USER'S MANUAL

ARP2600 V

TAE®
True Analog Emulation

Arturia

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Very special thanks to:

Celmar Engel, Mark Vail, Alan R. Perlman, Wally Badarou, Chad from Neptunes

Thanks to:

John Leimseder, Matt Lupo, Ron Kuper, Mephistoff Ellys, Pietro Caramelli
and the numerous beta testers.

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1 Introduction

1.1 The birth of ARP Instruments and the ARP2600

Alan R. Pearlman, whose initials would form the name of ARP Instruments, became interested in instruments for electronic music as early as 1948, when he was a student at the Worcester Polytechnic Institute. This was a means for him to associate his two passions: electronic music and the piano.

It was by commercializing the amplifier models for the NASA Gemini and Apollo programs that he would start his career. Around 1968 he started seriously imagining the possibility of building electronic instruments – after hearing a recording of "Switched-on Bach," according to legend.

In 1969, Alan R. Pearlman, David Friend and Lewis G. Pollock created ARP Instruments (originally called Tonus Inc.). The company, based in Newton Highlands (Massachusetts, USA), conceived electronic products, but also and above all else a large modular synthesizer, the ARP 2500. The machine used a matrix which connected the different sections of the synthesizer, instead of the traditional cables found in the Moog Modulars. The ARP 2500 found success in American universities.

The growth of ARP instruments was fast and in 1972 the ARP 2600, probably the most legendary of the entire range, was unveiled. This semi-modular synthesizer, conceived with an educational goal, was to become hugely successful after a shaky start. The ARP 2600 was notably used by Stevie Wonder, Joe Zawinul (Weather Report), Tony Banks (Genesis), Jean-Michel Jarre, Herbie Hancock... ARP was the market leader in synthesizers during the 70's with around 40% of the market share.

In ten years, three versions of the ARP 2600 were commercialized: The first version was called "Blue meanie" because of its steely blue finish. The "blue meanie" was quickly replaced by a second version, with a grey background finish and white silk screening (1972). This was to be more popular. In 1978 ARP decided to change the graphic chart for all of its machines: a black background color with orange silk screening was introduced. The ARP2600 benefited from its third and last version.

The great rival of ARP was Moog Music. The competition between the two manufacturers can easily be seen when we observe the machines: The ARP, for example, has linear potentiometers, while the Moog has rotating pitch bend and modulation wheels.

A well known episode of this competition was the 24 dB/octave filter, the 4012, used by the ARP. This was a replica of the famous Moog filter. In 1973, Moog threatened ARP with legal action and the firm decided to change the circuits on its filter. The 4072 was born and took the place of the 4012. This possessed a calibration error in the high frequencies – the maximum cut-off frequency was less than 11 kHz instead of the 16 kHz promised in the press. Luckily the repair for users was fast and not much of a burden. On the first ARP 2600's, the 4012 filter was still used (this was the case for the "Blue meanie" and on the first examples of the "grey and white") while the models that followed offered the 4072.

The ARP synthesizers possess very stable oscillators, more reliable than the Moog synthesizers (a fact admitted by Robert Moog himself). On the other hand, ARP for a long time dipped the electronic circuits for filtering in resin to avoid industrial piracy... this made for major problems when trying to perform a repair.

In 1972, ARP launched the Odyssey, which would be in direct competition with Moog Music and their Minimoog released one year earlier. The same year, the Pro-Soloist, a preset instrument, was also unveiled.

In 1976, ARP released a small 16 step sequencer in the form of 2 independent 8 step sequences. This became famous and is still very sought after (it is emulated in the ARP2600 V.) The same year they presented the Omni, which would become one of ARPs biggest successes. The instrument allowed the combination of two polyphonic violin sounds – a great innovation for the company – and 2 monophonic bass sounds.

In 1976, ARP released a small 16-step sequencer in the form of two independent 8-step sequences. This became famous and is still very sought after (it is emulated in the ARP2600 V.) The same year they presented the Omni, which would become one of ARPs biggest successes. The instrument allowed the combination of two polyphonic violin sounds – a great innovation for the company – and two monophonic bass sounds.

But in 1981, ARP was finally bought out by CBS. The following year, CBS with part of the ARP development team would produce the Chroma, a programmable polyphonic synthesizer, and in 1984 the Chroma Polaris, a simplified and MIDI-capable version of the Chroma.

1.2 A better emulation thanks to TAE®

TAE® - True Analog Emulation - is a new technology dedicated to the digital reproduction of analog circuits used in vintage synthesizers.

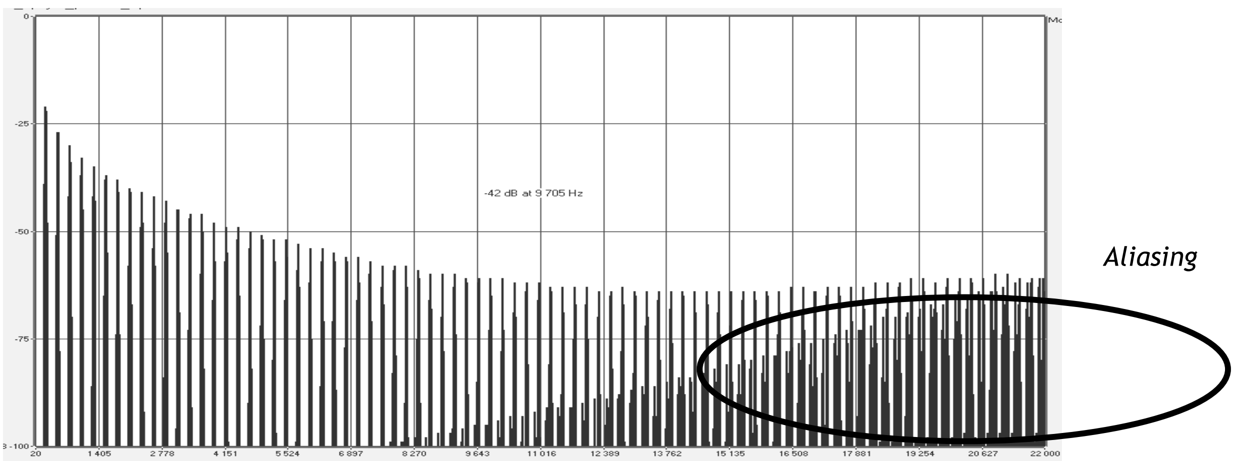
When implemented in software code, TAE's algorithms guarantee authentic emulation of hardware specifications. This is why your ARP2600 V offers an unparalleled quality of sound.

In detail, TAE® combines four major advances in the domain of synthesis:

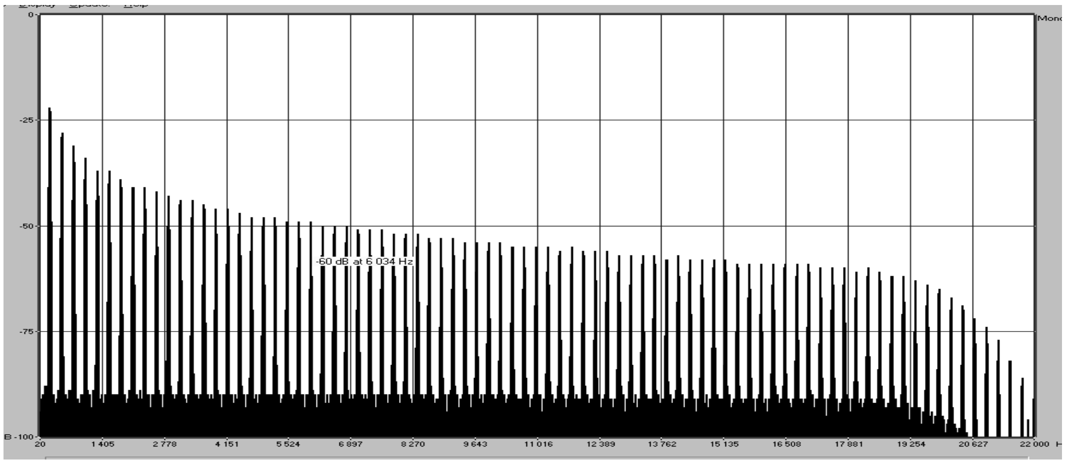
1.2.1 Aliasing-free oscillators

Standard digital synthesizers produce aliasing in high frequencies, and also when using Pulse Width Modulation or FM.

TAE® allows the production of totally aliasing-free oscillators in all contexts (PWM, FM…), and at no extra CPU cost.



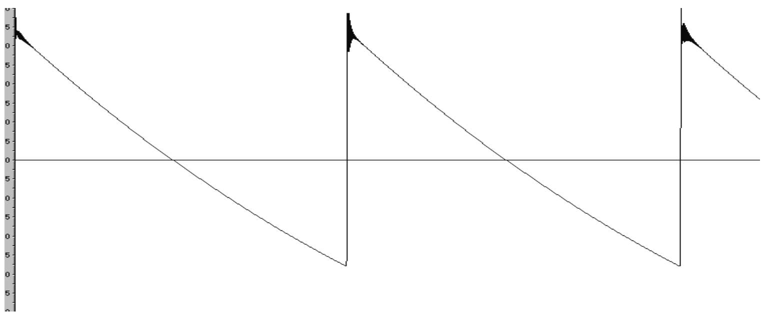
Linear frequency spectrum of an existing well-known software synthesizer



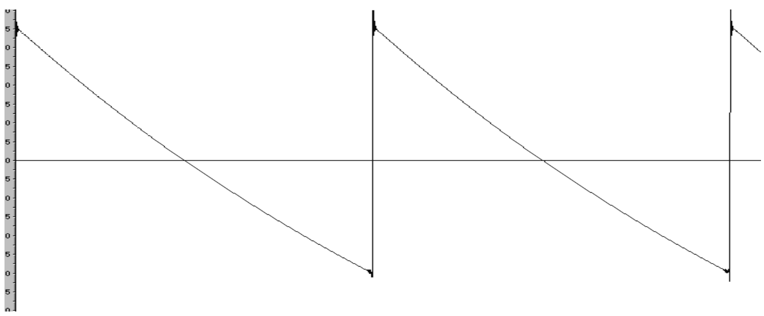
*Linear frequency spectrum of the ARP2600 V oscillator made with **TAE***

1.2.2 A better reproduction of analog oscillator waveforms

The waveforms produced by the oscillators in analog synthesizers are marked by the presence of a capacitor in the circuits. The discharge of the capacitor results in a light bend in the original waveform (notably for saw tooth, triangular and square waveforms). TAE allows the reproduction of this capacitor discharge. Underneath is the analysis of a waveform from the original ARP2600, and that of the ARP2600 V. They are both equally deformed by the ARP2600 V low-pass and high—pass filtering.



Temporal representation of a "saw tooth" waveform of the original ARP2600



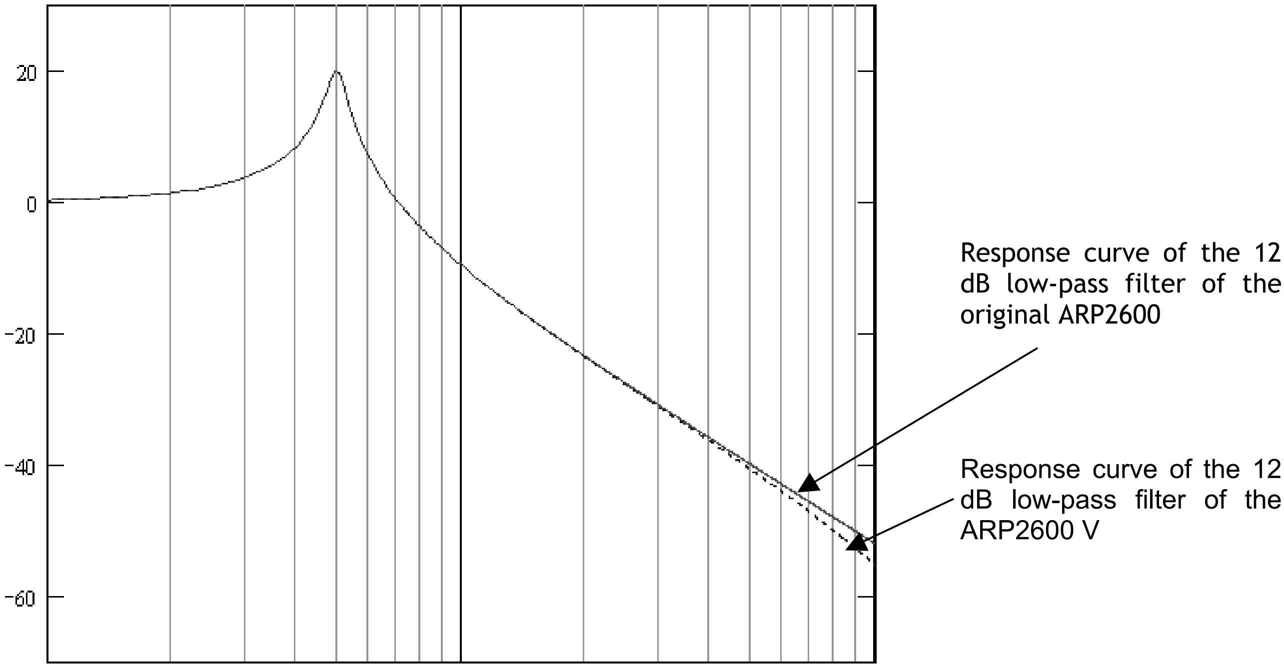
*Temporal representation of an ARP2600 V "saw tooth" waveform reproduced by **TAE***

What's more, the original analog oscillators were unstable. In fact, their waveform varied slightly from one period to another. If we add to this the fact that the starting point for each period (in Trigger mode) can vary with the temperature and other environmental conditions, we find one of the characteristics that contributed to the typical sound of vintage synthesizers.

TAE reproduces the instability of oscillators, bringing a fatter and "bigger" sound.

1.2.3 A better reproduction of analog filters

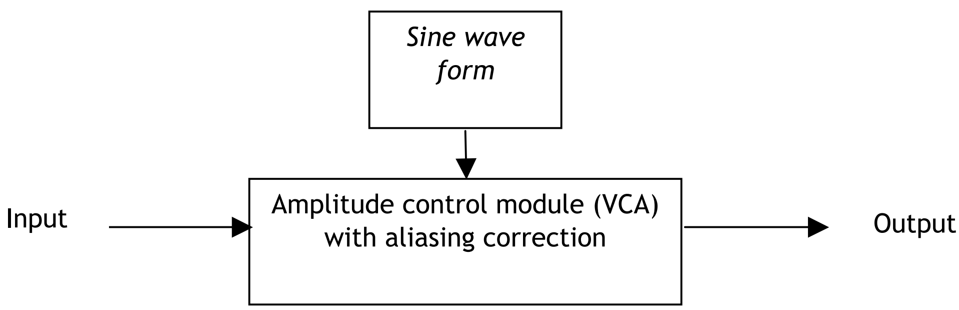
TAE allows more precise emulation of analog filters than standard digital filters. To obtain this result, the TAE technology is based on the analysis of the analog circuits to be reproduced, and converts them into algorithms that faithfully mimic the characteristics of the original filters. The curve underneath shows the comparison of the original ARP2600 filter and that of the ARP2600 V.



Response curve of the 12 dB low-pass filter of the original ARP2600 V and the ARP2600 V

1.2.4 Ring modulator

The ARP2600 V includes a ring modulator, just like the original ARP2600. The ring modulator allows the application of a waveform (a sinusoid) to another, in order to transform it. The result is a more brilliant sound, distorted, and enriched in harmonics. As a result of this increase in the number of harmonics, standard ring modulation algorithms create an audible aliasing. To avoid this unwanted effect, TAE includes a module for the dynamic control of the amplitude with aliasing correction, which removes every trace of aliasing in the signal coming from the ring modulator.



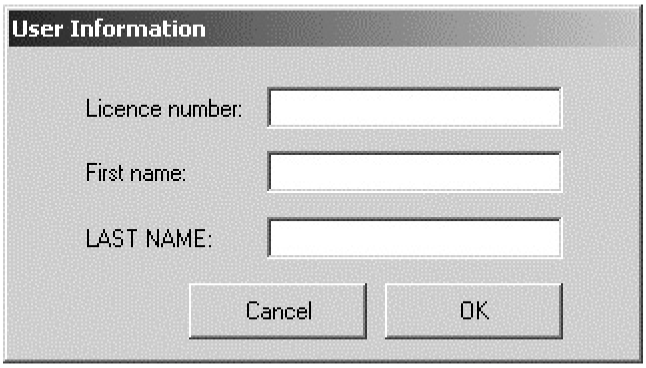
2 Installation

2.1 Windows installation (Win9x, Me, 2000, XP)

- Insert the CD-ROM into the drive. Explore the contents of the CD-ROM, and double-click on the icon named “ARP2600V Setup PC.exe”

First you will need to define an installation folder for the program. By default it will be installed in C:\Program Files\Arturia\ARP2600V. You can change this location with the Browse button.

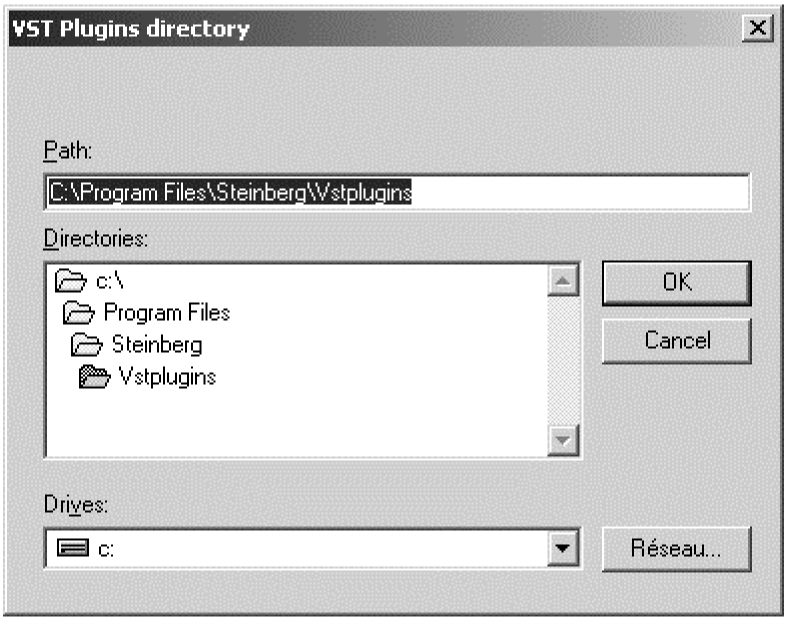
Next, have your license number ready and enter it along with your first and last name in the user information window.



User information window

The ARP 2600V will firstly be installed as a stand-alone program. The following step will give you the choice of configuring it as a plug-in as well. You will be asked to define the protocol(s) that you use (VST, RTAS, HTDM, DXI). For more information on these protocols please see Chapter 7.

For the VST and RTAS protocols, you will be asked to choose an installation folder so that the host application may use it as a plug-in. If you are not sure how to make this choice, please see Chapter 7.



Choice of installation folder for SVT plug-in

The installation program now has enough information to finish the procedure. In a few seconds, you will be using the ARP2600 V.

2.2 Mac OS X installation

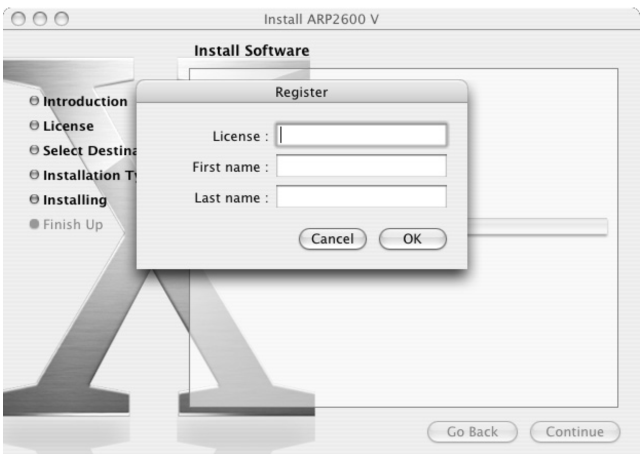
Insert the CD-ROM into the drive. Explore the contents of the CD-ROM, and double click on the icon named "ARP2600V Setup Mac".

Enter your administrator name and password in the authentication window.

The ARP2600 V will firstly be installed as a stand-alone program. By default, all the protocols (VST, RTAS, HTDM). will be directly installed on you computer. For more information on these protocols please see Chapter 7.

The ARP2600 V will be installed in your applications folder. You can also define a different drive and installation folder.

Next, have your license number ready and enter it along with your name and family name in the user information window. The installation program now has enough information to finish the procedure. In a few seconds, you will be using the ARP2600 V.

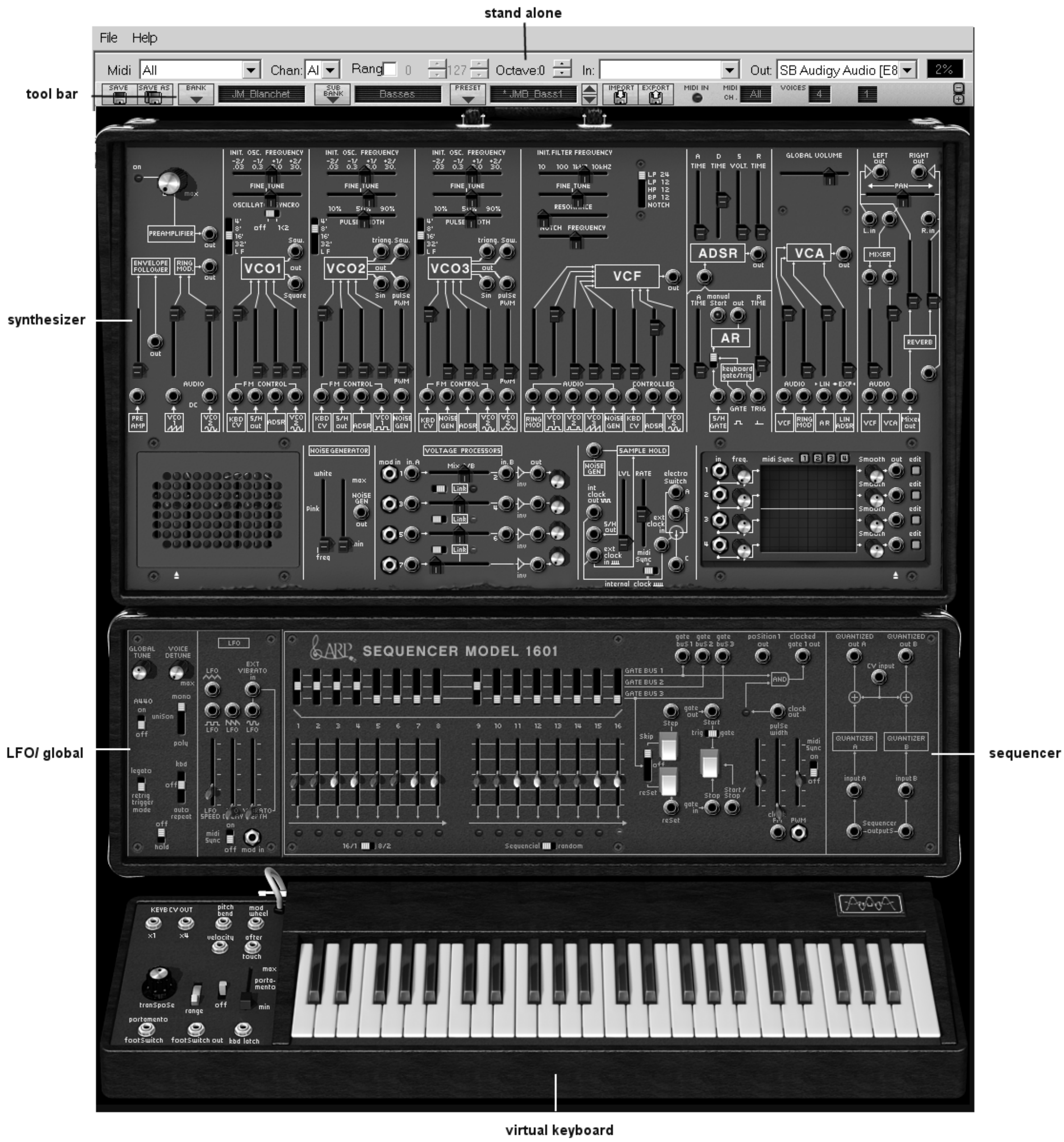


Enter your license number, then your first and last name

3 Quick start

This chapter will help you to familiarize yourself with the general basics of using the ARP2600 V. A summary of the different parts of the synthesizer will be presented to you as we guide you through your first use of the program. You will find a detailed and precise description of all settings and controllers in the following chapters.

