequency. In a resonant filter, frequencies located around the cutoff frequency can also be emphasized by an amount called the *quality factor* or *Q-factor* of the filter as illustrated by Figure 5. The higher the Q-factor, the louder and sharper the response of the filter around the cutoff frequency. When the Q-factor is set to 1 (*Q* knob fully turned to the left), there is no emphasis around the cutoff frequency and the attenuation is -3dB at the cutoff frequency. High values of the of the resonance result in interesting saturation effects.

The resonance or cutoff frequency of the filters is adjusted with the *Frequency* knob while the amount of resonance or *Q-factor* is controlled with the *Resonance* knob. The value of the *Frequency* parameter corresponds to the value of the cutoff frequency when a middle C note is played (MIDI note number 60). The cutoff frequency is then automatically modulated depending on the note played on the keyboard with increasing values for notes above middle C and a lower value for notes below middle C.

The cutoff frequency of the filter can be modulated by an envelope generator creating filter sweeps effects. The amplitude of this modulation is controlled using the *Envelope* parameter. There is no modulation when this knob is centred. The depth of the effect is increased by turning the knob to the left or to the right. When turned anti-clockwise the filter sweep starts at a frequency below the cutoff frequency while it starts above the cutoff frequency when turned clockwise. Note

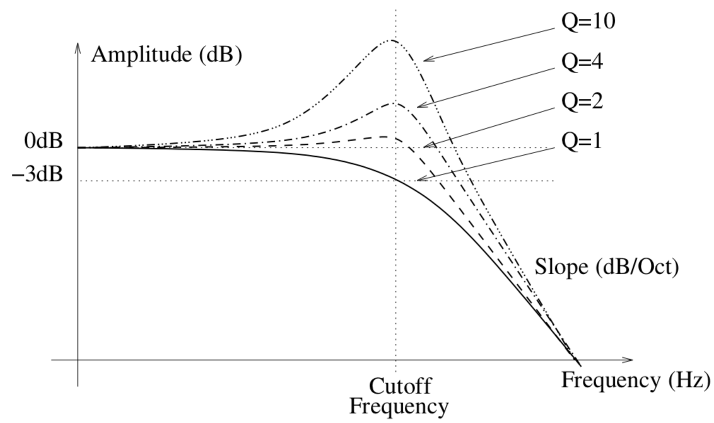


Figure 5: Frequency response of the low-pass filter.

that this knob can be set to its center frequency by double-clicking on it. The *Decay* slider is used to adjust the decay time of the envelope.

4.7 The Amp Module

After filtering, the signal is routed to an amplifier in order to add an amplitude envelope to the sound. The envelope is generated through the use of a standard ADSR (attack, decay, sustain, release) module with exponential segments.

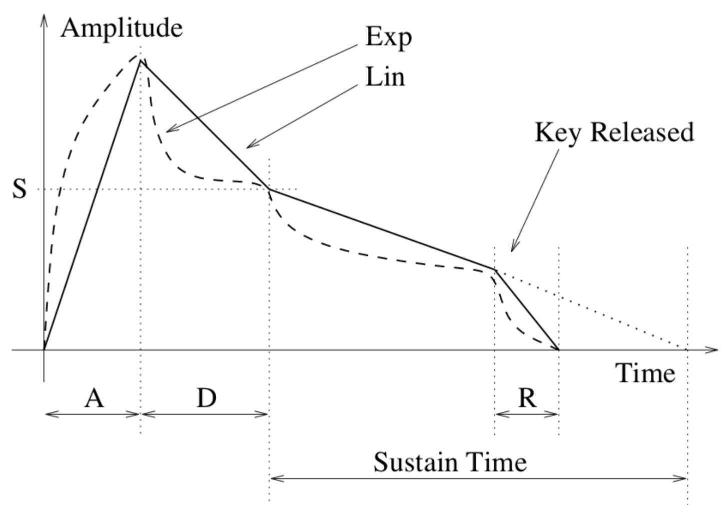


Figure 6: Response curve of an envelope generator. Broken line: exponential, full line: linear.