Chapter 7, *A few elements of sound design*, is highly recommended for users who have never worked with a subtractive synthesizer and who wish to become familiar with the fundamentals in this domain.



Overview of the ARP 2600V

3.1 Using presets

The use of presets is one of the biggest improvements of the ARP2600 V compared to the original. In fact, the latter was unable to save sounds!

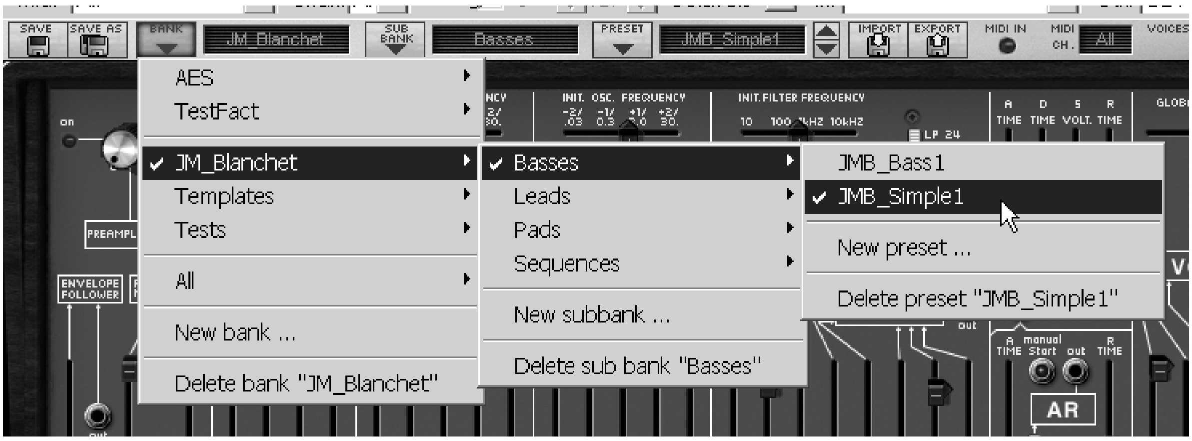
In the ARP2600 V, a preset contains all of the parameter settings of the synthesizer, including the synthesizer and different real-time controllers (eg: velocity, aftertouch, pitch-bend) as well as the effects (delay, chorus) needed to reproduce a sound.

To get to know the different sounds contained in the ARP2600 V, we will use the preset “Bass1” situated in the “JMBlanchet” / “Basses” bank.

- For this, click on the button above the “BANK” LCD display indicating “Factory” (this screen presents the name of the current bank). By clicking here you will see a menu appear containing the list of available banks. Choose the bank “JM_Blanchet” (the bank name is checked) .

When this menu is open, sub menus can be accessed (in the fashion of a drop-down menu). This system lets us access the “SUB BANK” and “PRESETS” for a sound designer with a single click.

- Choose the “SUB BANK” “Basses” and select “JMB_bass1” among the “PRESETS”.



Select the preset JMB_bass1

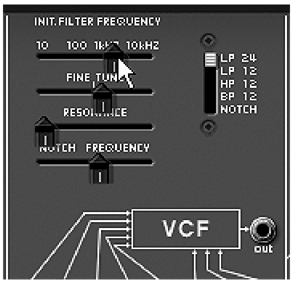
The ARP2600 V ships with more than 300 presets that will help you to get to know the sounds of the synthesizer. A bank named “User / Temp” offers a selection of template presets for beginning the programming of a sound (the sound “1_Osc”, for example, comes with: an oscillator directed to the low-pass filter, and then routed towards the VCA).

It is also possible to visualize the entirety of the presets corresponding to a type of sub bank by selecting the option “All” in the bank. For example, to see all of the bass presets, click on “All” in the bank selection and then on “Bass”.

3.1.1 Now modify a preset

For this, we will start with a very simple manipulation.

- Modify the brightness of the sound “JMB_Simple1” with the linear “Initial Cutoff Frequency” potentiometer of the filter. Raise or lower the potentiometer and notice the sound become more or less “bright”. Set this potentiometer to a pleasing value.



Change the brightness of the sound

- In the same manner, you can change the range of oscillator 1 by setting the “Range” selector to one of the values expressed in steps: LF = low frequencies, 32’ = -2 octaves, 16’ = -1 octave, 8’ = standard tuning, and 4’ = + 1 octave.

Setting the range of oscillator1

By performing these first settings, you have already modified the preset “JMB_Simple1”. You will now see how to save the sound that you have created.

- To choose another destination for the sound, click on the “Save as” icon and choose the location. For example select “new” in the choice of bank. Two new banks, sub banks and a preset are immediately created. The names “new bank”, “new sub bank…” and “new preset…” appear in their respective displays.
- Click on each of these displays to rename the 3 parts.

Saving a preset

To save a user preset (“**Users**”), click on the “Save” icon in the toolbar: The new settings will be saved in the current preset without changing the name (but if the selected preset is one of the “factory” presets, the factory setting will not be overwritten).

Attention! It is important to specify that changing the name of a preset does not create a new one! Only the name of the current preset will be changed.

3.2 The 3 sections of the ARP2600 V

The ARP 2600V offers three main sections separated into flight cases:

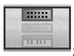
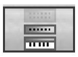
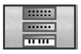
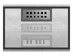
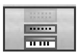
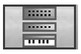
From top to bottom:

The synthesizer, the sequencer / LFO / general settings, and keyboard

To access the different parts of the ARP2600 V there are two simple methods:

- Click on a part of the synthesizer that des not have any controllers (potentiometers, switches..) or jacks, then slide the mouse towards the top or the bottom without releasing.

Slide the mouse towards the top or bottom

- Click on one of the three shortcuts ,  and  situated on the toolbar to quickly move from one section to another:
  - The  (SYNTHESIZER) section brings you to the synthesizer.
  - The  (SEQUENCER and KEYBOARD) section brings you to the sequencer.
  - The  (“ALL”) section brings you all the interface of the ARP2600 V

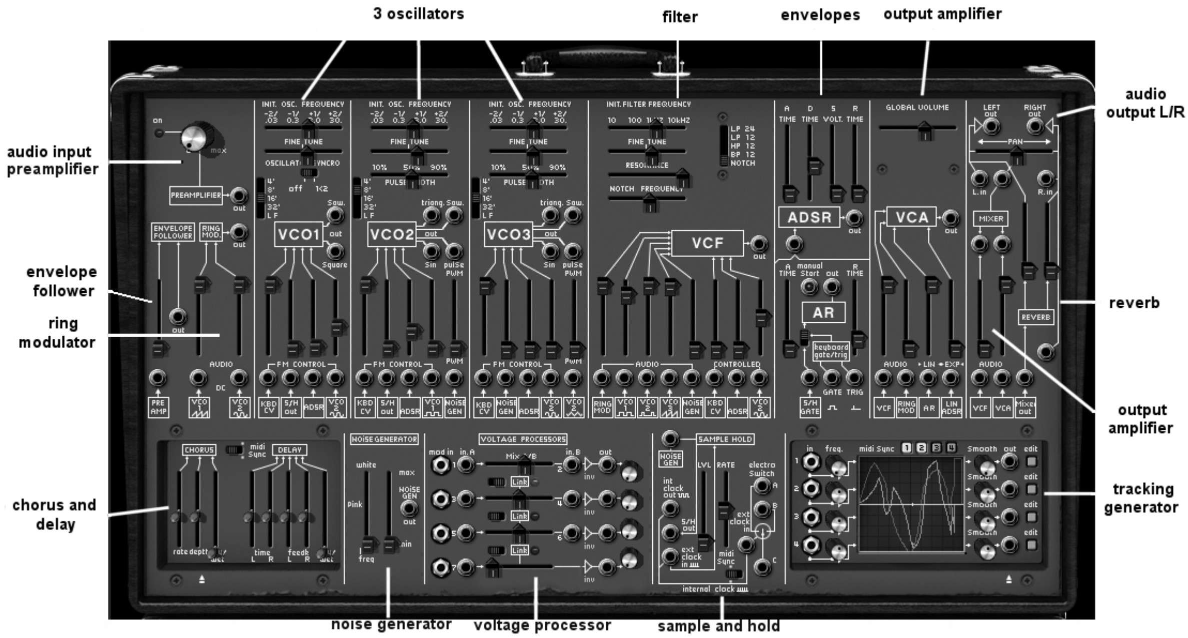
If you are in the “All” section, you can resize the window by clicking on the 2 arrows situated on the right of the 3 sections shortcuts.

3.3 Overview of the Synthesizer

The “SYNTH” section contains 73 synthesis parameters as well as jack inputs and outputs that you can connect to one another with virtual cables. The potentiometers or switches associated to these parameters will help you to create an infinite variety of sounds.

These parameters are made up of:

- Three oscillators (VCO) which release the audio signal through wave forms (triangle, sinusoid, saw-tooth, square and rectangle) and which manage the pitch (frequency) of the sound.
- A noise module.
- A ring modulator
- A sample / hold module.
- A mixer acting on the signals coming from the oscillators, noise module and ring modulator.
- A low-pass resonant 24 dB filter and multimode 12 dB (LP, HP, BP and notch)
- An amplifier (VCA) allowing the amplification of the signal coming from the filter and its direction towards the stereo output.
- Two envelopes (ADSR and AR) modulating the low-pass filters and amplifier.

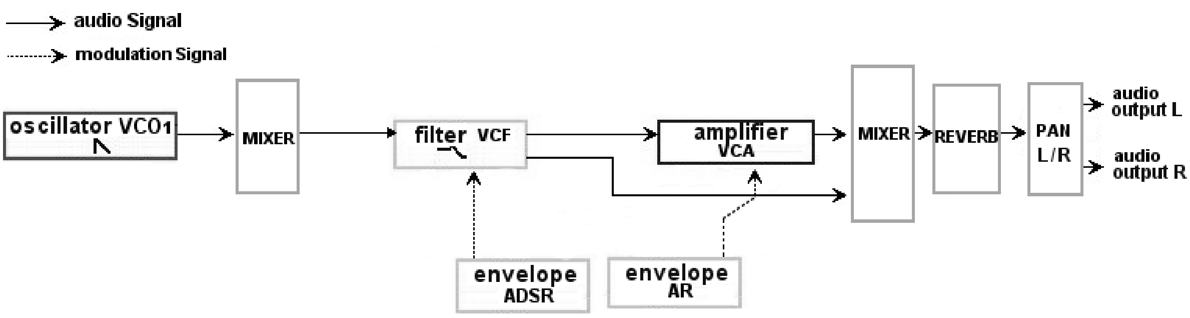


Synthesis parameters

Let’s look at quickly creating an evolving lead sound:

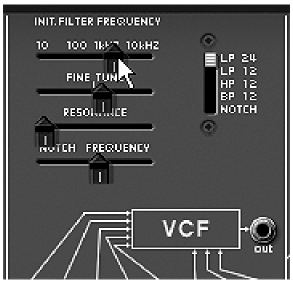
- To really understand the programming of the ARP2600 V, let’s use a very simple sound. Select the preset “1_Osc” in the “Template / Temp_Synth” sub bank. The structure of synthesis for this sound is relatively simple: the square wave form oscillator1 is active and the signal is directed through the low-pass filter after an intermediary mixer, and then on to the output amplifier. An ADSR envelope modulates the filter cut-off frequency and a second envelope, AR, modulates the volume of the amplifier.

  The following block diagram recaps the architecture of the creation of the sound:



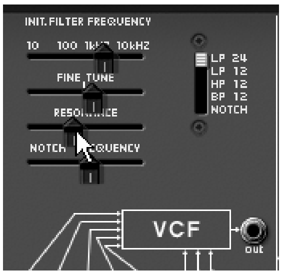
The routing taken by the sound of preset “1_Osc”

- Start by reducing the low-pass filter cut-off frequency (LPF). This will dampen the sound.
  For this, set the linear “Initial Cutoff frequency” potentiometer (for fine tuning, use the “Fine tune” potentiometer).
  Notice that the filter cut-off frequency is modulated by an ADSR envelope (Attack, Decay, hold - Sustain – and Release).



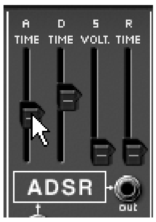
Change the brightness of the sound

- To clearly hear the effect produced by the ADSR envelope on the filter cut-off frequency, increase the resonance value. This will amplify the filtering effect on the sound and it will begin to “whistle”.



increase the resonance value

- Change the attack length of this envelope (“Attack time”) so that the brightness increases faster or slower when the note is sent.



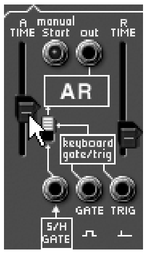
Change the attack length of the ADSR envelope

- In the same manner, change the value for the decay, and the brightness will increase faster or slower while you hold the note on the keyboard.

The “Decay time” parameter on the filter envelope

Now let’s perform a short modification on the second envelope, the “AR” envelope.

- Increase the “Attack time” for this envelope so that the volume of the sound progressively increases.



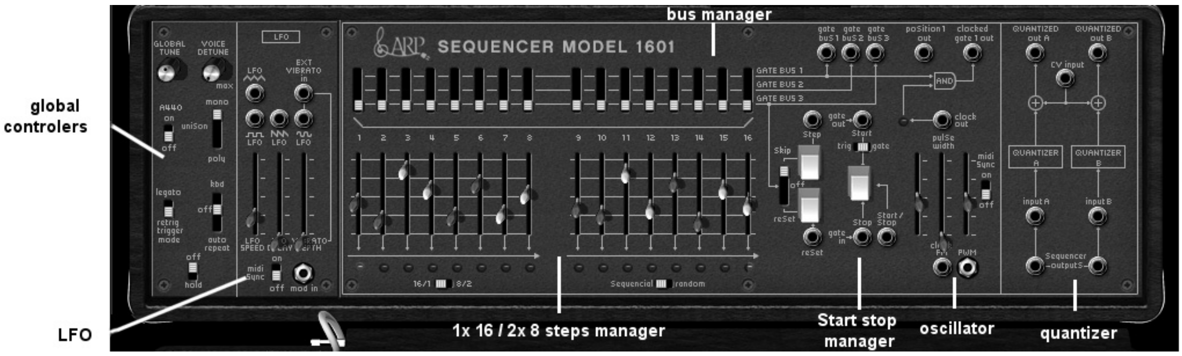
Increase the “Attack time” of the AR envelope

3.4 The sequencer

The “SEQ” section gives you access to the sequencer as well as different functions allowing an extension of the possibilities of synthesis and playing. It is situated under the “Synth” section. It contains a sequencer identical to the 16 step ARP (model 1601) sequencer, a module for play settings, and a low frequency oscillator (LFO) which was added as a complement to oscillator2 which was often used as LFO.

3.4.1 The ARP sequencer

The ARP sequencer greatly increases the possibilities for sound and melodic creation. It allows you to create two simultaneous 8-step melodic lines, or one 16-step line (by putting two 8-step lines in a series). It is also possible to modulate any parameter of synthesis through one of the two sequencer outputs.

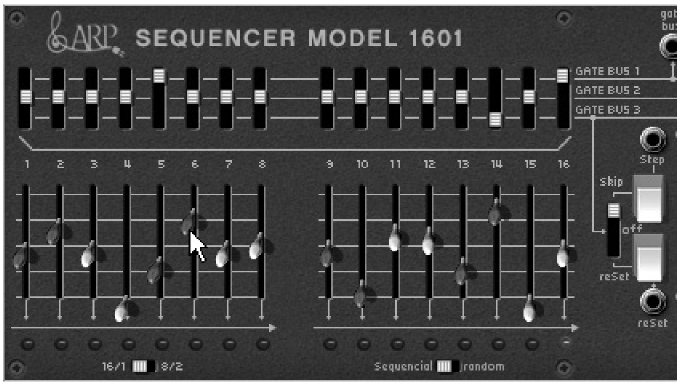


The ARP sequencer

The ARP sequencer contains 3 parts:

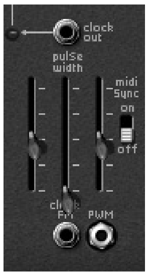
From left to right:

- The two lines of faders and selector switches situated at the top give access to the tuning of the 16 steps as well as management of their triggering (gates).



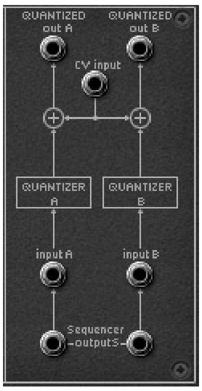
The two lines of faders and selector switches

- The oscillator sets the speed of the sequencer as well as the start and stop.



set the oscillator speed

- The “Quantizer” quantifies the values for the 16 steps by semi-tone.

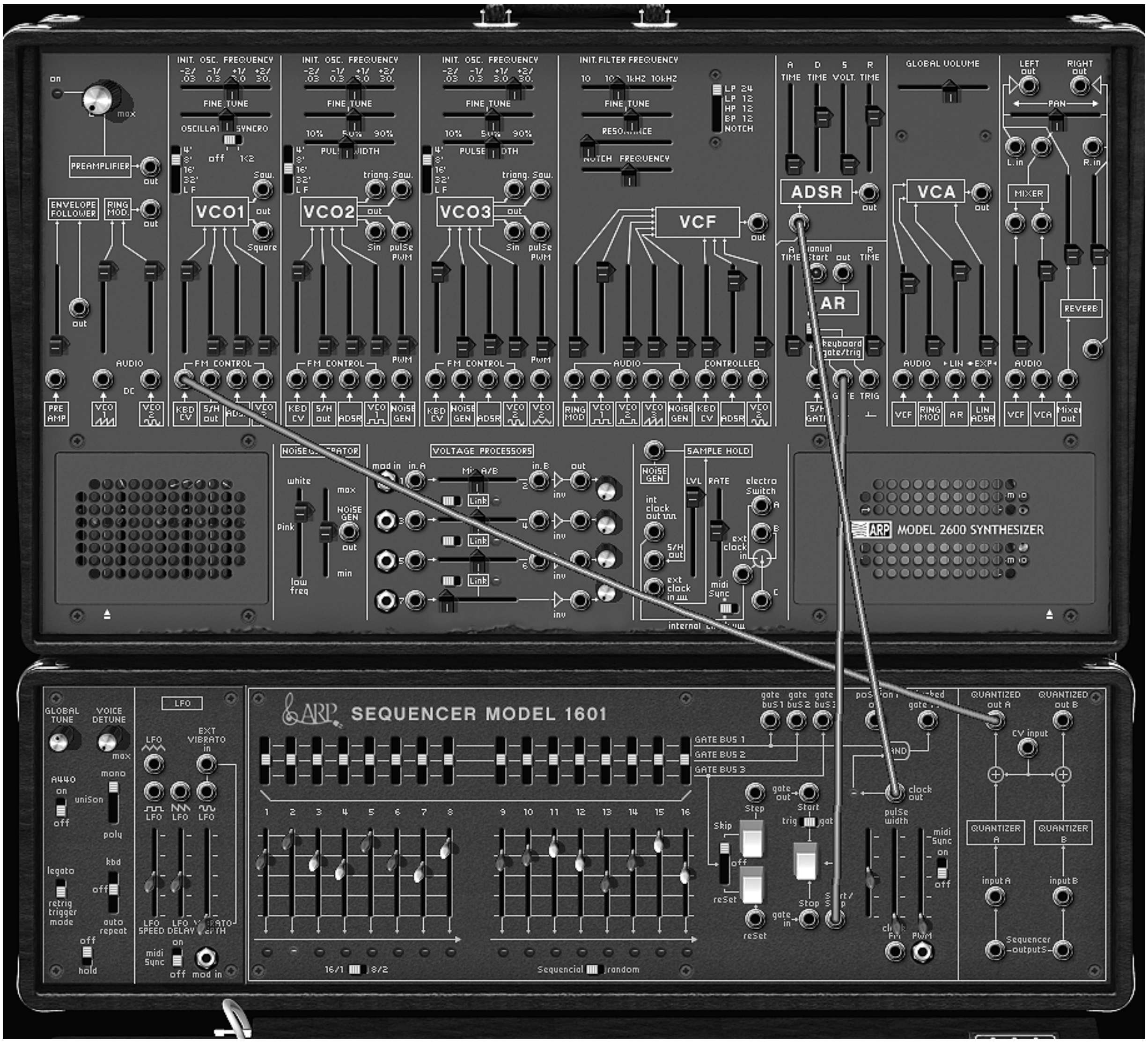


the quantizer section

Let’s take a simple melodic sequence for example:

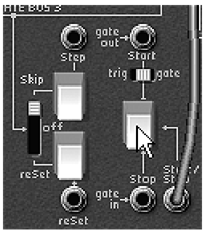
- Load the preset “Template”/ “Temp_SEQ”/ “1x16_sequencer”

You will notice that the connexions between the sequencer and synthesizer are already done:



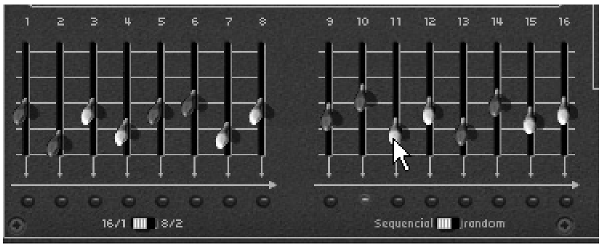
the connexions between the sequencer and synthesizer

- The “Clock Out” sequencer output is directed to the “Gate” input on the ADSR envelope module.
- The sequencer “Quantized A Out” output is directed to input “KBD CV” of the “VCO 1” module.
- Start the sequencer by clicking on the “Start” button. This “turns” in a loop and you will hear the repetition of the melody.



click on the “Start” button

- You can change the settings of the 16 linear potentiometers to create an other melody.



Set the 16 linear potentiometers

3.4.2 The LFO

On the original ARP 2600, oscillator2 could be set to low frequency position ("LF" position in the range) for use in LFO mode. Although practical, this solution prevented us from using 3 oscillators simultaneously and a slow modulation on the filter cut-off frequency for example.

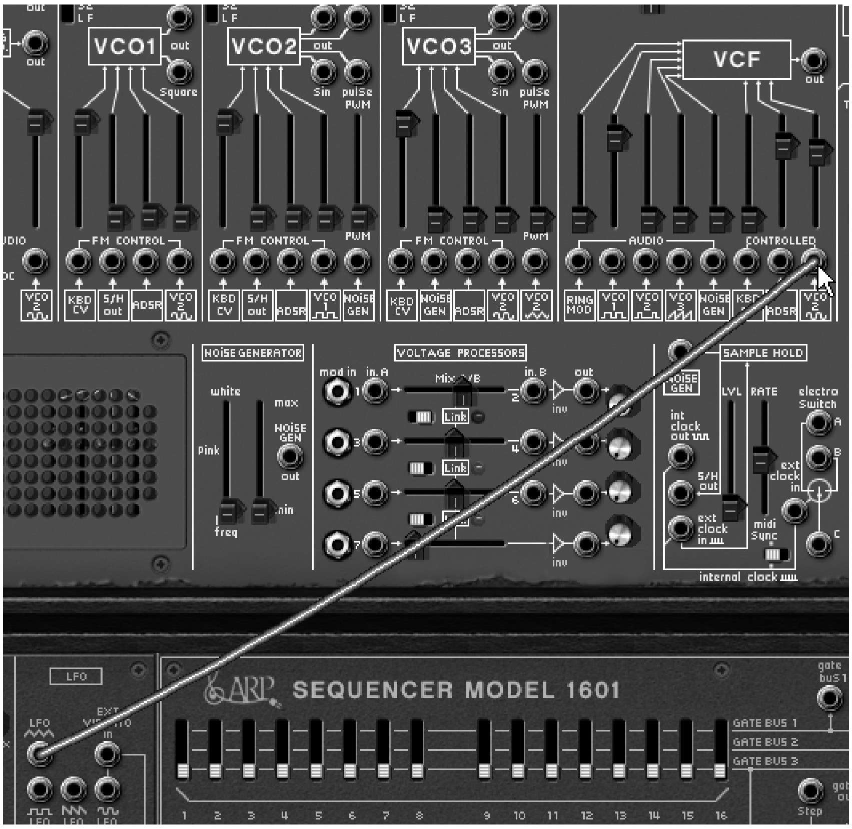
Thanks to the LFO module situated on the "Keyboard control" module of the "All" mode, you can keep the third oscillator as base sound and obtain an additional source of modulation for one of the 13 available destinations. It is also possible to synchronize the clock speed of the LFO to that of the MIDI sequencer by clicking on the "MIDI sync" interrupter.

Apply two additional types of modulation to the preset "1_Osc":

- For example: the LFO is "pre-cabled" to obtain a vibrato (simultaneous frequency modulation) of the two oscillators. Simply raise the linear potentiometer "Vibrato Depth", situated on the LFO module, to create this effect.

Raise the "Vibrato Depth" potentiometer

- Another example would be to click on the LFO triangle output and direct the cable to the "VCO2 sin" modulation input of the filter module. Raise the potentiometer above it. Lower the cut-off frequency to hear the result more clearly. The brightness of the sound will vary in a cyclic fashion, to the rhythm of the LFO.

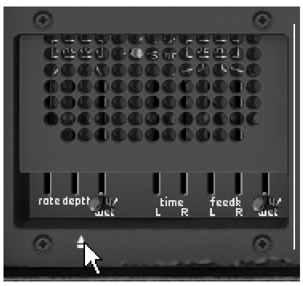


Modulate the filter frequency ("Cutoff Frequency") with the LFO

3.5 <u>The effects</u>

The effects section lets you add a Stereo Delay and Chorus to your sound on top of the reverberation which is already present in the original instrument. The two effects can be found in the place of the left speaker grid on the synthesizer.

To open it, click on the "open Effects" button under this grid.

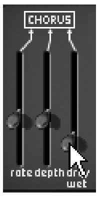


Open the effects grid

3.5.1 Chorus

Chorus is used to copy your sound, and slightly detune the copy, to give it more depth and the Chorus “ON/OFF” button in the effects section, on the right of the toolbar.

- Set the Chorus “Dry/Wet” potentiometer to balance the “raw” sound and the one returning from the effect.
- Next turn the Chorus “Rate” potentiometer to set the speed of the oscillations.
- Finally set the depth of the chorus using the “Depth” potentiometer.



the chorus effect settings

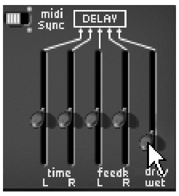
3.5.2 Delay

Delay brings a stereo echo effect to bring more space to your sound.

It possesses independent settings for the speed and number of repetitions for the left and right sides. It is also possible to create a large number of rhythmic combinations between the repetitions. The delay speed can also be synchronized with the MIDI tempo.

Let’s keep the preset “JMB_Simple1” and see how to use the effects on this sound:

- Activate the “Delay” button on the toolbar. The effect becomes active.
- Set the Delay “Dry/Wet” potentiometer so as to balance the “raw” sound with the one coming from the delay.
- Next turn the two Delay “Speed” potentiometers to set the rate of echo repetitions for the right side (Time Right) and left side (Time Left).
- It is also possible to set the number of repetitions for each side (“Feedb. Right” and “Feedb. Left”).



The Delay effect settings

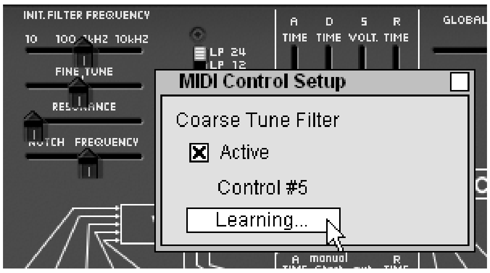
3.6 Real-time controllers and MIDI assign

Like its brilliant ancestor, the ARP2600 V is particularly adapted to real-time playing. One of the main improvements when compared to the original is that we can assign any potentiometer on the ARP2600 V to an external MIDI controller.

Here’s an example of assigning:

- Click on the “Initial Cutoff frequency” potentiometer of the filter while keeping the Ctrl button held down. A MIDI assign dialog appears.
- Click on “Learn” and then move the MIDI controller of your choice (the modulation wheel for example). The ARP2600 V potentiometer will start to move at the same time.

- You can then record movements from your MIDI controller on your MIDI sequencer or simply play live.



MIDI assign for "Cutoff frequency" potentiometer

Attention ! The MIDI assign settings are only saved when you quit the ARP2600 V application – be it standalone or plug-in.

To save the settings that we have seen in this chapter, click on the "Save" button in the toolbar.

4 The interface

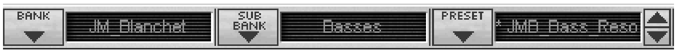
4.1 Using the presets

The presets memorize the ARP2600 V sounds. A preset contains all of the inter-module connections and the different controller information necessary for the recreation of an identical sound. In the ARP2600 V, presets are classed in “banks” and “sub-banks”. Each bank contains a certain number of sub-banks, which determine a type of sound: sub-bank “basses”, sub-bank “sound effects”, etc. Each sub-bank contains a certain number of presets.

The ARP2600 V comes with several “factory” sound banks. It is possible to create new “user” sound banks, each containing an arbitrary number of sub-banks and presets. For security, the “factory” settings are not directly modifiable. It is, however, possible to modify a sound based on a factory preset and to record it to a “user” bank.

4.1.1 Choice of bank, sub-bank, preset

The banks, sub-banks and presets being currently used are always displayed in the synthesizer tool-bar.



display of bank, sub-bank, and preset being used

To choose a preset in the current sub-bank, click on the button on the left of the current preset, a drop-down menu appears with a list of presets from the same sub-bank. You can choose another preset in the menu by selecting the corresponding line. Once the preset has been chosen, you can play the new sound from your MIDI keyboard or sequencer.

choice of preset in the same sub-bank

To choose a preset in the same main bank, but in a different sub-bank, click on the button on the left of the current sub-bank, a drop-down menu will appear with the list of sub-banks contained in the same main bank. Each sub-bank in the menu allows you to open a sub-menu containing its pre-sets. A click on a preset lets you directly choose a preset in the new sub-bank.

To choose a preset in another main bank, click on the button . A drop-down menu appears with the choice of the main banks that are available, and the sub-lists corresponding to the sub-banks defined in each main bank and the presets contained in each sub-bank. You can now freely choose a preset by clicking on its name.

Once a preset has been changed (modification of a controller or connection), an asterisk appears next to its name in the tool bar.

In the “BANK” dropdown menu, the “All” option allows you to open a sub-list with all of the sub-banks available in all of the banks. This gives you access directly to all of the presets of a given type, for example all of the basses, no matter which bank they are in.

This function is particularly useful to quickly see all of the presets of the same type.

the “All” option

4.1.2 Creation of a bank, sub-bank, preset

To create a new bank of sounds, click on the button on the left of the current bank. In the drop-down menu, select “New bank...” to create a new bank of sounds. You can then change the name of this bank by clicking on its name in the toolbar and typing the new name.

To create a new sub-bank, again just click on the button on the left of the current sub-bank, and select “New sub bank...”. You can also change the name of the new sub-bank.

Finally, to create a new preset, click on the button on the left of the name of the current preset and select “New preset...”. The new preset is created using the current ARP2600 V settings (controllers and connections). You can then work on the settings of the sound, and save it by clicking on the save button (see the next paragraph). You can also change the new preset name by clicking on its name.

4.1.3 Saving a user preset

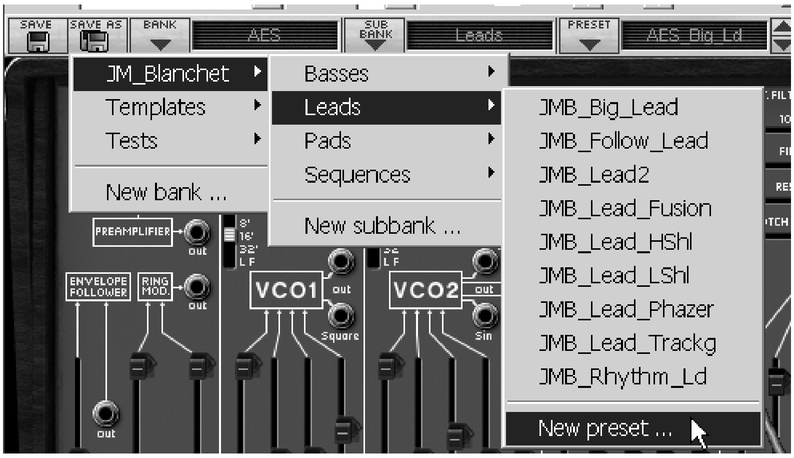
To save your current settings under the current preset, click on the “Save” button on the ARP2600 V toolbar.

"Save" button on the toolbar

If you want to save your preset under a different preset name, click on the "Save As" button in the toolbar. A drop-down menu will appear allowing the choice of either an existing preset (in this case, the preset contents will be replaced by the current setting), or to save your preset as a new preset (in this case, click on "New Preset…" in the sub-bank of your choice).

"Save As" menu on the toolbar

- When you are working on a factory preset, which cannot be erased, clicking on the "Save" button will not replace the current factory setting, but will automatically open the "Save As" function to save the current setting as a user preset. Click on the "New bank" option. The three LED displays indicate "New": you can click on each of these displays to give it a name or save your setting as a new preset (in this case, click on "New preset..." in the sub-bank of your choice).



the "New preset..." option

4.1.4 Importation / Exportation of a preset bank

It is possible to import new preset banks created for the ARP2600 V. To import a new bank of presets, click on the preset bank import button in the toolbar:

Preset bank import button on the toolbar

When you click on this button, a dialog appears allowing the choice of ARP2600 V preset bank files (.AMB file type on PC, AMpB file type on Mac). Choose the file that you want to import, and click on "Open". The new preset bank will automatically appear in the available banks.

The ARP2600 V also offers the option to export your own sound banks to save them, use them on another machine, or share them with other users. It is possible to export a preset, a sub-bank, or a complete bank. To export a bank, sub-bank, or current preset, click on the export preset bank button on the toolbar:

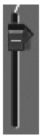
Current preset bank export button in the toolbar

Select the type of export that you wish to perform (bank, sub-bank or preset) from the list and a window will appear prompting you to choose a destination folder and a name for the file you are about to export.

4.2 Use of controllers

4.2.1 Vertical linear potentiometers

The ARP 2600V mainly uses linear potentiometers. To change the value of a vertical potentiometer, click on it and move it vertically to the desired value.



linear potentiometer

4.2.2 Horizontal linear potentiometers

To move these potentiometers, click on it and slide to the right or left to reach the desired value.

horizontal potentiometer

4.2.3 Knobs

You can find some knobs among the global settings, by exemple.

The default mode of control for knobs with the mouse is the linear mode: the knob can be set only by vertically moving the mouse. It is possible to obtain a higher precision by right clicking or Shift+Click on the knob concerned.

a knob

In circular mode, click on the knob and turn around it in order to change the value of the controller. The circular mode gives high precision in the manipulation of controls: The further the mouse goes from the knob, the higher the precision of the setting.

The linear mode is often more simple to use than rotation. But at the same time it cam be less precise (the precision is limited by the number of vertical pixels on the screen from which the mouse movements are evaluated). It is possible to attain a higher level of precision by right-clicking, or shift-click, on the potentiometer you want to control.

4.2.4 The selectors:

The selectors (the filter mode selector for example) are manipulated like vertical potentiometers by dragging with the mouse).

The filter "Modes" selector

4.2.5 Switches

The ARP2600 V has several types of switches. Simply click on these switches to change their state.

MIDI sync on/off switch

4.2.6 Pitch Bend

The pitch bend knob is used to change the pitch of the oscillators. Just click on the knob and move the mouse up or down to change the pitch of your sound. The knob goes back to the center position when the mouse is released.

The pitch bend knob

4.3 <u>Using cables</u>

The connection of different modules (Oscillators, Filter, VCA...) to one another broadens the possibilities for sound design. The ARP2600 V owes its extraordinary possibilities for creation in a large part to the wide range of connections possible. On the original ARP 2600, all of the connections were done by two types of cables:

4.3.1 Audio and modulation connections

The *audio* connections can be used, for example, to route the audio output of a wave form from an oscillator to the audio input of the filter or VCA mixer. In the same manner, the *modulation* connections allow you to route the output of a LFO or envelope generator to an oscillator PWM or the VCA modulation input. These *audio* and *modulation* signals are perfectly compatible with each other, the only difference being that the *audio* signals are "audible" of you connect them directly to the VCA, while the *modulation* signals are generally not audible (as the frequency is too low for the human ear). *Modulation* signals are generally used to program "slow" variations on certain synthesis settings, like the filter cut-off frequency for example.

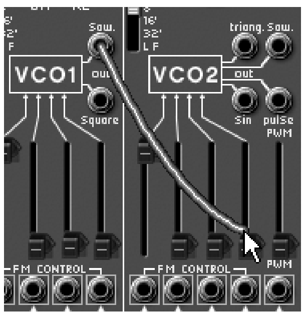
- The audio output and input connectors, and the modulation out connectors are represented differently in the graphical interface:

Audio input connector

- Additional modulation input connectors can be found on the tracking module located under the grid of the right loud speaker. They are graphically distinguished from the other connectors as they include an additional function: the modulation level setting, described later on in this chapter:

Modulation input connector

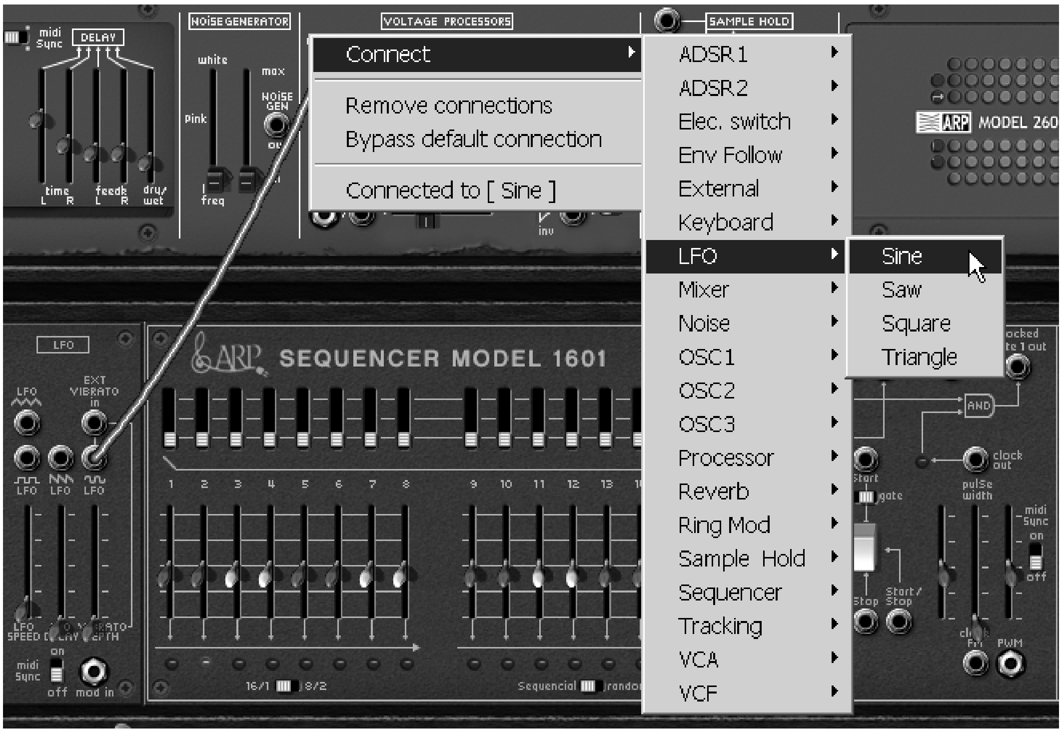
To connect the output of a module to the input of another module, click on the output and, while holding the mouse button depressed, drag the mouse to the input connector of the second module. When you pass over a connector on which the current cable can be placed, the connector will light up. In this case, when you release the mouse button, the connection will be created between the selected input and output.



Creation of a new connection by drag and drop

Another way is to right click (on Mac: click while holding down the shift key) on the input or output of a module. In this case, a menu appears with a list of possible connection points for this input/output. It can happen that the input/output will already be connected to certain modules; in this case the connection points already in use will appear in this menu. Simply choose a connection point in the “Connect to” menu to create a connection. It is also possible to delete the curent connection by choosing the “Remove connection” option.

Remark: Each modulation input can only receive one connection, but each output connector can be connected to any number of inputs. This allows us to use one modulation signal on several synthesis settings.



Right click for input/output connection menu (or Shift + click)

4.3.2 Modifying a connection

To disconnect a cable from an input connector, and reconnect it to another, click on the end of the cable and hold the mouse button down. You can now use the mouse to drag the cable to another input and release the button.

To delete a connection, you can right click and use the menu (or Shift + click). It is also possible to click on the cable to be removed to select it. The cable will appear in a lighter color to indicate that it is selected. Then simply press the “DEL” key to delete the connection.

Remark: You will have trouble selecting a cable if the “Move away cables” option is active !! In this mode, the cables automatically avoid the mouse pointer. To avoid this, deactivate this option before trying to select a cable (see. 1.4.4.3)

that it is selected. Then simply press the “DEL” key to delete the connection.

4.3.3 Setting the level of modulation

As explained above, the modulation input connectors have an interesting property: they allow the setting of the level of modulation (from –100% to +100%) directly at connector level, and thus avoid the need to use a VCA which would normally be necessary to set the amplitude of the modulation signal. When the modulation input connector is connected, click on one of the edges of the jack's hex nut and drag the mouse up or down to change the depth of modulation:

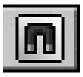
Setting the quantity of modulation

By right clicking (or Shift + click) instead of the left click, you will obtain a more precise setting.

Attention: If you click on the centre of the connector, you will select the cable end to change the connection. To access the modulation setting, make sure you click on the nut. In the same way, if you right click (or Shift + click) on the centre of the connector, you will reach the connection menu instead of the precise modulation quantity setting.

4.3.4 Separate the cables

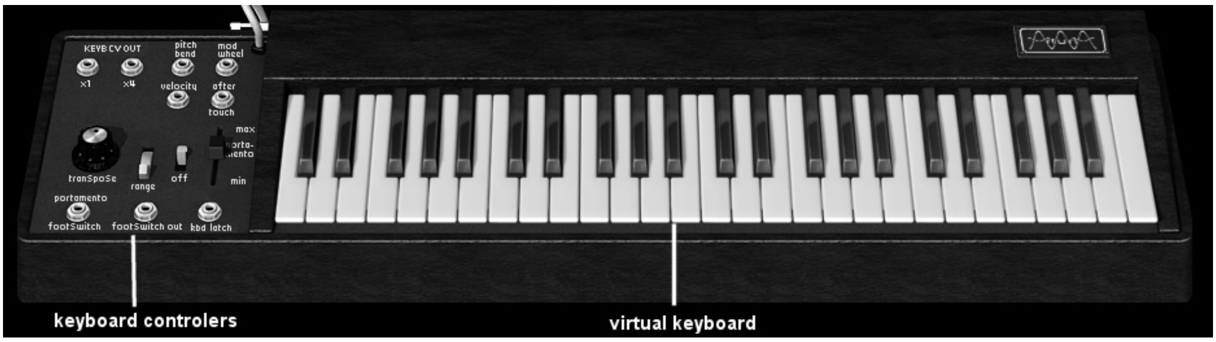
Visualizing the existing connections between different modules is very useful when creating a patch on the ARP2600 V. At the same time, the cables can sometimes hide access to some of the settings available on the modules. So as not to be hindered in your manipulations, it can be useful to activate the "Move away cables" mode. In this mode, the cables will automatically avoid the mouse pointer, leaving you with a clearer view to checking or modifying potentiometer values. Don't forget to deactivate this mode when you want to select a cable with the mouse!



Deactivate the "separate the cables" mode

4.3.5 Virtual keyboard

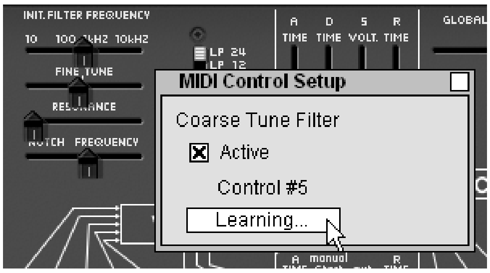
The keyboard lets you listen to the synthesizer sounds without the need for an external master MIDI keyboard, and without programming a melody in the sequencer. Just click on a virtual key to hear the corresponding sound.



4.3.6 MIDI control

Most of the knobs, sliders and switches on the ARP2600 V can be manipulated with external MIDI controllers. Before anything else, make sure that the MIDI device that you wish to use is correctly connected to the computer, and that the sequencer or the ARP2600 V application is correctly configured to receive MIDI events coming from the device.

Every instance of the ARP2600 V receives MIDI events transmitted on a given channel. This reception channel is defined in a global manner for the synthesizer, either in your sequencer, or in the stand-alone ARP2600 V application. On the reception channel, the ARP2600 V can receive up to 120 different MIDI controls. It is possible to choose a reception control for each knob. For this, click on the knob that you wish to control while holding down the Control key. A configuration window appears and will allow you to choose a MIDI control number. You can also click on the “Learn” button and move one of your physical MIDI controllers. In this case, the control number will be detected and configured automatically. To deactivate the MIDI control of a knob, simply uncheck the “Active” option in the MIDI control window.



MIDI configuration of a knob

5 The modules

The ARP2600 V can be separated into 3 parts: from top to bottom, the first is a cabinet dedicated to sound programming and effects, the second concerns the ARP sequencer and the playing mode configuration interface with the keyboard and LFO, and finally the third contains the keyboard.

5.1 Sound programming cabinet

5.1.1 Description

The programming section groups all modules which can be used to program sounds. It is also in this screen where the different patches needed in the programming of a sound are made.

It is sometimes necessary to connect a module in the programming section to a module located in the sequencer section.