The ADSR module generates a four segment envelope: attack, decay, sustain, release as illustrated in 6. The attack time is adjusted using the *Attack* knob and controls how long the envelope takes to reach its peak amplitude. In its leftmost position, attacks are extremely short and smoother attacks are obtained by turning the knob clockwise. The decay time is the time taken by the envelope to go from its peak level reached at the end of the attack phase to the sustain level. It is controlled using the *Decay* knob. The sustain phase of the envelope generator lasts from the end of the decay phase until the key is released. The level of the envelope is controlled using the the *Sustain* knob. In its leftmost position, the sustain level is zero and there is no sustain phase while fully turned to the right, the sustain level is set to the peak value of the envelope and there is therefore no decay phase. When the key is released, the envelope generator toggles to the release phase and the envelope signal decreases from the value at the end of the sustain phase to zero in a time set by the *Release* knob.

Note that the envelope is triggered every time a new note is played except when notes are played legato or when the *Mono Glide* or *Arp Glide* modes are used.

4.8 The Effect Module

The **Effect** module allows one to process and shape the signal from the synthesizer before sending it to the output. The active effect is selected using the *Type* drop-down menu and options include, chorus, delay, distortion, equalizer, flanger, phaser and reverb. The controls appearing on the module depend on the choice of effect. Note that the **Effect** module can be turned *on* or *off* by using the green switch in the upper right corner of the module.

4.8.1 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 7. 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.

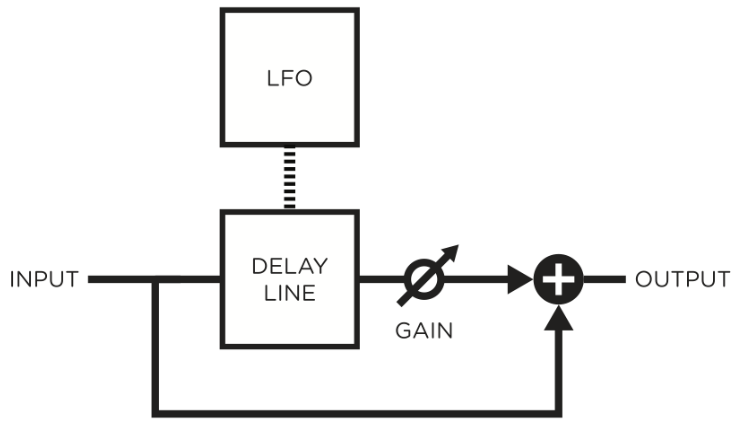
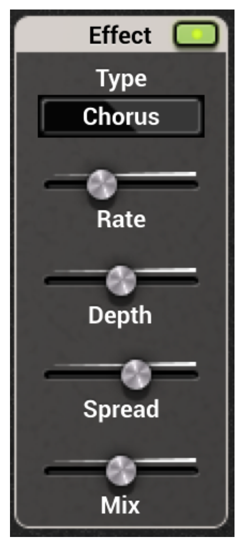


Figure 7: **Chorus** module.

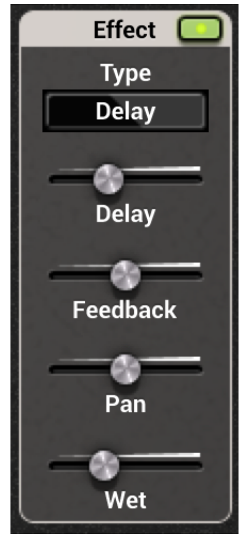
Tuning



The amount of modulation of the length of the delay lines is adjusted using the *Depth* slider. In its left position, there is no modulation and the length of the delay lines remains constant. As the slider is moved to the right, the length of the delay line starts to oscillate by an amount proportional to the value of this parameter thereby increasing the amount by which the different voices are detuned. The frequency of the modulation is fixed with the *Rate* slider.

The *Spread* slider 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* slider allows one to mix the dry and wet signals. In its leftmost position, there is no output signal from the chorus and one only ears the dry input signals. In its rightmost position, one only ears the wet signal from the chorus module. In its center position, there is an equal amount of dry and wet signal in the output signal from the module.

4.8.2 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* slider is used to adjust the amount of delay, in seconds, introduced by the effect. Pushing this slider to the right 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. Moving this slider to the right 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.

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

The output signal from the **Delay** module can include a mix of input signal (dry) and delayed signal (wet). The *Wet* slider is used to adjust the amplitude of the wet component in the final output. The amplitude is increased by moving the slider to the right from no signal to an amplitude of +6dB.