The sound programming section contains:

3 oscillators, which can also be used as sources of modulation. (VCO)

1 multimode filter. (VCF)

1 amplifier (VCA)

2 envelopes dedicated to modulations

1 noise generator

1 ring modulator

1 envelope follower

1 sample and hold

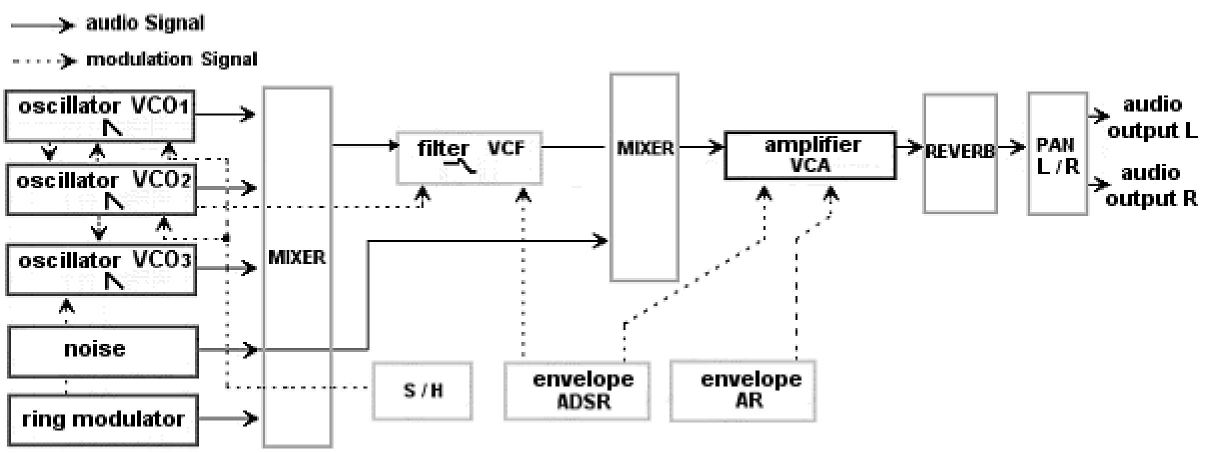
2 mixers (on the filter and the VCA)

1 electronic switch

A tracking generator module

5 mixer / lag module

2 effects (chorus / delay)



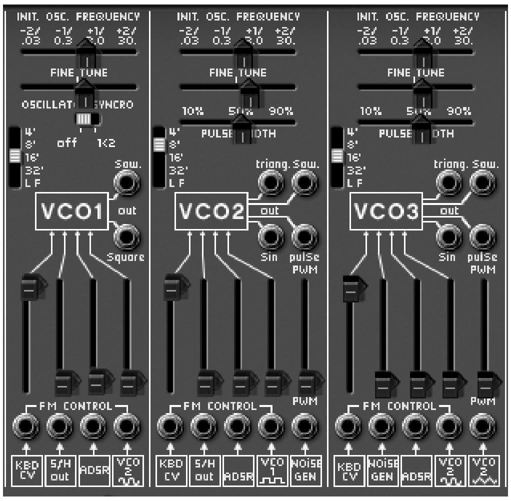
5.1.2 The Oscillators (Oscillators - VCO)

There are three oscillators in the ARP2600.

The oscillators permit the management of the base frequency and tone of the ARP 2600. It also manages the impulse width of the wave forms. These changes can either be made with linear potentiometers, or thanks to modulation inputs which can be connected to the output of any module (envelope, low frequency oscillator - LFO -, modulation wheel…).

The oscillators can also be tuned and modulated separately with the potentiometers and a range selector. These oscillators supply up to 4 wave forms that can be used simultaneously.

These 3 oscillators, tuned separately and providing mixed wave forms, quickly give us very rich sounds. The tone can also be easily modified with modulation inputs.



The 3 oscillators settings

5.1.2.1 Oscillator 1

Range : General tuning of the oscillator by octaves. Up or down 4 octaves and low frequencies

Frequency : Tuning by semi-tone (Initial Oscillator Frequency). Up or down 2 octaves
Fine tuning : Fine tune up or down by up to a semi-tone

Audio outputs : Output sawtooth & square wave form connection jacks

Sync : Synchronization of oscillator1 with 2, 3, or 2 and 3

FM Inputs : Frequency modulation input connection jacks

- Key follow : pre-cabled for the manual setting of the (KYBD CV)
- Sample and hold : pre-cabled for the manual setting of the modulation by sample and hold (S/H out)
- ADSR Env : pre-cabled for the manual setting of the modulation by ADSR envelope ADSR
- Oscillator 2 Sin : pre-cabled for the manual setting of the modulation by sine wave form of oscillator 2 (VCO2 Sin)

5.1.2.2 Oscillator 2

Range : General tuning of the oscillator by octaves. Up or down 4 octaves and low frequencies

Frequency : Tuning by semi-tone (Initial Oscillator Frequency). Up or down 2 octaves

Fine tuning : Fine tune up or down by up to a semi-tone

Audio outputs : Connection jacks for 4 wave form outputs: triangle, sine, saw-tooth and square

Impulse width : Impulse width for “Sawtooth”, “Square”, “Triangle” wave forms

FM Inputs : connection jack for frequency modulation inputs

- Key follow : pre-cabled for manual setting of the key follow (KYBD CV)
- ADSR Env : pre-cabled for the manual setting of modulation by ADSR envelope
- Sample and hold : pre-cabled for the manual setting of modulation by sample and hold (S/H out)
- Square oscillator 1 : pre-cabled for the manual setting of modulation by the square wave form of oscillator1 (VCO1 square)
- Noise : pre-cabled for the manual setting of modulation of the impulse width of the square by the noise (PWM _ noise out)

5.1.2.3 Oscillator 3

Range : General tuning of the oscillator by octaves. Up or down 4 octaves and low frequencies

Frequency : Tuning by semi-tone (Initial Oscillator Frequency). Up or down 2 octaves
Fine tuning : Fine tune on up or down by up to a semi-tone

Audio output : Audio output connection jacks for the sawtooth and square wave forms

Impulse width : Impulse width of the “Square” signal

FM Inputs : Frequency modulation input connection jacks:

- Key follow (KYBD CV) : pre-cabled for the manual setting of the key follow
- Noise(Noise) : pre-cabled for the manual setting of the modulation by the noise
- ADSR Env : pre-cabled for the manual setting of the modulation by ADSR envelope
- Sine oscillator 2 (VCO1 sin) : pre-cabled for the manual setting of the modulation by the sine wave form of oscillator 2
- triangle oscillator2 : pre-cabled for the manual setting of the modulation of the impulse width of the square by the oscillator2 triangle (Pulse width modulation _ VCO2 Triangle)

- The general tuning of the 3 oscillators is done with the “VCO Initial frequency” potentiometer by +- one octave per semi-tone.
- For fine tuning, set the Fine tune potentiometer to +/- one semi-tone.
- Depending on the position of the “Range” selector switch, the range is of +/- 4 octaves. It is also possible to set it to low frequency position (LF – for low frequency). You will no longer hear any sound, but can now use it as LFO source of modulation.
- The impulse width applied to the “sawtooth”, “triangle” and “square” wave forms of oscillators 2 and 3, can be modified with the “pulse width” potentiometer.

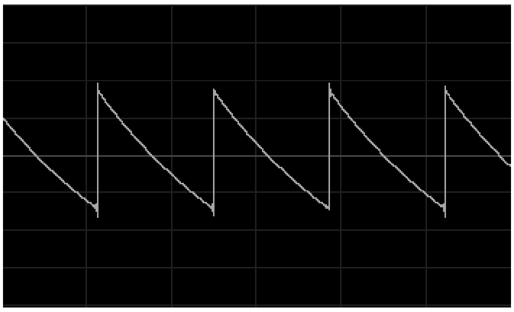
Oscillator 1 possesses two audio outputs for the sawtooth and square wave forms. These can be used simultaneously. The second and third oscillators possess four outputs for the sawtooth, sine, triangle and square.

The frequency modulation inputs (FM Control) and Pulse Width Modulation allow the control of these parameters thanks to the outputs of other modules (an envelope, a LFO for example) They are all pre-cabled to a defined module in advance by the developer of the ARP 2600V. This is to simplify the use of the synthesizer. For example: the frequency of oscillator 1 can be modulated, from the left to the right, by the key follow, the sample and hold, the ADSR envelope and the oscillator 2 sine. In low frequency position (LF), the oscillators perform modulation using the lower CPU power compared to the other positions.

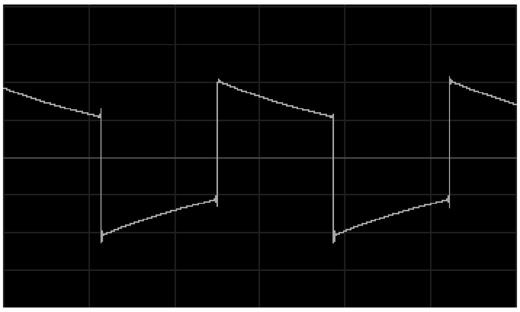
- To set the modulation rate, use the linear potentiometer situated above the corresponding jack.
- You can also connect another source of modulation to each input. This considerably widens the possibilities for sound creation.

For a conventional key follow setting (in relation to the scale) place the potentiometer completely to the top.

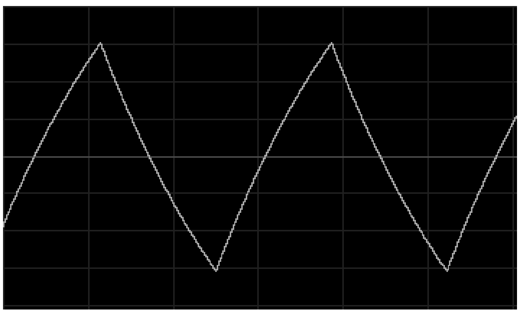
Here are graphical representations of the different wave forms used by ARP 2600V oscillators:



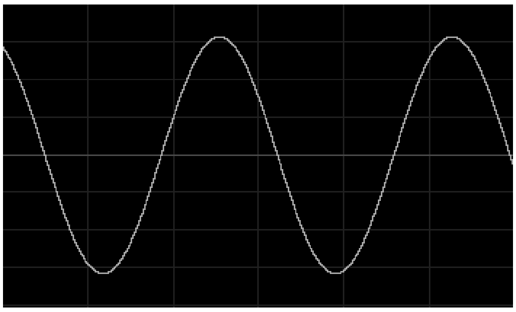
Sawtooth



Square



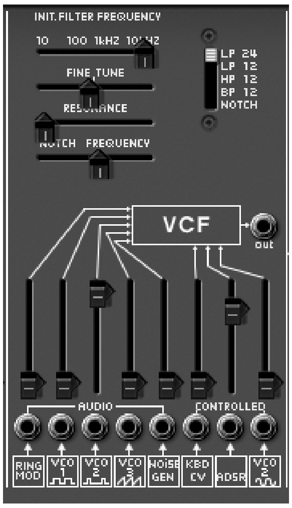
Triangle



Sine

5.1.3 The Filter (Filter - VCF)

The ARP2600 V possesses a multimode filter module (the original had only one resonant low pass mode). It is possible to choose a filter type among the five offered: a low pass 24 dB (identical to that found on the ARP2600), a low pass, a high pass, a band-pass and a notch 12 dB of the same kind as the one found on the ARP 2500 modulars. The change of type is done by setting the selector situated on the right of the filter module.



The filter settings

As with the oscillators, the filter possesses audio connections (a mixer) and internal modulation inputs allowing the simplification of its use.

Frequency (Initial Filter Frequency) : Sets the filter cut-off frequency, tuned between 10 Hz and 10 KHz

Fine tuning (Fine tune) : Fine tuning of the filter cut-off frequency

Resonance : Sets the filter resonance

Notch frequency / fc (Notch Frequency/ fc) : Sets the frequency of the notch divided by the filter cut-off frequency

Filter type selector (Types) : Type of filter (LP 2600 and 2500, HP, BP and notch)

Audio output (Output) : Filter audio output connection jacks

Audio input (Audio) : Filter input connection jacks (initially connected to the ring modulator, to the oscillators1 and 2 square, oscillator3 sawtooth and noise)

Modulation inputs (Control) : Filter frequency modulation input connection jacks.

- Key follow (KYBD CV) : Pre-cabled for the manual setting of the key follow
- Oscillator3 sine (VCO3 Sin) :pre-cabled for the manual setting of the modulation by sine wave form of oscillator2
- ADSR Env : re-cabled for the manual setting of the modulation by ADSR envelope.

For a conventional key follow setting (in relation to the scale) place the potentiometer completely to the top.

It possesses a cut-off frequency setting and a resonance setting.

A setting separated from the notch filter frequency (Notch frequency) divided by the initial filter cut-off frequency has been added. This very particular parameter (present on the module of the ARP2500) transforms the notch filter to a low shell or high shell filter.

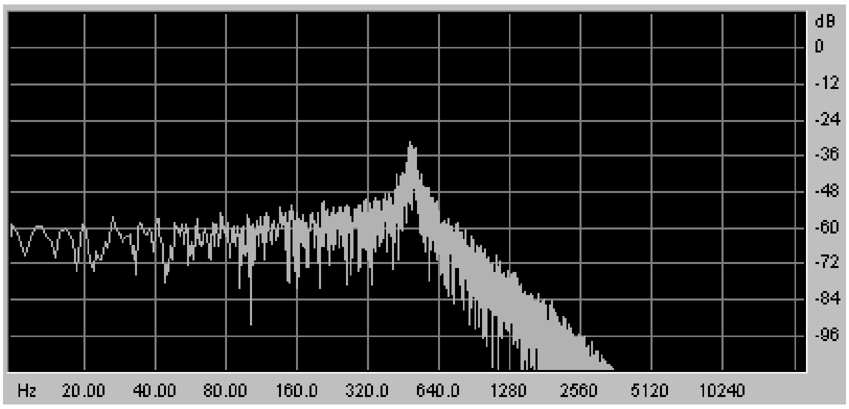
As for all other modulation inputs, once connected, its amplitude is set by raising the linear potentiometer. With a right click, we obtain a higher level of precision. Receiving a modulation coming directly from the output of a generator (envelope, oscillator, sequencer), the maximum amplitude of modulation of +/- 9 octaves.

Only the cut-off frequency can be dynamically modulated by one of the 3 modulation inputs.

5.1.3.1 The filter types:

- The low pass 24dB / oct Low pass (LP 24)

The low pass 24dB filter is typical of the ARP2600. It eliminates the frequencies situated below the pivotal frequency (the cut-off frequency).



The low pass filter

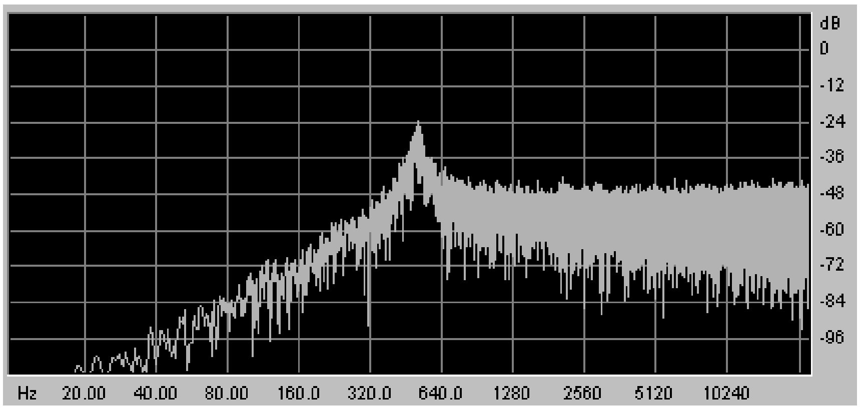
The four other filtering modes didn't exist on the original ARP2600 but existed on the ARP2500 modular systems. They all use a filtering slope at 12 dB/ octave. These modes were added to increase the possibilities for sound creation on the ARP 2600V.

- The low pass 12dB / octave (LP 12)

The low pass 12 dB filter is works in the same manner as the 24 dB/ oct. on the the ARP2600. It will just give you a slightly different result due to the fact that its filtering slope is not as fast.

- The high pass (HP 12)

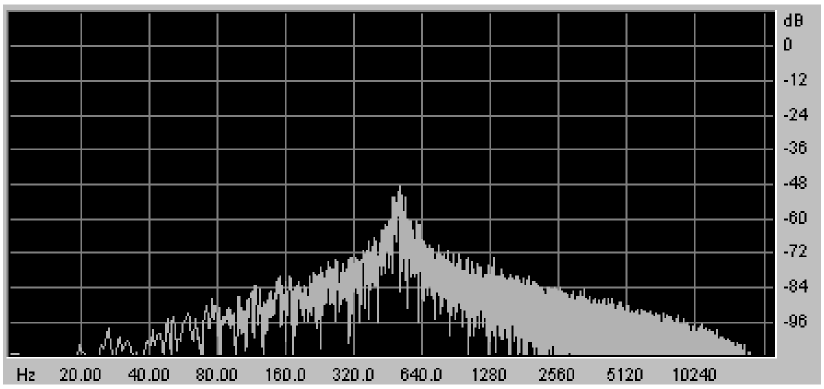
The high pass filter is the opposite of the low pass filter. It eliminates the frequencies above the cut-off frequency.



The high pass filter

- The band pass filter (BP 12)

The band pass filter is a combination of the low pass and high pass: It eliminates the frequencies on each side of the cut-off frequency.



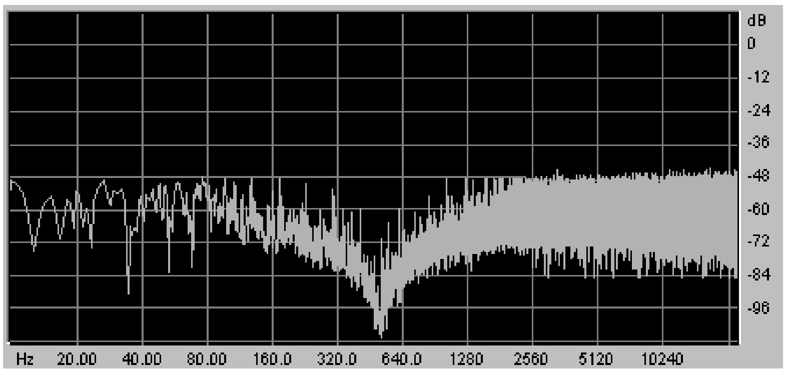
The band pass filter

- The notch filter (Notch)

The notch filter coupe is the opposite of the band pass filter. It eliminates the frequency band situated on both sides of the cut-off frequency.

The resonance can be used to accentuate the “hollow” of this frequency band.

Attention! if you raise the resonance too high, the result of the filtering will no longer be heard as the frequency band will be too big to allow efficient filtering.



The notch filter

5.1.4 The envelopes

Two in number, the modulation envelopes allow the evolution of the sound in relation to the time.

The first envelope (ADSR) possesses four temporal periods which are performed sequentially: the Attack time, Decay time, Sustain voltage and the Release time. When the trigger input moves to active state (triggering of a note), the envelope performs the sequences “Attack” followed by “Decay” and remains in the “hold” state (Sustain) as long as the input trigger remains active. When it goes to the inactive state (release of the note), the envelope performs the “fall” sequence (Release).

ADSR and AR envelopes settings

5.1.4.1 ADSR

Attack (Attack time)	: Sets the attack time
Decay (Decay time)	: Sets the decay time
Hold (Sustain Voltage)	: Sets the level of the hold
Release (Release time)	: Sets the release time
Output (Output)	: Envelope output signal

5.1.4.2 AR

Attack (Attack time) : Sets the attack time

Release (Release time) : Sets the release time

5.1.4.3 Trigger modes

Trigg input (S/H gate) : Input connection for an external trigger signal (pre-cabled to the Sample and Hold clock)

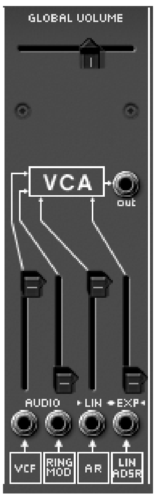
"Gate" output type (Gate) : Output connection for a "gate" type signal: for every note played on the keyboard, the signal remains active as long as the note is held down.

"Trigger" output type (trig) : Output connection for a "trigger" type signal: each note played by the keyboard presents a sustain time reduced to the minimum.

Switch "trigger by Sample and Hold clock" (S/H gate): Selection of the choice of envelope trigger - AR and ADSR – by the keyboard or Sample and Hold clock. This function reactivates the envelopes at every clock cycle.

5.1.5 Output amplifiers (Voltage Control Amplifier - VCA)

The amplifier is the last step in the conception of a preset. It sets the general volume of the sound.



the VCA settings

This module is very simple; it contains:

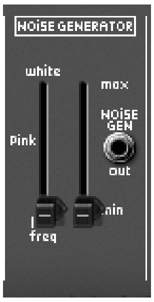
Gain (Initial Gain) : Sets the general synthesizer volume

Audio input (Audio VCF / ring mod) : Audio input connection jack (pre-cabled to the filter and ring mod audio outputs).

Modulation inputs (Control) : Modulation input connection jacks (pre-cabled on the AR and ADSR envelopes).

5.1.6 Noise Generator

The noise generator generates white noise or colored noise. It possesses a low pass filter to dampen high pitched frequencies.



the noise generator

Low pass frequency (white / Low Freq) : Sets the cut-off frequency of the low pass filter.

Noise volume : Sets the noise volume.

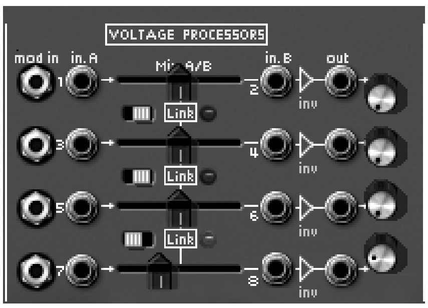
Noise generator output (Noise generator output) **:** Noise out connection jack.

5.1.7 Voltage processor (Mixer / inverter / lag generator)

The voltage processor allows you to mix up to 8 inputs (audio or modulation) to one (or several) outputs.

It also allows you to invert the input signal – a positive-going modulation (an envelope for example) will thus become negative.

A lag generator allows you to smooth an input signal. For example, the square signal from a LFO approximates a triangle the more we raise the potentiometer value, increasing the lag time.



the voltage processor

8 signal inputs (input) : Audio or modulation signal input connection jacks.

4 balance (mix) : Sets the balance between the two input signals

4 inverters (inverter) : Inverts the input signal

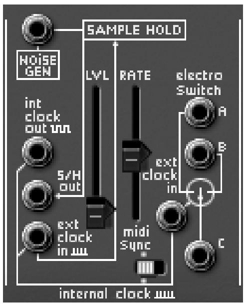
4 lag generators : Smoothes the input signal

4 signal outputs (output)	: Audio or modulation signal output connection jacks
4 “Mix” switches	: This switch allows you to merge one or several inputs pairs into the one above.
4 “volume” potentiometer	: potentiometer to set the volume of 2 input signals (A and B for example)

5.1.8 Sample and Hold generator

This module lets you sample the signal connected as input. The source can be external (source of trigger connected to the input) or pre-cabled to the noise generator. This module allows you to create random modulations by sampling the noise signal, for example.

You can also connect an external clock source to pilot the sample and hold speed (the wave form output of an oscillator for example).

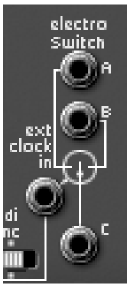


the sample and hold settings

Level (Level)	: Sets the Sample & Hold modulation level.
Rate (Rate)	: Sets the Sample & Hold clock speed.
External Input (“Noise gen”) (pre-cabled to the noise module)	: Audio or modulation external signal input connection jack.
Sample and Hold output	: Sample and Hold output connection jack
Internal clock output (Int Clock out)	: Sample and Hold internal clock output connection jack
External clock input (Ext Clock in)	: External clock input connection jack
MIDI Synchronization (MIDI sync)	: Selector switch for the synchronization of the clock to a MIDI sequencer.

5.1.9 The”electronic switch” module (Electronic switch)

The “Electronic switch” module allows you to alternate the two sources connected to inputs A and B depending on the speed of the clock connected (pre-cabled to the Sample and Hold clock) to create a composite source of modulation.



the electronic switch

An example:

- Connect the output of the square wave form of oscillator 1 to input A and the sine output of oscillator 2 to input B.
- Place these two oscillators to low frequency position (LF) so as to slow the oscillation speed. Set the frequency potentiometers for the two oscillators to 0.3 Hz.
- Connect output C of the interrupter to one of the filter modulation inputs.
- Set the Sample and Hold clock speed potentiometer towards the bottom to clearly hear the alternation between the two modulations.

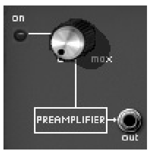
Inputs (A / B) : Audio or modulation signal input connection jacks.

Selector switch (C) : Selector switch C connection jacks between signals A and B

External clock input (Ext clock In) : External; clock connection jacks (pre-cabled to the Sample and Hold clock)

5.1.10 Envelope follower

The envelope follower allows the generation of a modulation from an external (or internal) source of audio signal. The (Pre amp.) volume parameter sets the fineness of the envelope follow. The lower it is, the more closely the variations of the input signal will be followed.



the preamplifier

The envelope follower

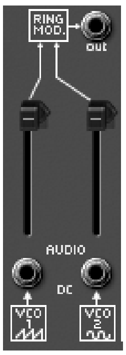
An example:

- Try for example to insert an audio signal coming from a drum loop sample to the input of the envelope follower.
- Connect the output of the envelope follower to one of the filter modulation inputs.
- Increase the modulation rate for the amp (VCA).
- Do the same for the envelope follow level. The VCA volume will modulate the volume of the VCA with the envelope of the drum loop audio sample.

Audio input (Pre-Amp)	: Envelope follower audio input connection jack
Output (Output)	: output jack fo the audio signal
Level (Level)	: Sets the amount level of the envelope follower
Gate out (gate out)	: Trigger signal output connection jacks

5.1.11 Ring modulator

The ring modulator is a module which allows you to multiply two signals in order to generate non-harmonic frequency components. It is useful for creating metallic sounds.

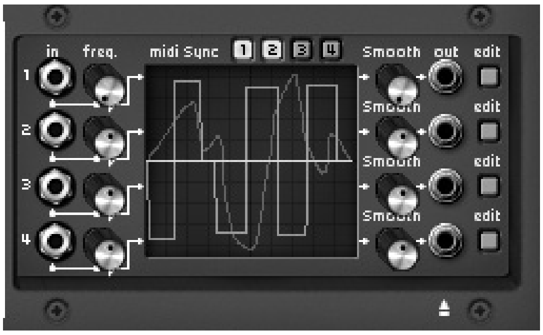


the ring modulator

2 Audio inputs (VCO1 Saw / VCO2 Sin)	: Audio input connection jacks (pre-cabled to the oscillator1 sawtooth and oscillator 2 sine outputs).
Output (Output)	: Ring modulator output connection jacks

5.1.12 "Tracking" generator

This very original module was added to modify the course of a modulation thanks to four curves which can be edited in real-time by the user. It can also be used as source of modulation to create envelope forms or complex LFO waves.



the tracking generator's main interface

- Main interface:

4 audio inputs : Audio input signal connection jacks.

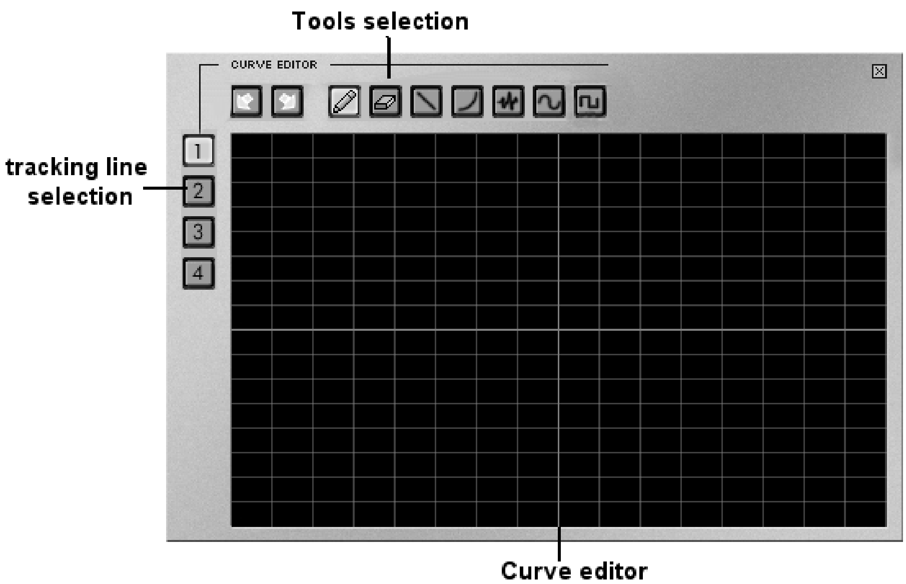
4 modulation inputs : modulation input signal connection jacks

4 audio outputs : Audio signal input connection jacks.

4 "Smooth" : Sets the smoothness of the tracking curve.

4 Edit : Button for opening the tracking curve edit mode.

- Edit interface



Curve editing screen : Screen for real-time curve editing.

Drawing tool : Tool for drawing a freehand curve.

Line tool : Tool for drawing a straight line.

Curve tool : Tool for drawing an exponential curve.

Sine tool : Tool for drawing a sinusoid.

Square tool : Tool for drawing a square signal.

Noise tool : Tool for adding noise to an existing signal.

Eraser tool : Tool for erasing an existing signal.

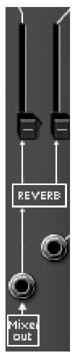
Undo / Redo : To undo or redo a drawn signal.

A few tips for using it:

- To create an exponential curve: Firstly click on the editing screen to place the starting curve for the curve.
- Drag (without releasing) to draw the curve.
- Click and drag up or down to set the amplitude.
- Click once again to validate this curve.

5.1.13 Reverberation

This module lets us add reverberation to the sound.



the reverb module

Right effect level (Level) : Sets the output level for the right reverberation

Left effect level (Level) : Sets the output level for the left reverberation

Right output (Output R) : Right output connection jack (with or without reverberation)

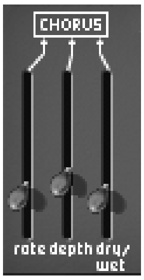
Left output (Output L) : Left output connection jack (with or without reverberation)

Dry input right (Dry input right) : Dry right input connection jack

Dry input left (Dry input Left) : Dry left input connection jack

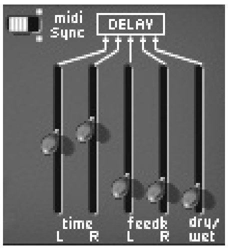
5.1.14 Chorus and delay effects

Chorus has three potentiometers, “Rate” “Depth” and “Dry/Wet” which allow us to respectively set the speed, the depth and the relation between the original and modified signals.



The chorus effect

Delay has two potentiometers, “Time Left” and “Time Right”, respectively for setting the time for the left channel, and the right channel. The two potentiometers, “FeedB Left” and “FeedB Right” respectively set the channel return gain for left and right channels. Finally the “Dry/Wet” potentiometer sets the ratio between the original and modified signals.

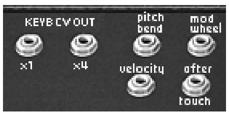


The delay effect

The “Midi Sync” selector switch allows us to synchronize the return time for the delay to the tempo of the host application.

5.1.15 Control voltages (CV control)

These modulation outputs allow us to control the synthesizer parameters with the real-time controllers of your MIDI keyboard.



the CV controls

Pitch bend wheel (Pitch bend) : Output connection jack for the control of modulation of three oscillators with the pitch bend wheel.

Modulation wheel (Mod Wheel) : Output connection jack for modulation control with the help of the modulation wheel.

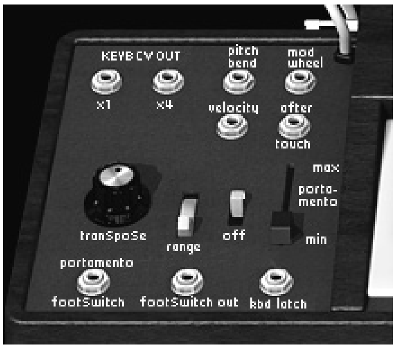
Velocity (Velocity) : Output connection jack for velocity control.

After Touch : Output connection jack for After Touch control.

Key follow outputs x1 and x 4 (KYBD CV output) : Output connection jacks for key follow x 1 (tempered key follow) and x 4

5.2 Keyboard interface (Model 3620)

The keyboard interface contains all of the parameters necessary for playing on a keyboard: the monophonic or polyphonic playing modes, the portamento, the LFO (mainly used for vibrato), keyboard triggering modes, etc...



the keyboard interface

Portamento trigger input pedal (portamento footswitch) : Output connection jack for portamento trigger control with a pedal.

Sustain trigger input pedal (KBd Latch) : input connection jack for Hold trigger control with a pedal.

Play mode selector (Modes) : Selection of keyboard play mode - monophonic, unison, polyphonic.

Detune polyphonic voices (Unison detune) : Sets the detuning of the voices in “unison” mode.

Select trigger modes (Trigger modes) : Selects trigger mode for keyboard notes: legato / retrig

Pitch Bend : Simultaneously sets the frequency of the three oscillators.

Global tuning (Global tune) : Simultaneously set the global tuning of the three oscillators

5.3 Low frequency oscillator (LFO)

The use or a low frequency oscillator as source of modulation is quite common. It allows us to make the tone of a sound evolve gently or to create a vibrato or tremolo effect.

Even though three oscillators can be used at very low frequencies, there is a specific module for this, allowing you to dedicate the oscillators to generation in the audible domain.

the LFO

Low frequency oscillator speed (LFO speed) : Sets the clock speed for the low frequency oscillator.

Low frequency oscillator delay (LFO delay) : Sets the delay of the low frequency oscillator action.

Vibrato depth (vibrato depth) : Sets the depth of the vibrato.

MIDI synchronization (MIDI sync) : The “Midi Sync” selector switch lets you synchronize the low frequency oscillator clock with the MIDI tempo.

Triangle output (LFO triangle) : Audio output of the LFO triangle wave form.

Square output (LFO Square) : Audio output of the LFO square wave form.

Sawtooth output (LFO Saw) : Audio output of the LFO sawtooth wave form.

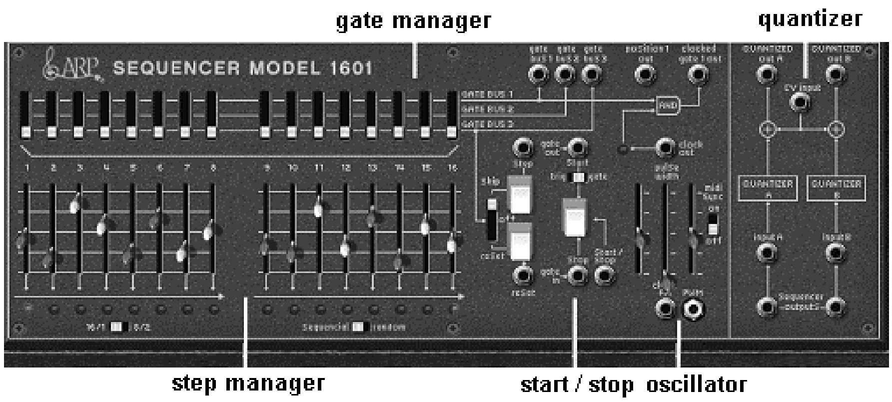
Sine output (LFO Sine) : Audio output of the LFO sinusoid wave form

External vibrato input (LFO External Vibrato in) : Audio input for external LFO source.

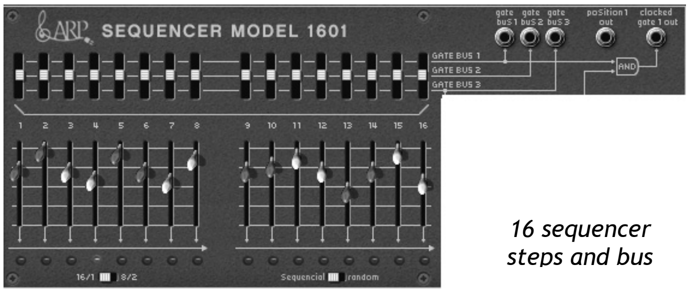
5.4 The ARP sequencer

This module was born of the original ARP sequencer model 1601 (18). It was one of the most widely used sequencers in the 70’s and early 80’s.

With this module, you will create melodic sequences or step by step variations applied to synthesizer parameters (a sequence line applied to the opening of the frequency of a filter can be very effective, for example).



- Sequence step management, setting and management of the 16 sequencer steps



16 sequencer steps and bus

2x 8 output levels (1 ... 16 faders) : Sets the modulation or tuning level for sequence steps.

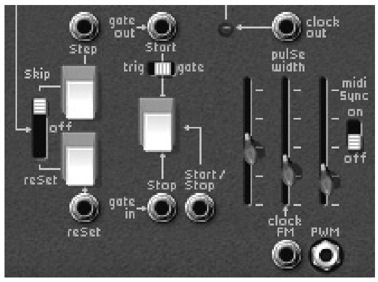
Bus 1, 2 and 3 (switches Gate bus 1, 2, 3) : Trigger selector switch for steps by bus 1, 2 or 3

Bus inputs1, 2 and 3 (inputs Gate bus 1,2,3) : Input jacks for bus 1, 2 or 3

Position : Position input jacks

Sequencer clock output(Clocked gate out) **:** Sequencer clock output jack

- Sequencer oscillator



the sequencer oscillator

Skip / return (Skip/Off/reset Steps) **:** Selector switch for skipping steps or returning to step 1

Skip steps (Step) : Button for manual choice of steps.

Return input (Reset) : Input jack for a forced return to step 1.

Start input (trig or gate) (jack Start) : Input jack to start the sequencer with a trig or gate signal.

Trig or Gate (trig / gate) : Selector switch for choice of trig or gate (starting the sequencer).

Start / stop (Start / Stop) : Button for manual starting / stopping of the sequencer.

Start / stop jack (Start / Stop jack) : Input jack for sequencer start / stop.

Start / stop pedal (Start / Stop pedal) : Pedal to start and stop the sequencer.

Clock frequency (Clock Freq) : Sets the sequencer clock speed

Clock frequency output (Clock out) : Input jack for the sequencer clock output.

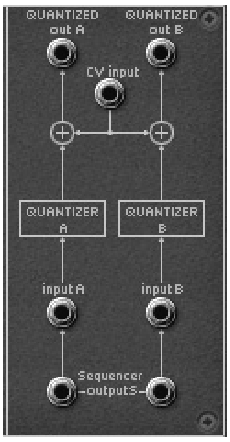
Clock FM (FM) : Sets the modulation rate for the clock frequency.

FM input (Clock FM) : Jack input for the modulation of the clock frequency (initially connected to the “gate” input)

Impulse modulation (Pulse width) : Sets the rate of modulation for the signal impulse width.

PW input (PWM) : Jack input for the modulation of the impulse width of the signal.

- Setting the sequencer input / outputs (quantizer)



the sequencer input / outputs

Quantized outputs A and B (Quantized outputs A/B) : Output jacks A and B for the quantization of the step signals in semi tones.

CV input (CV input) : Jack inputs manage the reference voltage for the sequencer (generally connected to CV keyboard control - Kybd CV output)

Quantized inputs A and B (Quantized inputs A/B) : Input jacks A and B for the signal quantization.

Non quantized outputs A and B (Sequencer outputs A/B) : Output jacks A and B non quantized (continuous values to +/- 4 octaves)

6 The basics of subtractive synthesis

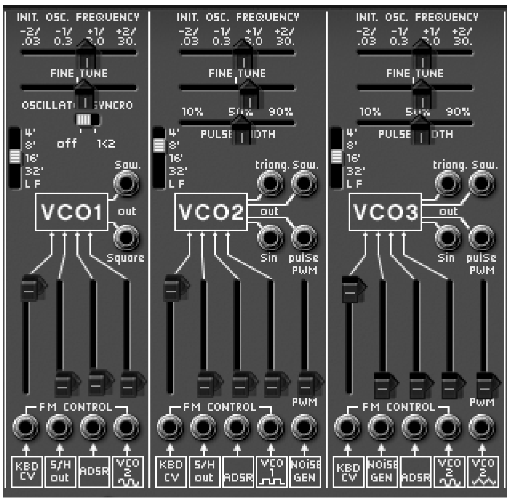
Of all forms of sound synthesis, subtractive synthesis is one of the oldest and still certainly one of the most employed today. It is this method that was developed toward the end of the 60's on analog synthesizers like the ARP, Moog, Oberheim, Yamaha, Buchla, Sequential Circuits (Prophet series), Roland, Korg (MS and PS series) and many others. This concept of synthesis is still used on most current digital synthesizers, complementing sample reading or wave tables, which progressively replaced the analog oscillators of the first synthesizers in the 80's. The ARP2600, or even your own ARP2600 V are among the best illustrations of the enormous possibilities of subtractive synthesis.

6.1 The three main elements

6.1.1 The oscillator or VCO

The oscillator (**Voltage Controlled Oscillator**) is the starting module (with the noise module which is often classed among the oscillators) for the creation of a sound on an analog system.

It will generate the initial sound signal. We can think of the oscillator like a violin string that once stroked or plucked, vibrates to create its sound.

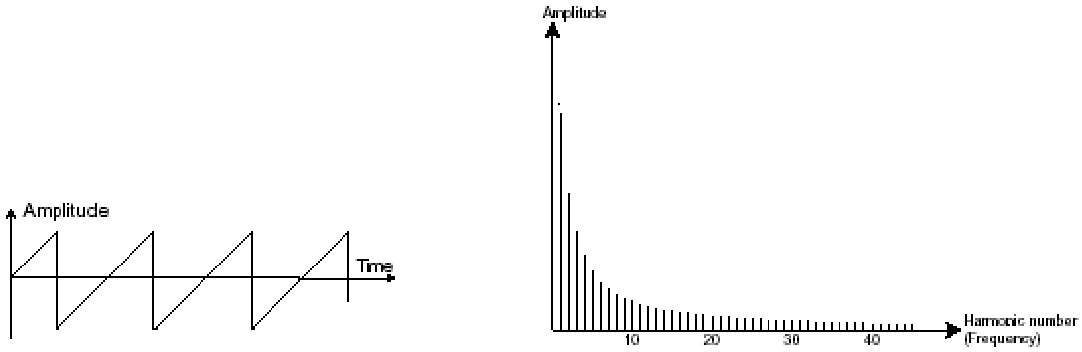


The oscillator settings on the ARP2600 V

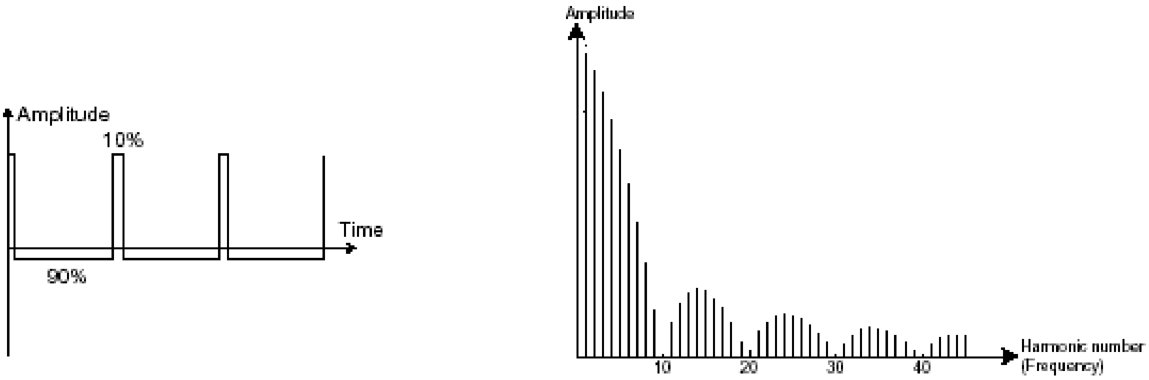
The main oscillator settings are:

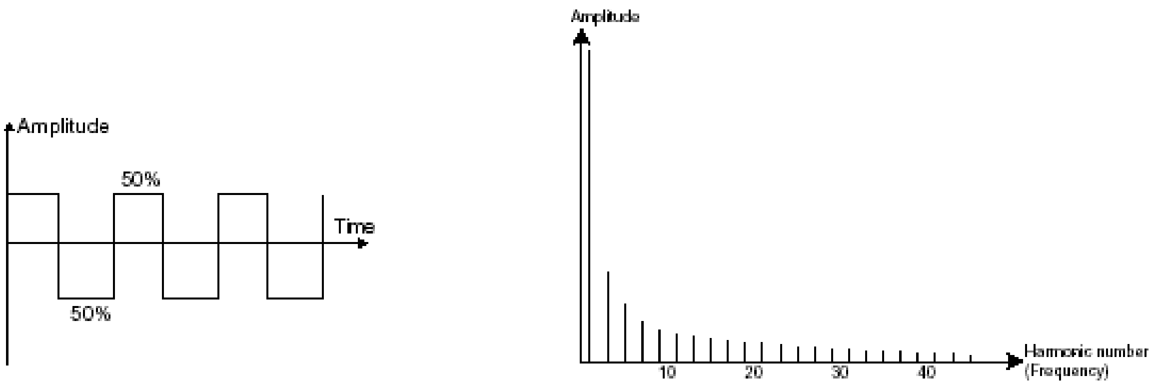
- **The pitch** is determined by the oscillation frequency. You can set the frequency of the oscillator with 2 controllers: first, the "RANGE" selector which determines the fundamental frequency – it is expressed in feet- : Low, 32,16,8,4,2 ; the highest number (32) brings the deepest tone, inversely, the smallest number (2) brings the highest tone. – Secondly, the detune setting ("FREQUENCY"*)* lets you tune the oscillator more precisely.
- **The waveform** which determines the harmonic richness of the audio signal. On the ARP2600 V, four waveforms are available:

  - The **saw tooth** presents the richest audio signal of the four waveforms (it contains all of the harmonics at decreasing volume levels in high frequencies). Its sound is ideal for brass sounds, percussive bass sounds or rich accompaniments.