4.8.3 Distortion

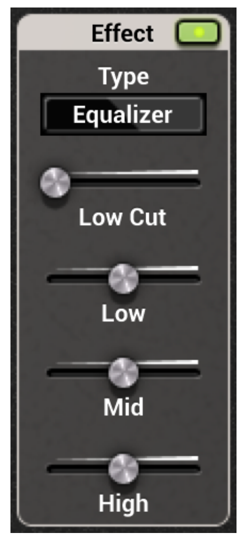
The distortion effect colors the sound by the addition of overtones in the spectrum. In the time-domain, this is obtained by limiting or clipping the amplitude of the input signal of the module.

The *Drive* parameter is a gain control 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; moving this slider to the right allows one to increase its amplitude.

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

The *Volume* parameter is a gain control acting on the amplitude of the distorted signal, the amplitude increasing when moving the slider rightward. Finally, the *Mix* slider 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.

4.8.4 Equalizer



The **Equalizer** effects provides equalization over the low, mid, and high frequency bands. It is composed of a low cut filter, a low shelf filter, a peak filters, and a high shelf filter in series.

The low cut filter is a simple 12 dB/oct high pass filter. The cutoff frequency of the filter is set using the *Low Cut* slider and can take values between 0 and 5 Hz and 2 kHz.

The functioning of the low shelf filter is depicted in Figure 8. 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 set to 130.81 Hz (C3) and the *Low* slider is used to adjust the gain factor applied to the signal. In its center position there is no attenuation (0 dB). Moving it to the right boosts the amplitude of low frequencies while moving it to the left 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 8. The filter applies a gain factor to components located above a cutoff frequency while leaving those below unchanged. The cutoff frequency of this filter is set to 523.25 Hz (C5) and the gain factor applied to the signal is adjusted using *High* slider. In its center position there is no attenuation (0 dB). Moving it to the right boosts the amplitude of high frequencies while moving it to the left reduces it.

The **EQ** module features a peak filter allowing one to shape the signal in a frequency band as illustrated in Figure 9. The filters apply a gain factor to frequency components in a band located around the cutoff frequency of the filters. This cutoff frequency is set to 261.63 Hz (C4) and the gain factor applied a this frequency is controlled by the *Mid* slider. When in its center position there is no attenuation (0 dB). Moving it to the right boosts the amplitude of frequencies located around the cutoff frequency while turning it to the left reduces it.

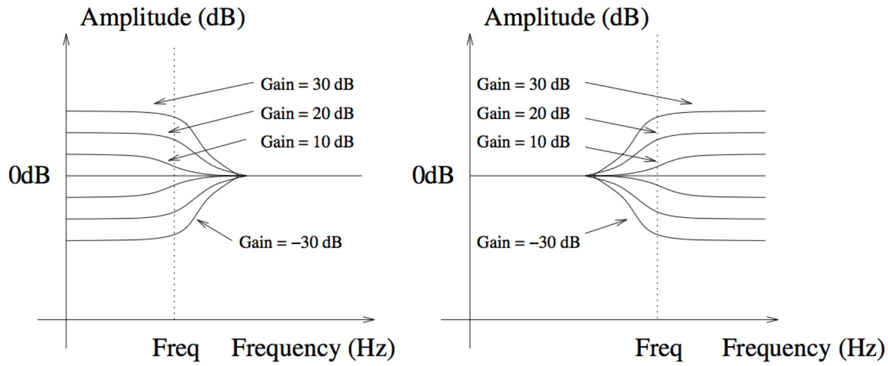


Figure 8: Low and high shelf filters.