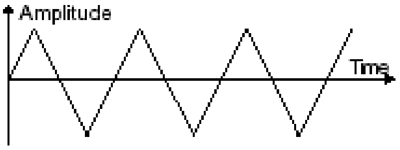


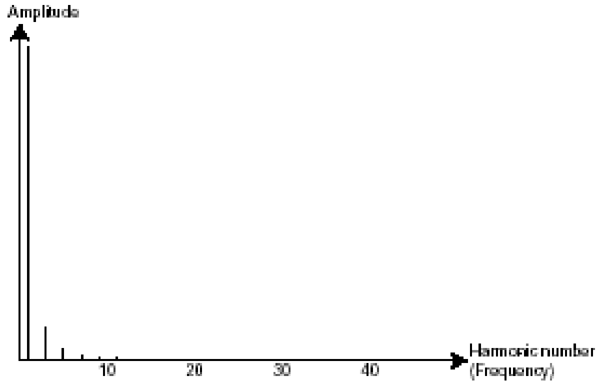
* The **square** possesses a more “hollow” sound that the saw tooth (it only contains odd harmonics) but none the less, its rich sound (notably in low frequencies) can be used for sub-bass sounds that will come out well in the mix (the square oscillator is often set an octave below that of the saw tooth), wood sounds (clarinet, if the square signal is a little filtered), etc.…





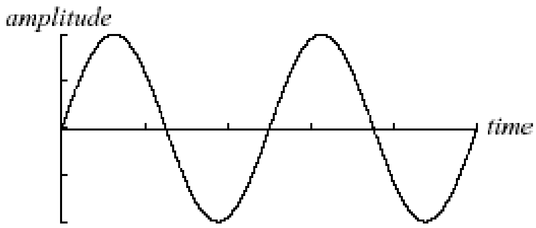
* The **triangle** can be considered like a very filtered (and soft) square signal. It is very low in harmonics (odd only) and will be very useful for creating sub basses, flute sounds, etc.…

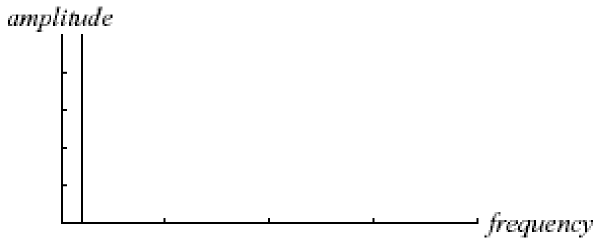




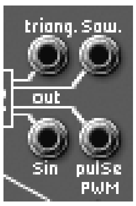
- The **sinusoid** is the purest of all. It is a single harmonic and produced a very “damped” sound. It can be used to reinforce the low frequencies of a bass sound or as a frequency modulator in order to create harmonics that don’t exist in the original waveforms.

sine



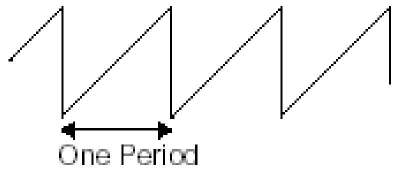


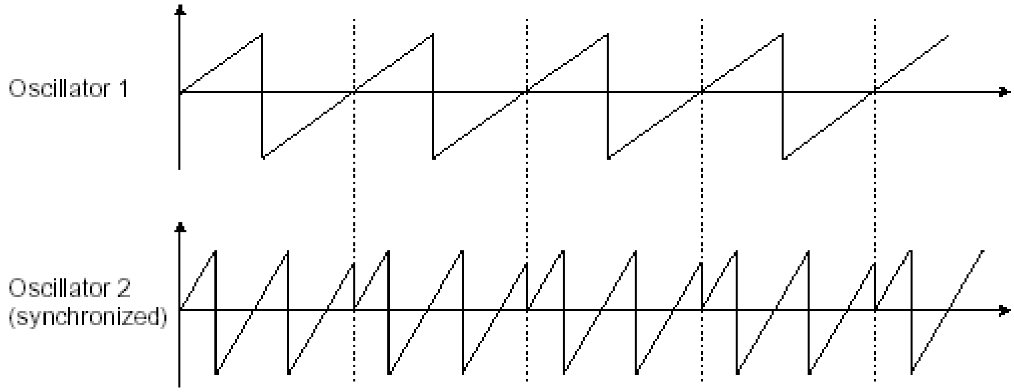
PWM (**P**ulse **W**idth **M**odulation) is a setting that allows you to modify the waveform cycle (or wave length). This can be done manually with the help of a knob “**PW**” or by modulation (with an envelope or LFO). This pulse width variation translates to a spectrum modification, resembling a waveform change.



The ARP2600 V waveforms on the oscillator2

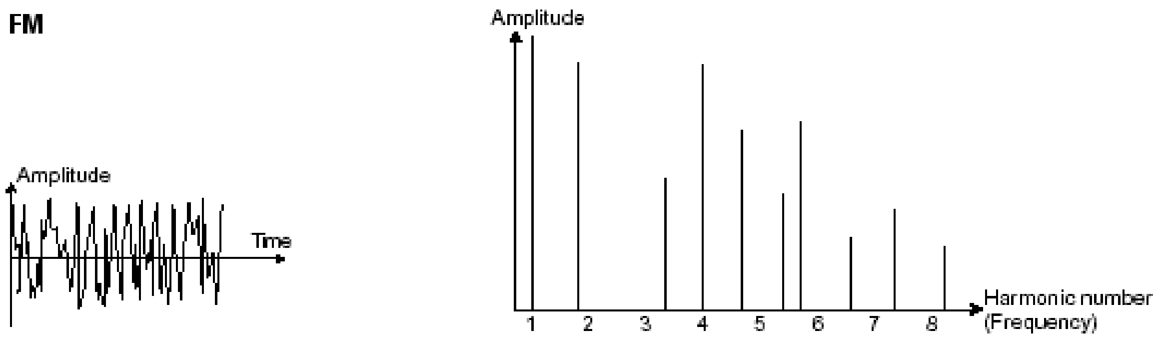
The **synchronization** of an oscillator with another creates more complex waveforms. If for example, you synchronize oscillator2 with oscillator1, oscillator2 will restart a new period every time the first oscillator completes a period, even if oscillator2 has not completed a complete period (this signifies that it is not tuned to the same tonality!) The more you tune oscillator2 upwards, the more you will encounter composite waveforms.





In the above image, oscillator2 is synchronized with the first and tuned to twice its frequency. The resulting waveform is unique in that it cannot be created by standard synthesis techniques such as layering or filtering.

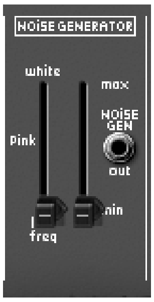
A **frequency modulation** (**FM**) can be created between 2 oscillators by connecting the audio output from a first sinusoidal oscillator to the modulation input of a second oscillator. On the ARP2600 V, if you turn the modulation rate ring, you will obtain a sound richer in harmonics. If you introduce a square or saw tooth signal, the result can be quickly distorted… but interesting for inharmonic sonorities like bell sounds or special effects for example.



The **noise** module

The noise signal spectrum has all frequencies at an equal volume level, often referred to as “white noise”. For this reason, the noise module is used to create different noises like the imitation of wind or special effects. White noise is the richest of noises. Pink noise is also regularly present on synthesizers. It is less rich in the high frequencies than white noise.

Also note that the audio output of noise can also be used as a modulation signal (especially when strongly filtered) to create random cyclic variations.

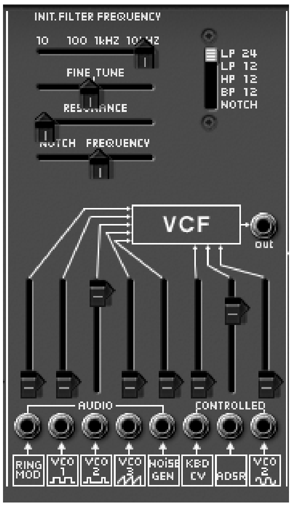


the ARP2600 V noise module

6.1.2 The filter or VCF

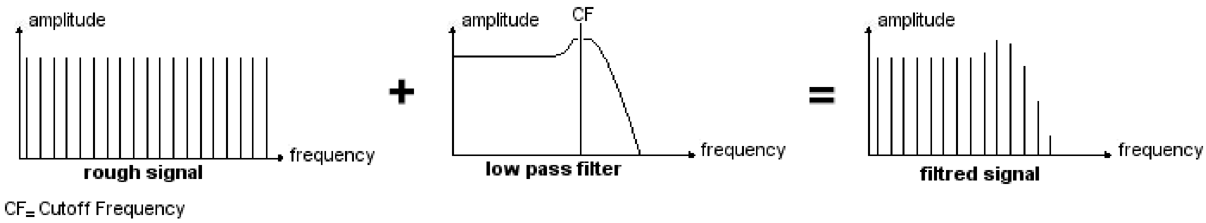
The audio signal generated by an oscillator (the waveform) is next generally directed to a filter module (**Voltage Controlled Filter**). It is this module that we use to control the sound by filtering (by subtraction, which explains the name given to this type of synthesis) the harmonics situated around a cut-off frequency. It can be considered to be a sophisticated equalizer that reduces, depending on the case, the high or low frequencies of a sound.

The removal of undesirable frequencies at the cut-off frequency is not done suddenly but progressively, depending on the filtering slope. This filtering slope is expressed in decibels per octave (or dB/Oct). The filters used in classic analog synthesizers have 24 dB/Oct or 12 dB/Oct slopes.



the ARP2600 V filter module

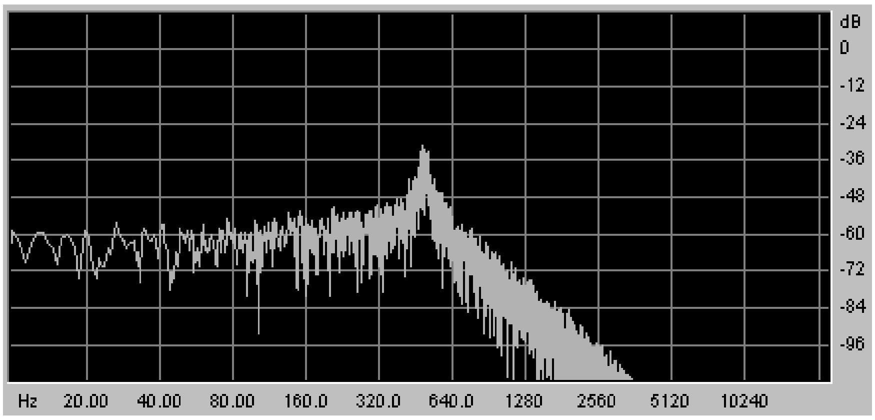
The ARP 2600 V offers two types of slope: 24 and 12 dB/Oct slopes.



On the ARP2600 V, you have access to four types of filtering. Let's have a look at some of his properties:

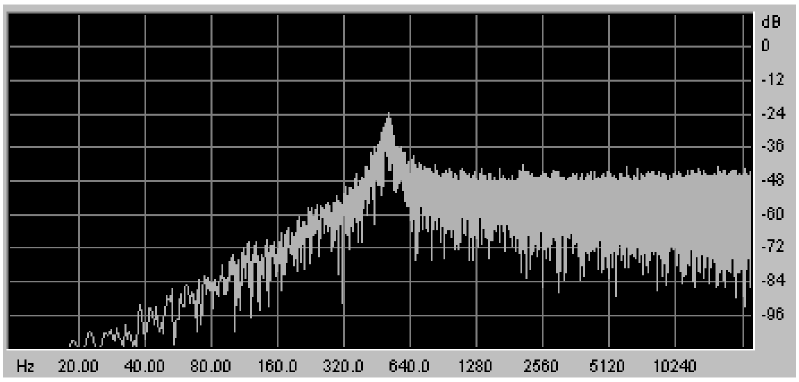
- The low-pass filter (*LPF*) progressively removes high frequencies above the assigned frequency limit (the cut-off frequency) and allows the sound below the cut-off to pass through unchanged. Depending on the setting we will hear the sound becoming more or less “brilliant”, more or less “dampened”.

This is the type of filtering that you will find more often than not on synthesizers that use subtractive synthesis. It can be found on most of the recent analog and digital synthesizers.

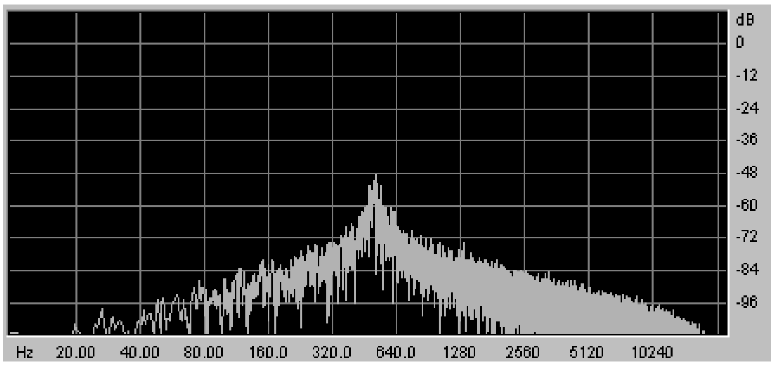


Spectrum of a noise signal proceed with a low-pass filter

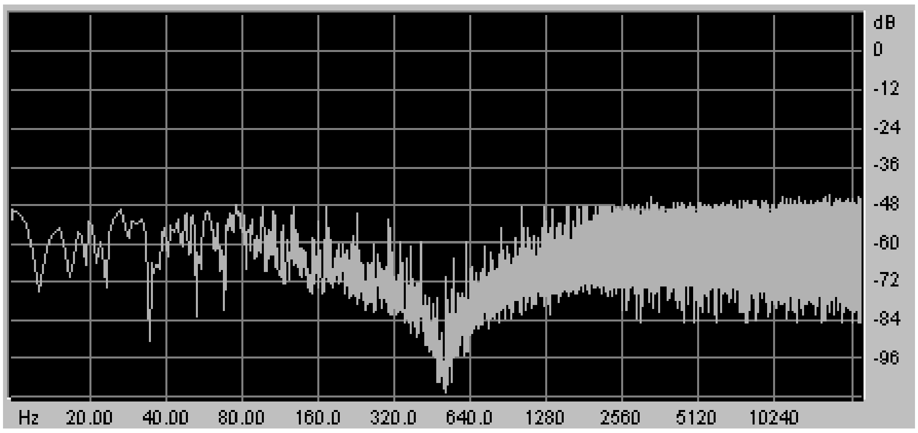
- The high-pass (high-pass filter or HPF), as opposed to the low-pass, eliminates low frequencies and only allows high frequencies past. The sound will thus become “thinner”. It is very useful for removing redundant low frequencies.



- The band-pass (band-pass filter or BPF) eliminates the frequencies situated on either side of the cut-off frequency. Use it to make a certain band of frequencies that you wish to emphasize appear. This will make the sound more “pinched”.



- The band-reject (band-reject filter or notch) eliminates the frequencies inside a band of frequencies. This filter is above all else interesting when we want to vary this band of frequencies (with the “frequency” on the ARP2600 V filters or the modulation of an LFO on this same parameter). You will thus obtain a sound close to a “phasing” effect.



These four types of filtering are often used on analog synthesizers.

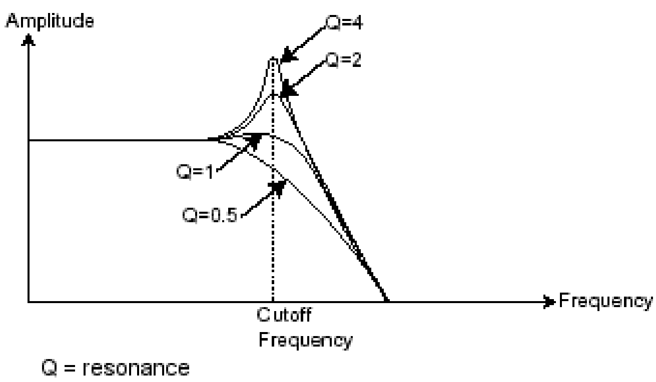
A second setting to compliment the cut-off frequency: the resonance.

(you will also find it called “emphasis” or “Q” – for Quality of filtering)

The resonance amplifies frequencies close to the cut-off frequency. The other frequencies remaining are either unchanged (below the cut-off frequency) or reduced (above the cut-off frequency).

On the ARP2600 V, you can increase the rate of resonance through the “resonance” potentiometer.

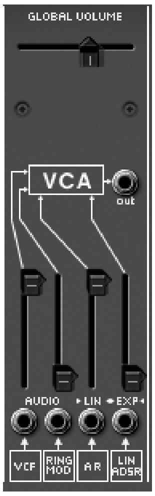
When you increase the resonance, the filter becomes more selective, the cut-off frequency is amplified, and the sound begins to “whistle” at the cut-off frequency.



With a high resonance level, the filter will begin to produce a sound close to a sine waveform. At this stage, the use of a key follow is very important as you can create a melody by tuning the cut-off frequency of the filter along with the frequency of the oscillators.

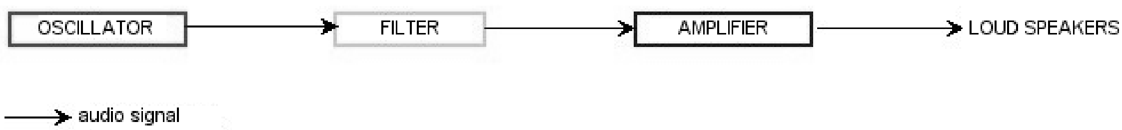
6.1.3 The amplifier or VCA

The amplifier (Voltage Controlled Amplifier) is charged with receiving the audio signal coming from the filter (or directly the one from the oscillator if it is not filtered) to adjust its volume with a potentiometer, before the signal is directed to the speakers.



the ARP 2600V amplifier

In conclusion, here is a scheme that may help you to understand the composition of a basic sound:



6.2 Complementary modules

6.2.1 The keyboard

If we hold down a key at this stage, the sound you will get from the speaker will be uniform, without life and without end! In fact, the oscillator delivers a continuous signal (the audio output of a wave form) of a fixed pitch. The only means of stopping this sound that quickly becomes unsupportable is by lowering the filter cut-off frequency so that it becomes more and more thick until it disappears; or more simply, to lower the volume of the amplifier!

- To trigger and stop this sound, at the tone that we want, we use a keyboard, which will be connected to the oscillator. This will "play" as soon as a key is pressed and will mute it as soon as released. Of course, this connection is made through MIDI (it replaces the "gate" type connection of analog synthesizers, which triggers the sound when the key is pressed and stops when released).
- In the second case, so that the sound is correctly tuned with the keyboard notes, we need to apply a key follow modulation (replacing the 1Volt/octave control present on most analog synthesizers).

If you don't have a keyboard, you can play on the ARP2600 V virtual keyboard.

the virtual keyboard of the ARP2600 V

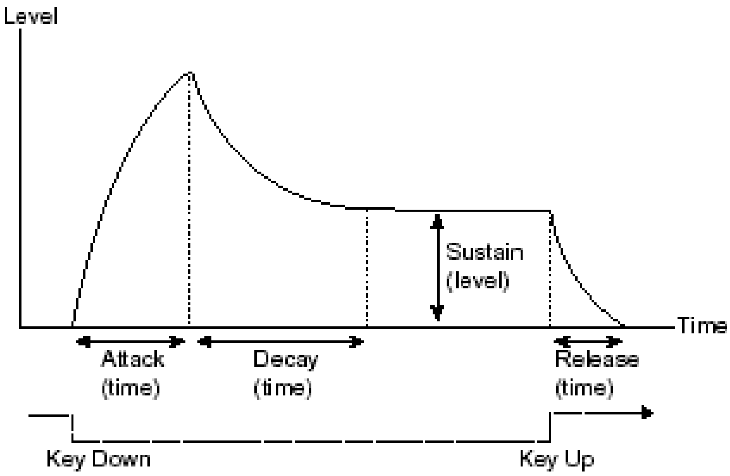
6.2.2 The envelope generator.

The envelope generator, connected to the amplifier, is used to "sculpt" the form of a sound during a cycle, which begins when we press a note on the keyboard, and ends when we release.

the envelope module

The most common envelope modules use 4 settings that we can vary:

- The Attack is the time that the sound will take to reach its maximum volume once the key has been pressed on the keyboard.
- The Decay is the time that the sound will take to decline after the key is played.
- The Sustain is the maximum level of volume that the sound will reach when a key is pressed.
- The Release is the decline time after the key has been released.



The ARP 2600 V also offers a second simplified envelope (Attack, Release).

The envelope generator can also be used to modulate other settings, like the filter cut-off frequency or the frequency of an oscillator for example.

6.2.3 The low frequency oscillator.

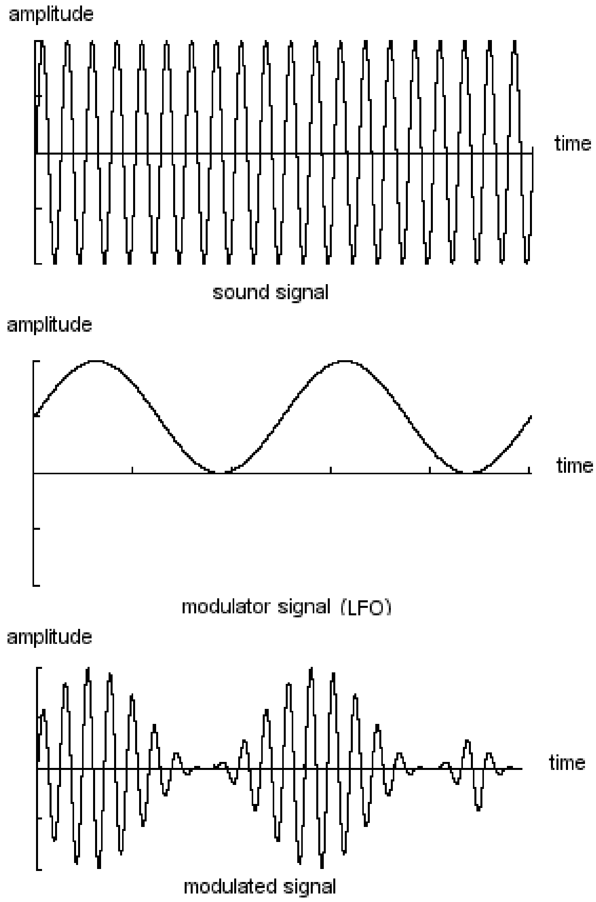
The LFO (Low Frequency Oscillator) possesses, among other things, the same characteristics as the classic audio-frequency oscillator but does not produce frequencies higher than 20 Hz. In other terms, you won't hear the sound if you connect the audio output of an LFO in an amplifier.
Not being used to produce a sound, it can be used to create a cyclic modulation on the parameter on which it is connected.

On the ARP2600 V,the LFO module is situated on the left of the sequencer .

the LFO module of the ARP2600 V

For example:

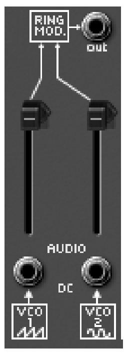
- If you connect an LFO to the modulation input of an amplifier, the sound volume will increase and decrease in an alternate manner depending on the speed (the frequency) of this LFO. This will create a tremolo effect.
- To produce a vibrato effect, simply connect the sinusoid output of an LFO to the modulation input of an oscillator. The frequency of this oscillator will thus be modulated up and then down.
- Finally, try to connect an LFO output to the modulation input of a lightly resonant low-pass filter and you will obtain a "wah wah" effect.



VCA modulated by a LFO

6.2.4 The ring modulator

The ring modulator is a module which multiplies two signals (generally two oscillators) so as to generate non-harmonic frequential components. We can thus easily obtain metallic sounds.



the ring modulator of the ARP2600 V

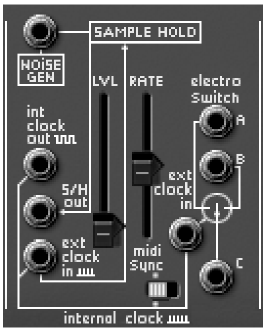
6.2.5 Sample and hold

The sample and hold module lets us sample the signal connected as input. The values for modulation are taken at every trigger signal connected as input.

This module is notably interesting for creating random modulations by sampling the noise signal.

It is also possible to use it for more rhythmic cyclic modulations by connecting a square or sawtooth wave form coming from a VCO to the sample and hold input.

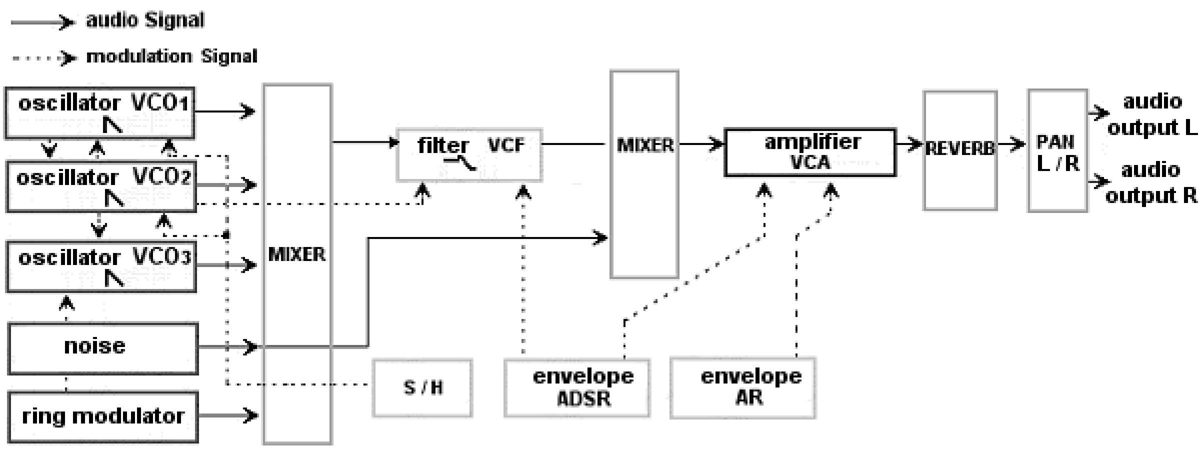
The famous sound of R2D2 in Star Wars was created with the sample and hold module on the ARP2600!



the sample and hold module of the ARP2600 V

Now to finish, the scheme of a full synthesizer containing:

- 3 oscillators (VCO)
- 1 noise module
- 1 ring modulator
- 1 sample and hold
- 2 mixers (mix of the 3 VCO, noise and ring modulator towards the filter and the VCA)
- 1 filter (VCF)
- 1 amplifier (VCA, can be placed in stereo space with the pan potentiometers)
- 1 envelope ADSR
- 1 envelope AR



7 A few elements of sound design

Here is a series of five examples designed to guide you through the creation of a sound and a sequence. They are classed in order of difficulty and organized into two parts:

- The first part will give you an introduction to sound synthesis. For this you will start with the most simple patch (make a VCO oscillator “ring” in a VCA amp going through the filter) and finish by programming of a richer sound (several sources of VCO, VCF filter, VCA, envelopes…)
- The second will take you through every aspect of the sequencer

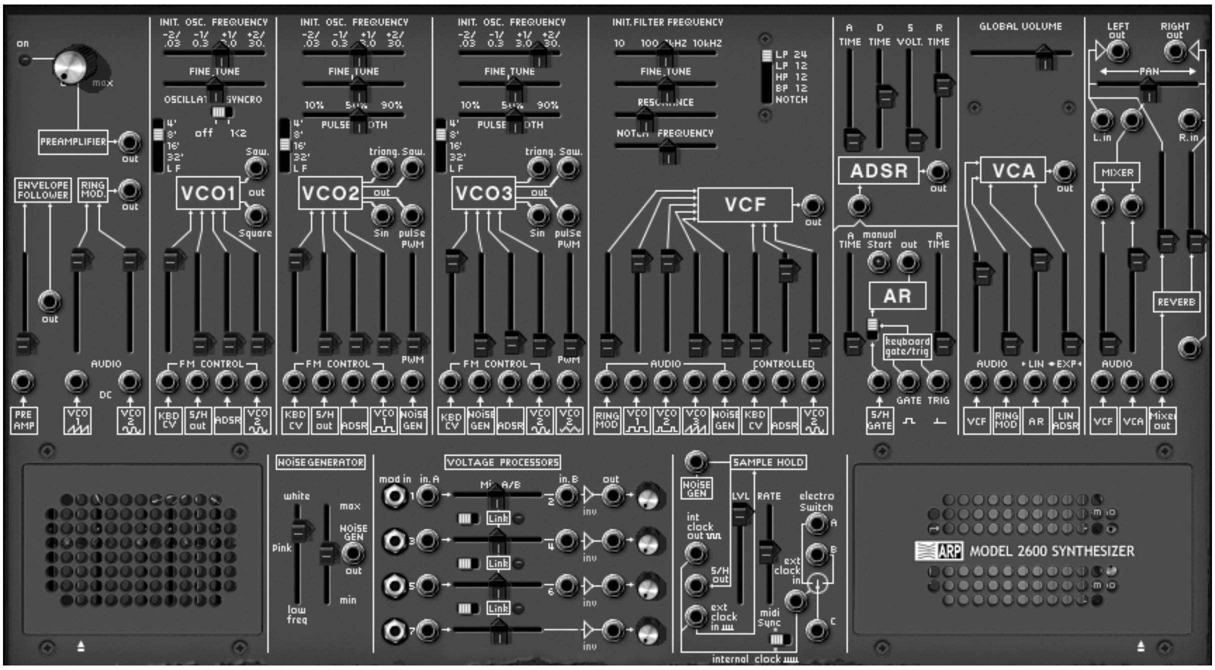
7.1 Simple patch without cabling

To begin, we will look at programming an elementary monophonic sound. It will be composed of four modules:

- two oscillators (VCO1 and 2)
- a low-pass filter (VCF)
- an amplifier (VCA)
- the ADSR envelope connected to the filter

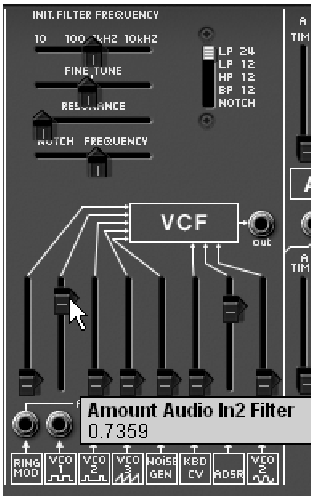
You will thus obtain a basic patch for subtractive synthesis.

Figure 1 shows the different potentiometer positions:



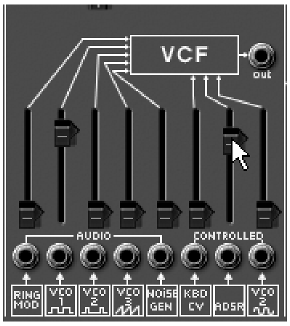
To begin, choose the preset “Temp/ Blank” in the “Template” / “Temp_Synth” bank. You will notice that no sound is heard when you play a note. Don’t worry, this is normal! Only the volume of the VCA amplifier mixer (the VCF filter output) and the ADSR envelope input of the VCA are open.

- On the filter module, raise the vertical “VCO1” potentiometer. This is the volume of the first oscillator. When you hold down a note, the sound seems continuous and flat!



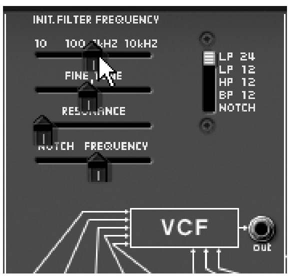
raise the "VCO1" potentiometer on the filter module

- To give it more life, still on the same module, raise the "ADSR" potentiometer corresponding to the envelope of the same name.



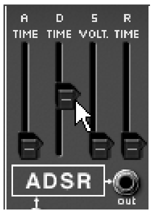
raise the "ADSR" potentiometer

- Lower the filter cut-off frequency ("Initial Filter Frequency") potentiometer almost completely so as to really hear the effect produced by the envelope on the filter.



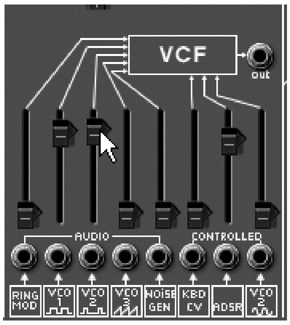
Lower the filter cut-off frequency

- Lower fully the Sustain potentiometer on the ADSR envelope and lower the Decay a bit as well. (set a value aroud 100ms). The duration of the sound becomes shorter.



lower the Decay potentiometer

- Now raise the volume of “VCO2” on the VCF mixer (“VCO2”). You will hear the 2 oscillators play in unison.



raise the volume of VCO2

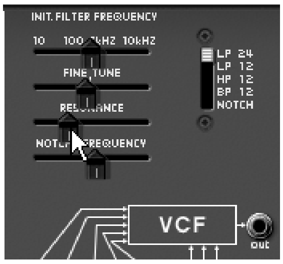
- Lightly detune the second oscillator (“VCO2”). The sound becomes more lively and “fatter” !

Lightly detune the second oscillator

- Set the VCO2 an octave lower: set the “Range” octave selector to “16’”.

set the “Range” octave selector to “16’”

- Finally, lightly raise the filter resonance. This will give you a typical 70’s bass sound !



raise the filter resonance

> Save this sound as “bass1”, you can reuse it later on. For this, click on the “Save_as” button and choose the option “new bank” (at the bottom of the menu). Provide a new name to the bank (your initials for example), the sub-bank (“basses” for example) and to the new preset (“Bass1” for example).

7.2 Polyphonic patch with cabling

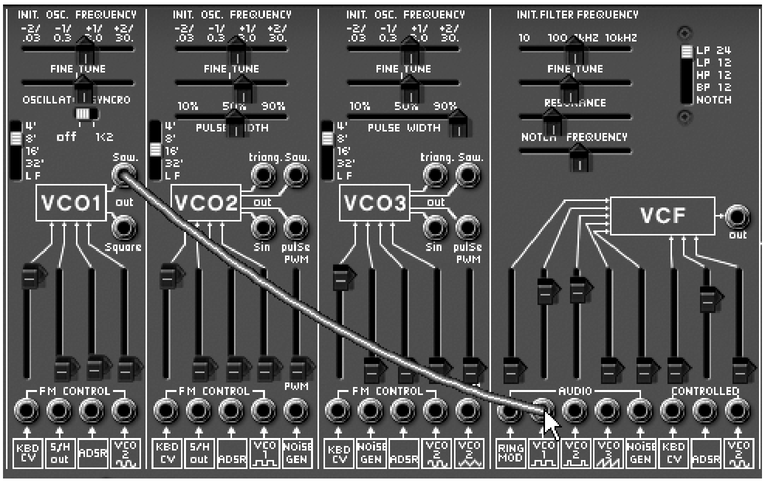
Let’s reuse the sound “bass1” as it will be useful to us for the next preset.

We will see how to surpass the limits of the pre-cabled (internal connections between the modules don’t need cables). Thanks to this method, you will discover the richness of the ARP2600V.

This preset contains:

- three oscillators (VCO1, 2 and 3)
- a low pass filter (VCF)
- an amplifier (VCA)
- the ADSR envelope connected to the filter
- the AR envelope connected to the amplifier
- a LFO connected to the filter

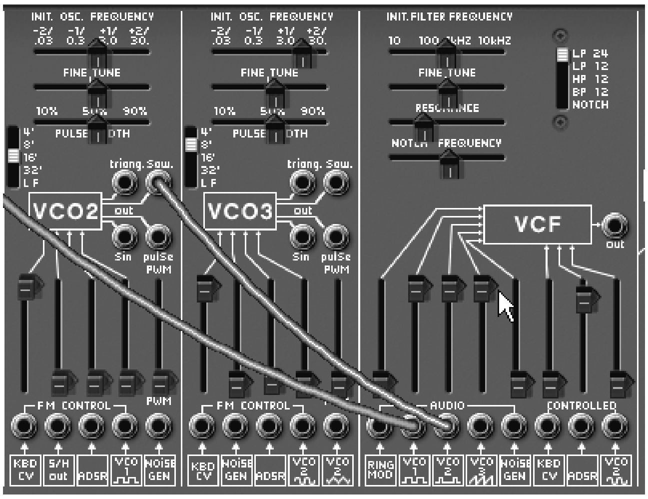
- Drag a cable from the saw-tooth output on oscillator1 to an audio input of the filter mixer. For this click on the VCO1 “Saw” output and direct the mouse to the “VCO1 square” audio input on the VCF mixer.



direct the “Saw” output to the “VCO1 square” audio input on the VCF mixer

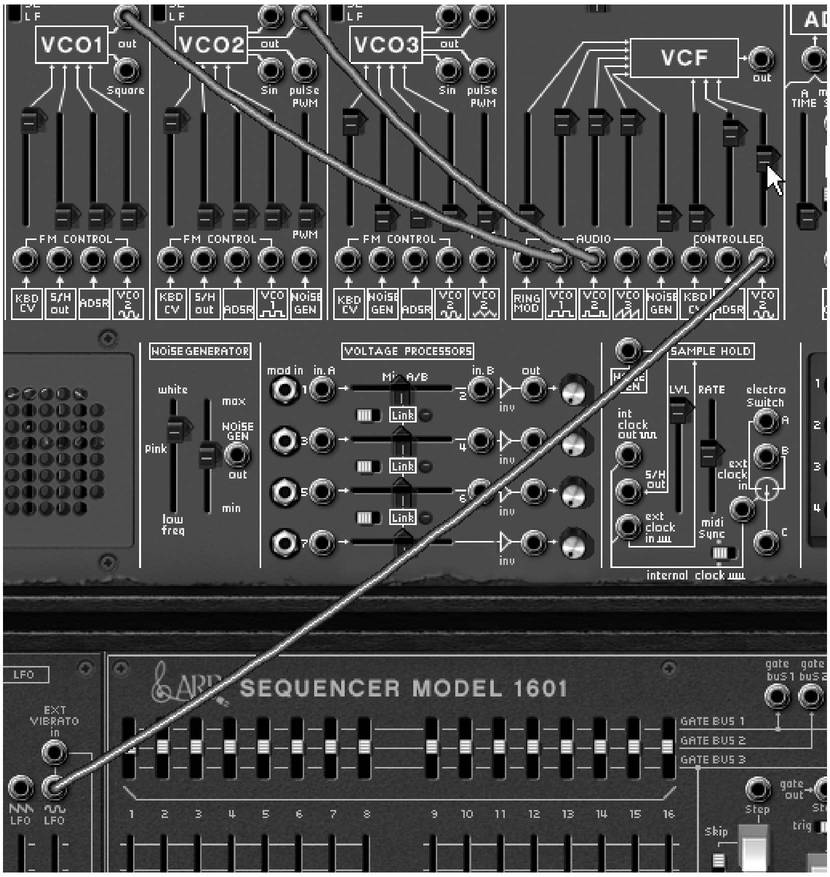
- Still on this mixer, lower the VCO2 volume. You will notice the change of tone resulting from the change of wave form on VCO1. If you need to be convinced, disconnect the cable (click on the mixer audio input to disconnect the cable). You thus move from a saw-tooth to a square signal. Reconnect the saw-tooth to continue with this preset.
- Do the same with oscillator2: direct the saw-tooth wave form to the VCO2 input on the VCF mixer.

- Still on the same mixer, raise the volume of oscillator3 to complete the base tone. You will hear a sound composed of three oscillators.



raise the volume of oscillator3

- Let's apply a variation to the filter cut-off frequency with the LFO. For this, connect the sinus wave form of the "LFO" module (on the left of the "ARP sequencer") to the "control VCO2 sin" modulation input on the VCF. Raise the volume potentiometer for this modulation. The filter cut-off frequency will vary in a cyclic manner.

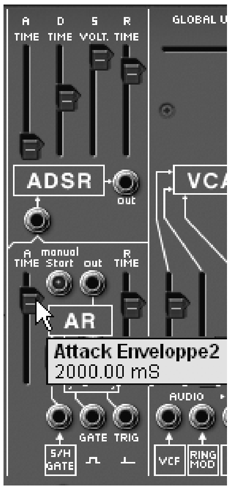


raise the volume of the LFO modulation

- If it has a tendency to “disappear” too much (you don’t hear any more sound), gently raise the cut-off frequency potentiometer.
- Set the LFO oscillation speed with the “LFO Speed” potentiometer. For this type of sound choose a slow speed (around 0.10 Hz).

set the LFO oscillation speed

- Raise the Attack time (“A” around 2000 ms) and the Release (“R” around 750ms) of the AR envelope. At last, raise fully the Decay time (“D”) of the ADSR envelope. These types of envelopes go better with accompaniment sounds.

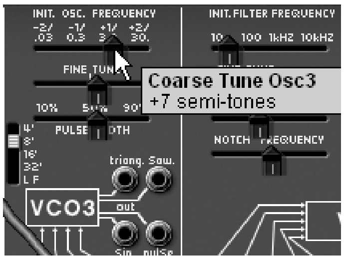


The envelopes settings

- Set the playing mode selector situated on the left of the LFO module to “Poly” (polyphonic) position so as to be able to play chords. Then choose the number of polyphonic voices by clicking on the “voices” button, situated on the toolbar. (For exemple, choose “6” voices)

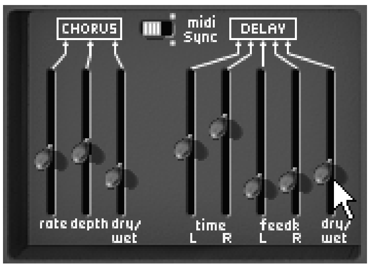
polyphonic mode

- At this stage, you can also enrich your chords by dedicating a third oscillator to sevenths (“+7 semi-tones”) - or fifths (“+5 semi-tones”).



set the third oscillator to sevenths

- If you wish to bring more space to your sound, add a delay or chorus effect to it. For this, click on the “Chorus” or “Delay” buttons situated on the toolbar, open the grid on the left of the synthesizer. This holds the settings for these 2 effects. If you use delay, you can synchronize it with the MIDI tempo of your host sequencer.



the “Chorus” and “Delay” settings

- Don’t forget to save your new sound, as like the previous, it will be useful as a template for other more complicated sounds.

7.3 Special effects patch with the help of the tracking generator

With this example, you will discover a few of the many possibilities for modulation with the tracking generator / LFO. This module is one of the major additions to the ARP2600 V compared to the original.

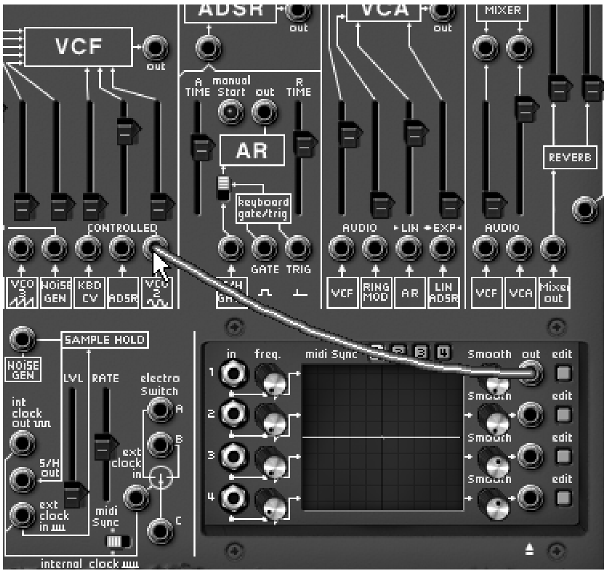
Let’s again use the “bass1” sound as template for the following preset:

- two oscillators (VCO1 and 2)
- a low pass filter (VCF)
- an amplifier (VCA)
- the tracking generator connected to the filter

We will first see how to create a “complex LFO” modulation type on the filter cut-off frequency with the help of the tracking generator.

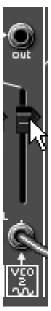
- On the tracking generator module (situated on the bottom right of the synthesizer section), click on the “open tracking” button under this grid to open it. Then, connect the output of the first tracking line (“out”) to the “VCO2 Sin” modulation input of the filter.

open the tracking grid...



... then, connect the output of the first tracking line to the filter modulation input

- Raise the potentiometer above the modulation input jack. This is the modulation rate.

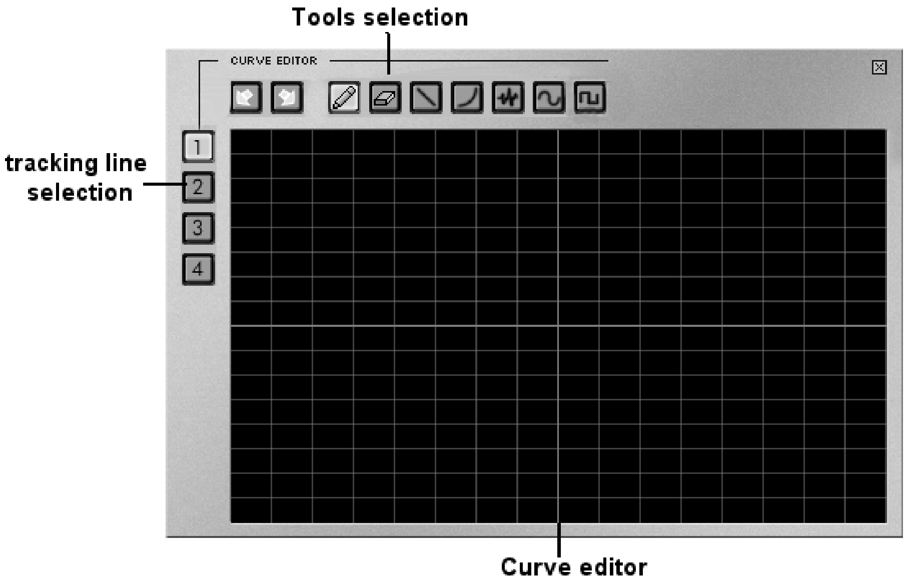


the modulation rate

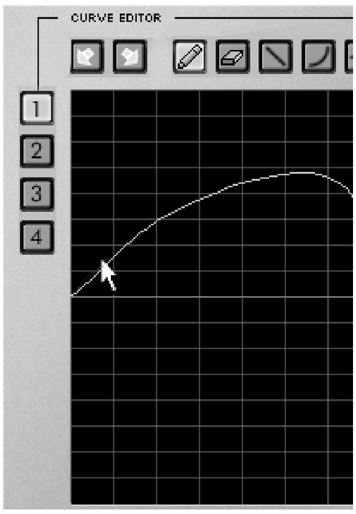
- Next click on the “edit” button for this tracking line. A secondary window appears. It will allow you to graphically draw the wave form that will modulate the cut-off frequency.

click on the “edit” button

- It is composed of three parts: on the left, the selection of the 4 tracking lines, in the middle, the interface for drawing the wave form, and above, the choice of drawing tools.



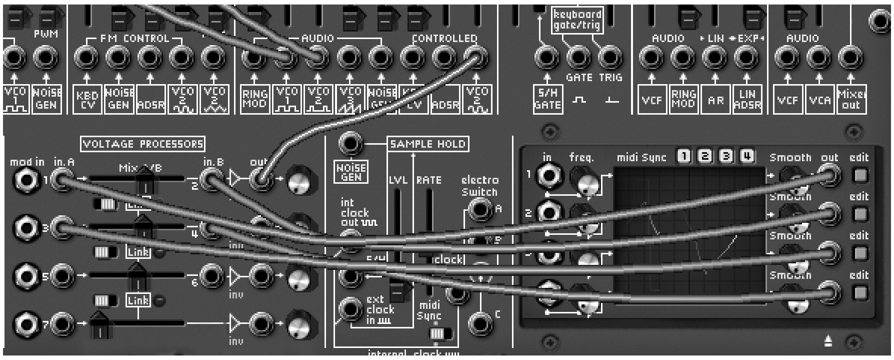
- By default, the “pencil” tool is selected. Start by drawing a freehand curve. The form is not important. Once this is done, play a note to hear the result of this modulation. You can set the oscillation speed with the “Freq” potentiometer which is on the left of the tracking generator.



drawing a freehand curve

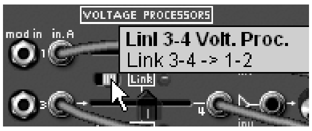
- If you wish to modify your curve, return to the editing screen to make changes. The modification is directly taken into account as soon as you click on the edit screen. Nothing stops you from creating several types of curves (freehand, curve, sinus, square, noise…) on a single line. To do this just use the appropriate tool.

- You can combine the four tracking lines for a single modulation input. For this, Connect the four tracking generator outputs to the “voltage processor” 1, 2, 3 and 4 inputs.



Connect the 4 tracking generator outputs to the “voltage processor” inputs

- Place the “link A/B” selector switch to the right to link the two lines of mix to the output of the first.



Place the “link A/B” selector switch to the right

- Of course, the more tracking lines you combine and send to the same destination, the more difficult it becomes to master and hear.