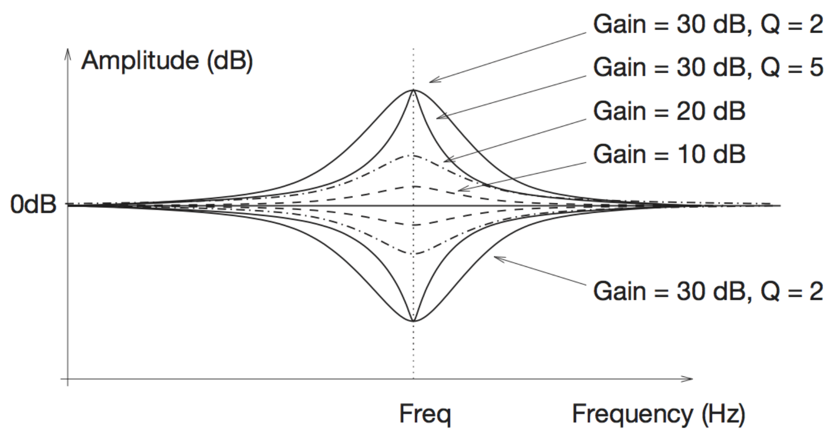


Figure 9: Peak filter.

4.8.5 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.

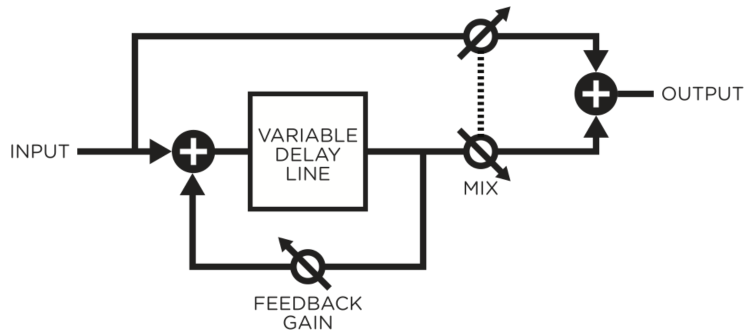


Figure 10: **Flanger** algorithm.

The algorithm implemented in this module is shown in Figure 10. 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.

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 11. The location of the fundamental frequency $f0$ and the spacing between the valleys and peaks of the frequency response is determined by the length of the delay line ($f0 = 1/(2\text{delay})$), the longer the delay, the lower is $f0$ and the smaller the spacing between the harmonics while decreasing the delay increases $f0$ and hence the distance between the harmonics.

The amount of effect is determined by the ratio of wet and dry signal mixed together as shown in Figure 12. 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 13. 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.

The delay length, in milliseconds, is adjusted with the *Delay* slider. The length of this delay can be modulated by a certain amount depending on the adjustment of the *Depth* slider. In the left position, there is no modulation and the length of the delay line remains constant. As the slider is moved towards the right, the length of the delay line starts to oscillate with increasing amplitude and at a frequency fixed with the *Rate* slider. Finally, the *Mix* slider determines the amount of dry

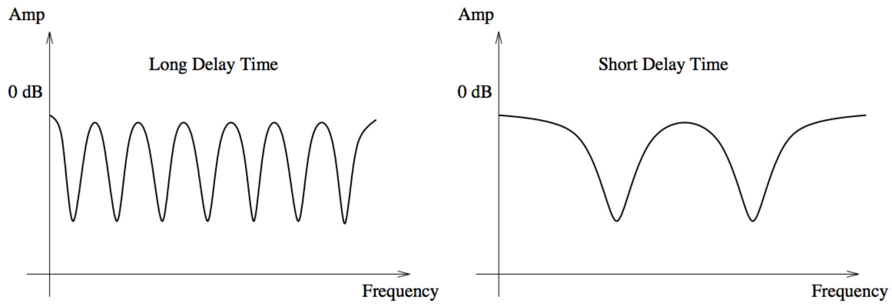


Figure 11: Frequency response of a **Flanger** module. Effect of the length of the delay line.

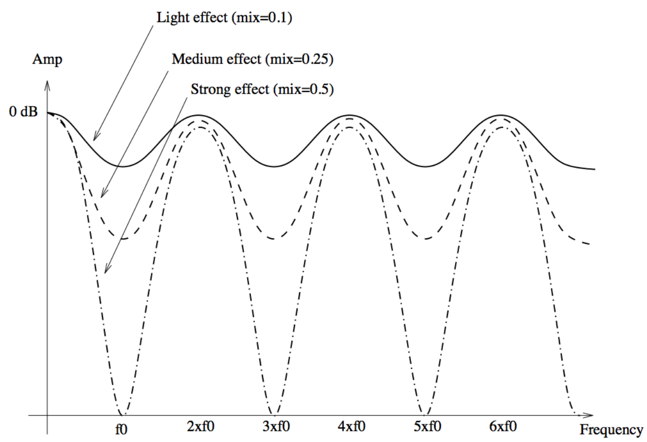


Figure 12: Effect of the mix between wet and dry signal on the frequency response of a **Flanger** module

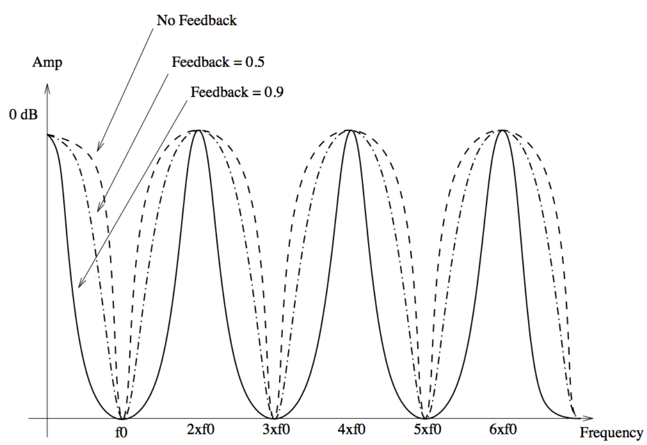
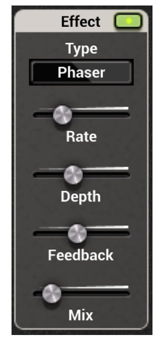


Figure 13: Effect of the amount of feedback on the frequency response of a **Flanger** module.

and wet signal in the output signal from the module. When 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.

4.8.6 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 14. 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.

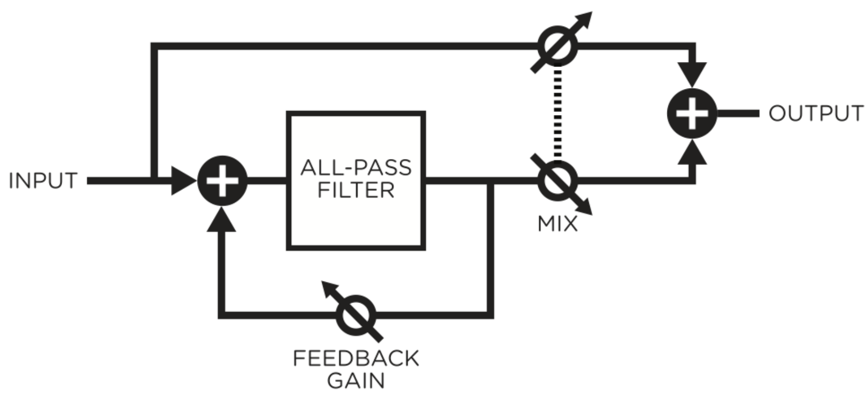


Figure 14: **Phaser** algorithm.

The all-pass filter modifies a signal by delaying its frequency components with a delay which increases with the frequency. This phase variations introduce a certain amount of cancellation when this wet signal is mixed down with the original dry signal as shown in Figure 15. 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 15. 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.

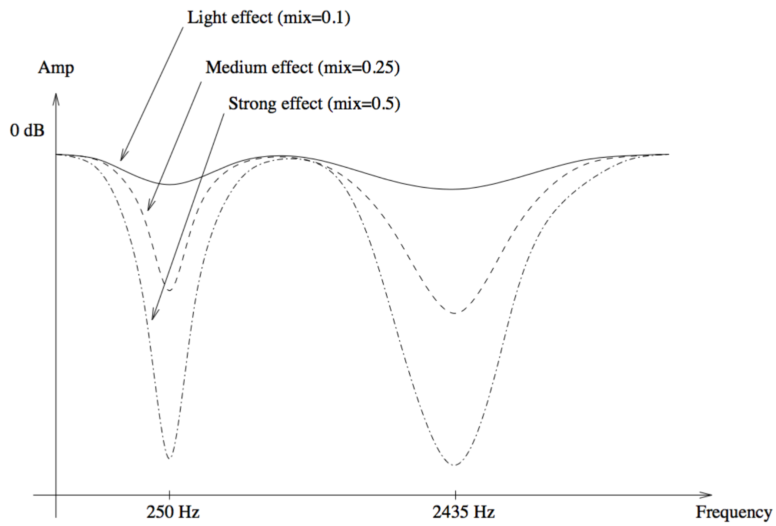


Figure 15: Frequency response of a **Phaser** module. Effect of the mix between wet and dry signal on the frequency response.

The location of the first notch in the frequency response of the module can be modulated by an amount controlled with the *Depth* slider. 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* slider is moved to the right. The frequency of the modulation is controlled using the the *Rate* slider. The *feedback* parameter is used to fix the amount of wet signal re-injected into the delay. Finally, the *Mix* slider determines the amount of dry and wet signal sent to the output. When this slider 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.

4.8.7 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 16 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).

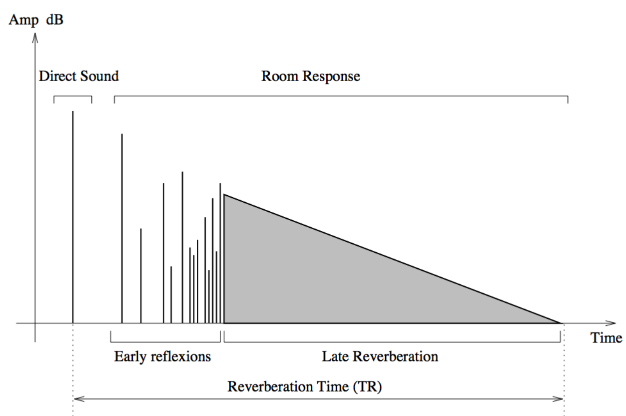


Figure 16: Impulse response of a room.

Adjusting the room effect

The size of a room strongly affects the reverberation effect. The *Size* parameter 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* slider. 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. The absorption level of high frequencies is adjusted with the help of the *High* slider.

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* slider 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.

4.9 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.

4.10 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 6.2.2.

4.11 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 *Ultra Analog Session* 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 *Ultra Analog Session*. 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.

Clicking on the *MIDI* button just below the MIDI led opens the MIDI configuration pop-up window from which one can adjust parameters such as the pitch bend range, enable bank and program changes and manage the MIDI map used by *Ultra Analog Session* as explained in Chapter 6

5.2 Polyphony

Ultra Analog Session always function in polyphonic mode with 16 voices of polyphony.

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 440Hz.

An interesting feature of *Ultra Analog Session* 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, *Ultra Analog Session* is set to Equal Temperament but other temperament 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 *Ultra Analog Session* 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.

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 *Ultra Analog Session*, 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 lever 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 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 *Ultra Analog Session*. Audio and MIDI configuration tools are accessed by clicking on the *Audio Setup* button located in the lower left corner of the *Ultra Analog Session* 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 *Ultra Analog Session* 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 is 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 doubles this time while changing the sampling frequency to 96 kHz and keeping the buffer size constant reduces the latency to 2.7 ms.

It is of course desirable to have as little latency as possible. *Ultra Analog Session* 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 your computer, the preset played, the sampling rate, and the number of voices of polyphony used. Note that it literally takes 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 you need 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 *Ultra Analog Session* 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 *Ultra Analog Session* 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, the controls move on the *Ultra Analog Session* interface just as if you had used the mouse.

In order to assign a MIDI link to a controller:

- On the *Ultra Analog Session* 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 links the control of *Ultra Analog Session* to the MIDI controller you just moved.

To deactivate a MIDI link, simply righ-click/Control-click on the corresponding control on the *Ultra Analog Session* interface and select the **MIDI Forget** command.

6.2.3 Creating a default MIDI Map

It is possible to define a a set of MIDI links, called a MIDI map, that will be loaded automatically when *Ultra Analog Session* 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

Ultra Analog Session 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 *Ultra Analog Session* 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 *Ultra Analog Session*, 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 *Ultra Analog Session* 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 *Ultra Analog Session*. 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

Ultra Analog Session responds to MIDI modulation (MIDI controller number 1). For more details, please refer to Section 4.10.

7 Using *Ultra Analog Session* as a Plug-In

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

7.1 Audio and MIDI Configuration

When *Ultra Analog Session* 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

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

7.3 Multiple Instances

Multiple instances of *Ultra Analog Session* can be launched simultaneously in a host sequencer.

7.4 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.

7.5 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.

8 License Agreement

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