7.4 Patch using the sequencer to create a melody

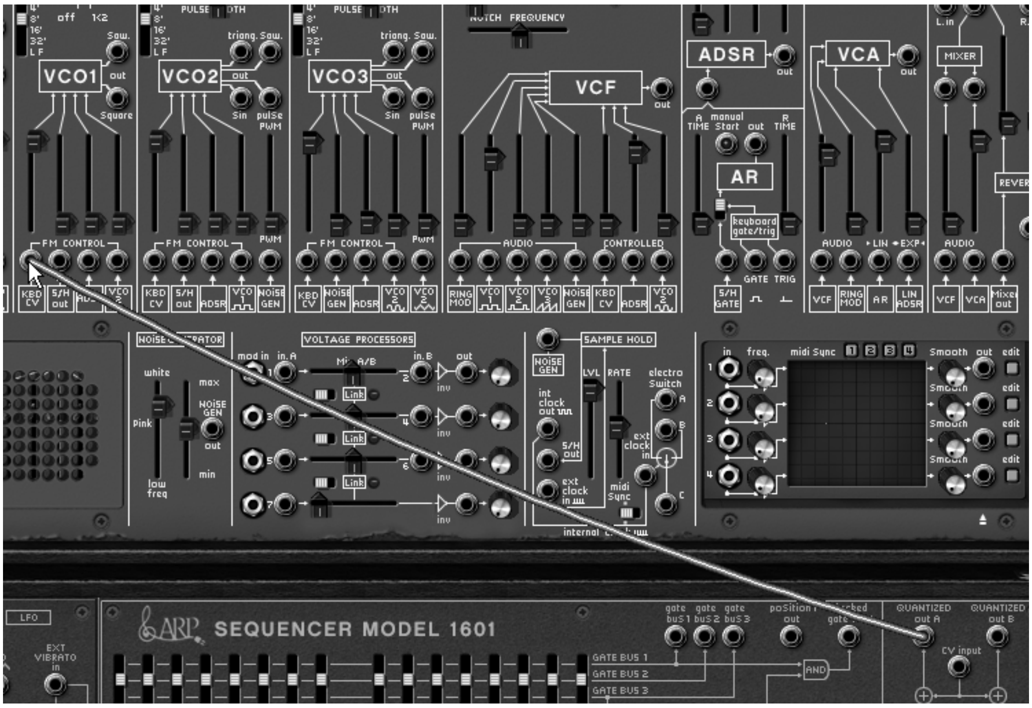
The ARP sequencer model 1601 was one of the most frequently used sequencers of its time (the 70’s). It was added to the ARP2600 V to increase the possibilities for creating melodies and sounds.

The next two examples show how to use the ARP sequencer to create a melody and a sequence of modulation of a synthesis parameter.

For these 2 examples, return to the “bass1” preset. It presents the ideal sound for a melodic sequence or for applying a modulation sequence to the filter cut-off frequency.

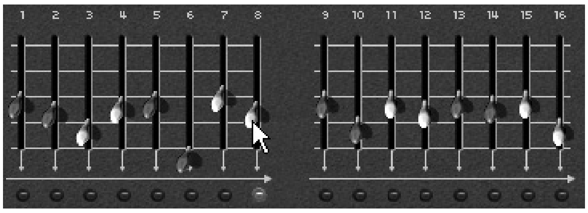
The “CV input” is connected directly to the key follow which allows is to trigger a sequence tuned to the note played. The sequencer will automatically start when you play a note.

- So that your sequence is tuned by following the semi-tones, connect the “Quantized output A” on the sequencer to the input oscillator1 key follow (“VCO1” > “CV KBD”).



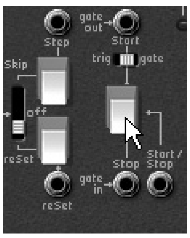
connect the "Quantized output A" of the sequencer to the KBD CV input oscillator1

- Play a note on your keyboard (C4 for example). The sequencer starts and you will hear a series of 16 notes of identical pitch, this is normal.
- Set each of these 16 potentiometers representing 16 sequence steps to a different value. This will be the melody played in a loop.



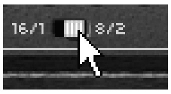
16 sequence steps to a different value

- You can stop the sequencer by clicking on the "Start/Stop" button.



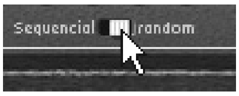
stop the sequencer by clicking on the Start/Stop" button

It is possible to obtain a sequence of 16 steps like the one you have just created by using 2 independent sequences of 8 steps. For an 8 step sequence, place the "1/16 > 2/8" selector switch to 2/8. The sequencer will only play half of the sequence.



2 independent sequences of 8 steps

To play the 16 (or 8) steps randomly, switch the “sequential / random” selector switch to the right (“random”)



To play the steps randomly

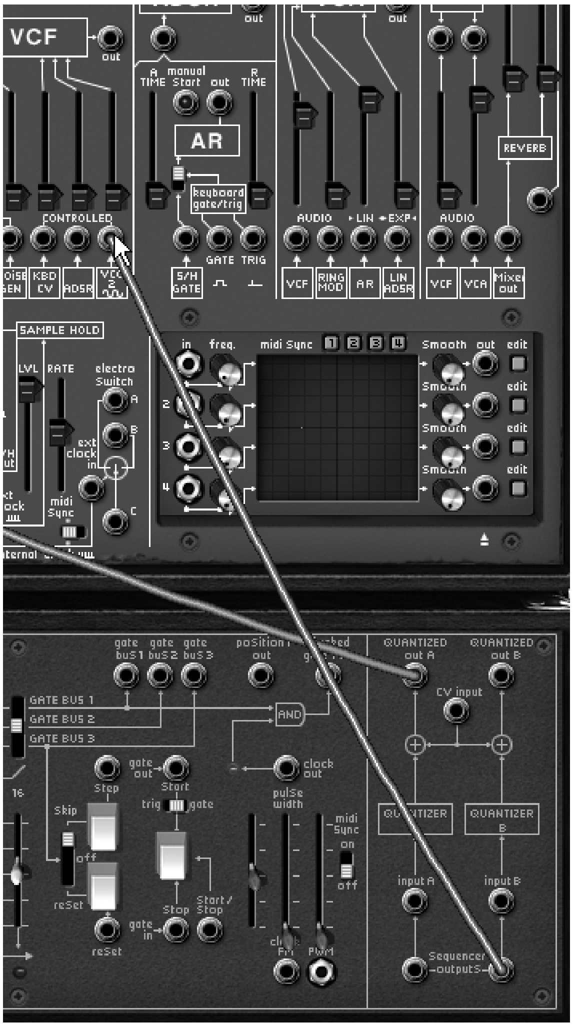
7.5 Patch using the sequencer to create a modulation sequence

Let’s look at creating a sequence of modulation on the filter cut-off frequency.

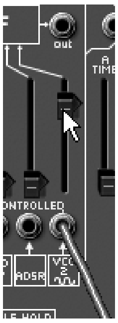
This will be very useful, notably for special effects such as a “techno” sequence.

Reuse the previous example as a template for this exercise (if you have not created it, go directly to the next step).

- Connect the sequencer B output to the second modulation input acting on the filter cut-off frequency (ADSR). Raise the potentiometer of the same name to increase the depth of this modulation.



Connect the sequencer B output to the second modulation input of the filter...



...and raise the modulation potentiometer

- Play a note on your keyboard to start the sequencer. Change the positions of the 16 potentiometers representing 16 steps in the sequence. You can hear the variations on the opening of the filter cut-off frequency. This creates an automatic wah wah effect.
- It is also possible to simultaneously create a melodic sequence on the eight first steps and a modulation sequence on the last eight. For this, set the “1/16 > 2/8” interrupter to 2/8. The sequencer will only play half of the sequence.

It is of course possible to connect any other ARP parameter to the sequencer, like the amplifier volume (VCA) for example, the change of wave form width for oscillator square wave form, etc…

These different examples were of varying difficulty. We hope that they will have helped you get to know some of the possibilities offered by the ARP2600V. But don’t hesitate in making your own attempts at programming; this is the best way to progress and find a certain originality.

8 Using the ARP2600 V in different modes

8.1 Stand-alone

The ARP2600 V application can be used as an instrument independent of a sequencer (stand alone mode). It allows you to open one or several instruments, and to play with a master MIDI keyboard.

Attention! The ARP2600 V application is only available for **Windows 2000/ XP and Mac OS X**

8.1.1 Launching the application

To launch the ARP2600 V application from Windows, go to **Start > Arturia > ARP2600 V** and choose **ARP2600 V** .

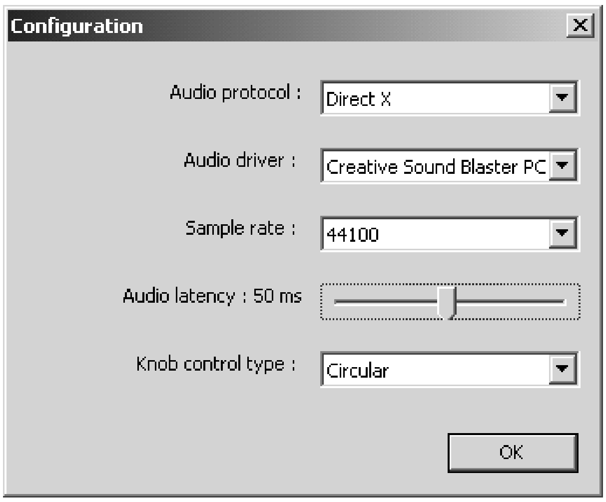
On Macintosh, open the installation folder and double click on the ARP2600 V application icon.

You can also double click on a saved document in order to open the corresponding configuration in the ARP2600 V application.

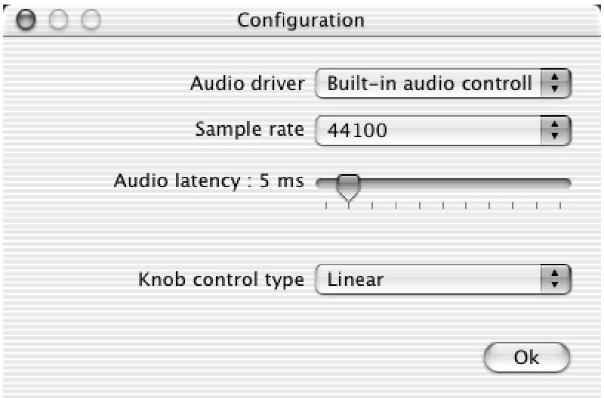
8.1.2 Setting preferences

The preferences window allows you to set the global preferences in the ARP2600 V application. These settings are automatically saved.

To display the preferences window go to the **File > Preferences** menu on Windows and **ARP2600 V > Preferences** on Mac.



The preferences window



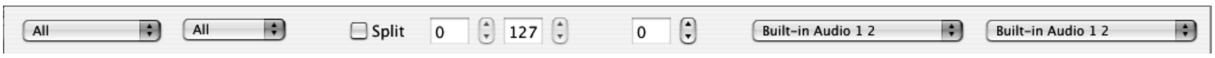
The Macintosh preferences window

- Protocol (Windows only): Select the audio protocol that you want to use. If you have ASIO drivers for your sound card, it is highly recommended that you use them instead of DirectX protocol; the ASIO drivers provide a higher performance rate in relation to the DirectX driver.
- Driver: Select the driver corresponding to the sound card that you want to use.
- Sample rate: Here choose a sample frequency among those offered by your sound card.
- Latency: Here you can set the optimal audio latency in relation to the performance of your sound card and your system. Attention: a latency set too low may cause occasional jumps, clicks, or pops in the sound.
- Potentiometer mode: Here choose the instrument potentiometer control mode.

8.1.3 The control bar

Each instrument has a control bar setting the routing of MIDI events as input, and sound as output.

The Windows control bar



The Macintosh control bar

8.1.3.1 Configuration of MIDI routing

The first section of the tool bar is for selecting the MIDI input that will be applied to the instrument.

Choose the MIDI port to which the keyboard that you are using to control the instrument is connected, and the channel that you wish to use. You can choose to respond to one or all of the channels of this MIDI port by selecting ‘All’ in the MIDI channel choice.

8.1.3.2 Configuration of the keyboard zone

The keyboard zone allows you to use only a section of the keyboard to control the instrument. In this manner, you can play several instruments on the same keyboard, each instrument responding to a different zone.

To activate this function, check the ‘zone’ option in the tool bar. You can then limit the keyboard zone by setting the lowest and highest note to be applied to this instrument.

8.1.3.3 Configuration of the octave

The octave allows you to transpose the notes of your keyboard by one or several octaves. This function is interesting if your keyboard doesn't cover the octave in which you wish to play or if you have activated the keyboard zone option.

8.1.3.4 Configuration of the audio input and output

The two last parts of the toolbar are used to select the audio channel on which you wish to play this instrument, and the one that will provide audio data.

If the selected soundcard has several audio outputs, respectively audio inputs, you will find those available in this list. Just choose the pair of outputs or inputs that you desire.

8.1.4 Information on the level of processor use

The processor usage gauge allows you to control the level of processor consumed by the sound synthesis in real time.

On Windows, this gauge is directly visible in the instrument tool bar.

On Mac, you can display it by choosing **Window > Cpu** in the application menu or from the shortcut ⌘+ L.

> Attention: this information only comprises the processor load corresponding to the sound synthesis; it is therefore less than the global system load.

8.1.5 Saving an instrument

Saving allows you to save the state of an instrument, that is to say its sound settings, as well as the audio and MIDI configuration.
To save a configuration, just select File > Save or File > Save as … to save the configuration under a new name.

> Attention: if saving a configuration in the ARP2600 V application saves the sound settings of the instrument, this has nothing to do with the saving of the presets of the instrument itself (cf. 4.1.3 Saving a user preset). The saving in the application does not imply the saving of the current preset.

8.2 VST™

8.2.1 Installation

8.2.1.1 For Windows

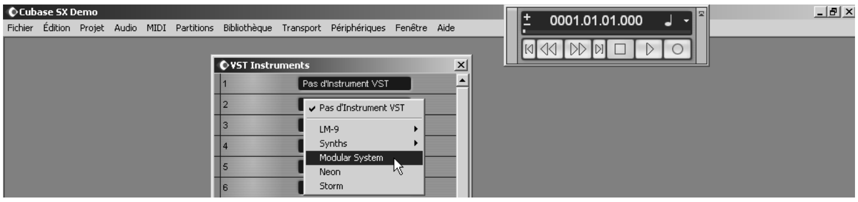
- During the installation, check the "VST" option among the choice of plug-in formats offered. The installer will automatically detect the VST instruments folder used by your version of Cubase. In the case of another VST compatible sequencer, like Logic Audio for example, you might need to manually copy the plug-in file to the appropriate folder. You can find this file after the installation in the folder "C:\Program Files\Arturia\ARP2600 V\VSTPlugin". The file is called "ARP2600 V.dll".

8.2.1.2 For Mac OS X

During the installation, check the "VST" option among the choice of plug-in formats offered. The VST plug-in will be automatically installed in the system folder corresponding to the VST instruments, and can be used by VST applications of the type host.

8.2.2 Using the instrument in VST mode

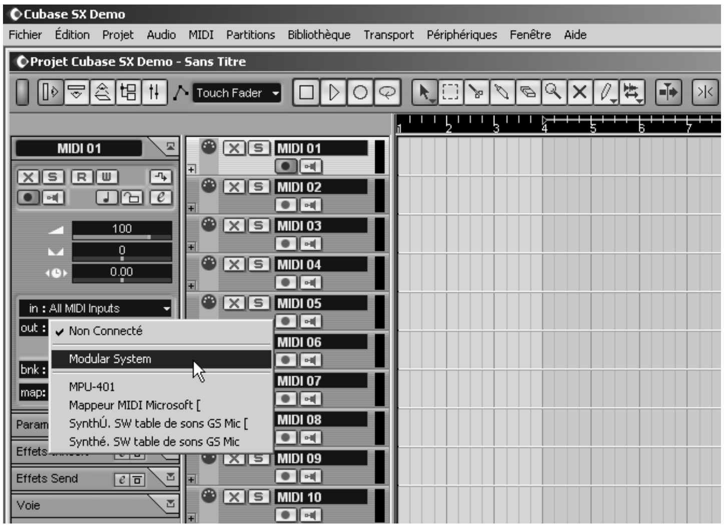
Opening the ARP2600 V VST plug-in is done like any other VST plug-in; please consult the user manual of the host sequencer for more information. For Cubase SX, open the menu “Devices / VST Instruments”, and choose ARP2600 V in the rack:



Opening the ARP2600 V in Cubase SX on PC

8.2.3 Connection to a MIDI track

So that the ARP2600 V can play the information coming from a MIDI track, you must choose a MIDI track and select the ARP2600 V as the MIDI output for this track using the menu in Cubase which is used for this:



Connection of a MIDI track to the ARP2600 V

The events played on a MIDI keyboard are then transmitted by your sequencer to the ARP2600 V. It is of course possible to record these MIDI events, and use all of the sequencer’s MIDI editing possibilities.

8.2.4 Saving presets

When the session is recorded, the state of the ARP2600 V is recorded as is, even if the programming does not correspond to any preset. For example, if you were working on preset “P1” on which you had modified the parameters (without having saved them under the name “P2”), the next time the song is opened, the ARP2600 V will load preset “P1” *with the modifications*.

The drop-down menu offered by the VST sequencer to save plug-in settings can of course be used with the ARP2600 V. Nevertheless, it is strongly recommended that you use the internal ARP2600 V menu: the presets saved in this manner can be used in any other mode (standalone or other sequencer), and can be exported as a distinct and separate file.

8.2.5 Automation

Automation works with the ARP2600 V as like any other VST plug-in (refer to the VST sequencer documentation for more information on the automation of plug-ins). Preset changing (and therefore cable changing) cannot be automated.

8.3 Pro Tools™

8.3.1 Installation

During the initial installation, select Install as a RTAS/HTDM plug-in.

Then, when you are asked, indicate the folder in which the other RTAS and HTDM plug-ins are placed. Generally the path is:

- for Mac OSX: System Folder/DAE Folder/Plug-Ins
- for Windows: C:\Program Files\Common Files\Digidesign\DAE\Plug-Ins
- If your system can or cannot use the HTDM plug-ins or not (see the next section), the installation is the same.

8.3.2 The RTAS and HTDM standards

The ARP2600 V can function with the audio Digidesign Audio Engine (DAE) in two ways:

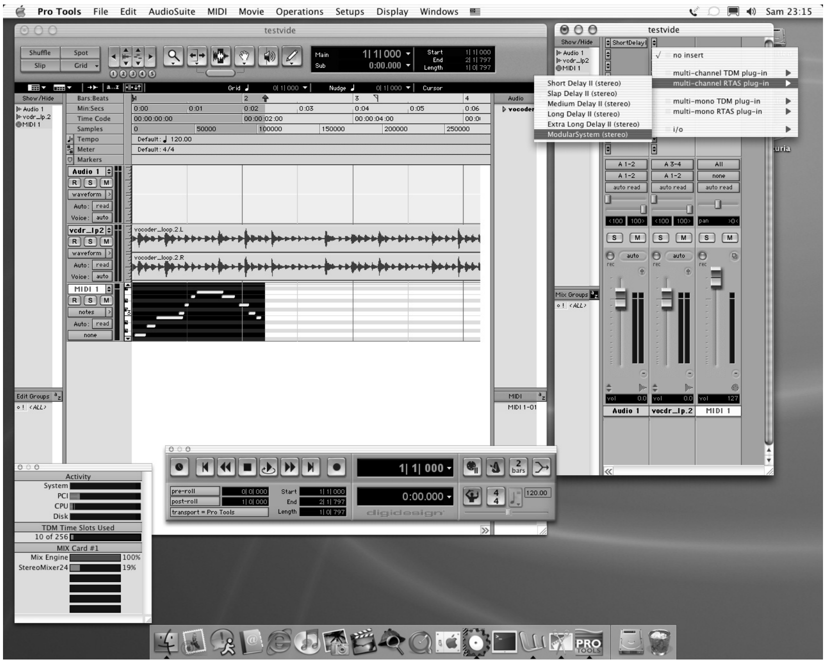
- As a RTAS plug-in (Real Time Audio Suite).
- All of the Pro Tools systems are compatible with this standard: the audio treatment is realized entirely by the central unit, and does not require any specific extension card (TDM system type). On the TDM systems, the RTAS plug-ins can only be loaded on an audio track before the TDM plug-ins. They can only be loaded on an Aux Input or a Master Fader.
- As a HTDM plug-in (Host Time Division Multiplexing).
- Only the TDM systems (with at least one extension card) under Mac OS X can use these plug-ins (refer to Digidesign for the future HTDM compatibility with Windows). The latter work exactly like TDM plug-ins (no limitations in the insertion positions, etc…), with the difference that in this case, the heart of the treatment is realized by the central unit, and not by the extension cards.
- The advantage of this standard is the subtlety of TDM, and this, loading at least the DSP of the cards.

Recap of the compatibilities:

	Mac OSX	Windows XP
TDM System	RTAS and HTDM (stereo in/stereo out)	RTAS (stereo in/stereo out)
Other Systems (Pro Tools LE, Free)	RTAS (mono in/stereo out and stereo in/stereo out)	RTAS (mono in/stereo out and stereo in/stereo out)

8.3.3 Opening the instrument

Access to the ARP2600 V plug-in is done like any other plug-in, through for example an audio track insert:



- TDM systems: the ARP2600 V must be loaded on an audio stereo track. Opening it as a HTDM plug-in is done through the TDM sub menu, as a HTDM plug-in is used exactly like a TDM plug-in.
- Other systems: the ARP2600 V can be loaded on a mono audio track (which becomes stereo after the insertion) or stereo.
- You can play the ARP2600 V with the mouse or the virtual keyboard.

8.3.4 Connection to a MIDI node

So that the ARP2600 V can play the information coming from a MIDI track, we must associate it with the MIDI node of the latter through the appropriate menu:

We can also control the ARP2600 V through a command keyboard (see the Pro Tools manual for the connection of this type of device).

8.3.5 Saving presets

Once the session is saved, the state of the ARP2600 V is saved as it is, even if the programming does not correspond to any preset. For example, if you had been working on a preset P for which you had modified parameters (without having saved them under the name P), the next time the song is opened, the ARP2600 V will load preset P *with the modifications*.

The “Librarian Menu” of Pro Tools can of course be used with the ARP2600 V like any other plug-in. Nevertheless, it is recommended that you prioritize the internal menu of the ARP2600 V, and this for several reasons:

- The saved presets can be used in any sequencer, notably making user exchanges easier.
- They are compatible with the different versions of the ARP2600 V that will appear with its evolution.

8.3.6 Automation under Pro Tools

Automation functions with the ARP2600 V like any RTAS/HTDM plug-in (refer to the Pro Tools documentation for more details on plug-in automation). Preset changes (and thus cable movements) cannot be automated.

8.4 DXi™

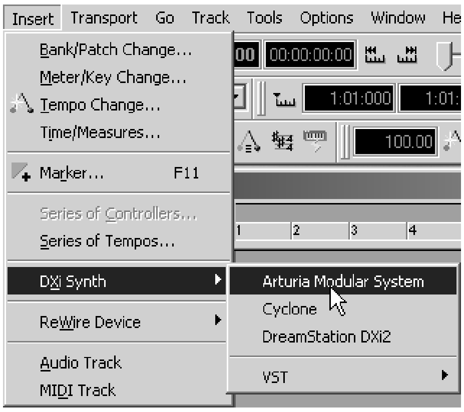
The ARP2600 V is compatible with the DXi protocol, and so can be used notably with Sonar, but also with any other sequencer that accepts DXi instruments.

8.4.1 Installation

During the installation, check the “DXi” box among the list of protocols that you wish to activate for the ARP2600 V on your computer, then simply follow the on-screen instructions until the installation is complete. Once the installation has finished, the ARP2600 V can be used as a DXi instrument.

8.4.2 Opening the instrument (SONAR™ 2.0)

In the “Insert” menu, open the “DXi Synth” sub-menu, and choose the ARP2600 V.



Opening the DXi instrument

- The “Synth Rack” then appears. To make the ARP2600 V graphical interface appear, double-click on its name in the “Synth Rack” window.

8.4.3 Connection to a MIDI track

So that the ARP2600 V can play the information coming from a MIDI track, you must choose a MIDI track and select the ARP2600 V as the MIDI output for this track using the menu in Sonar which is used for this:

Connecting a MIDI track to the ARP2600 V

The events played on a MIDI keyboard are then transmitted by your Sonar to the ARP2600 V. It is of course possible to record these MIDI events, and use all of the Sonar MIDI editing possibilities.

8.4.4 Saving presets

When the session is recorded, the state of the ARP2600 V is recorded as is, even if the programming does not correspond to any preset. For example, if you were working on preset “P1” on which you had modified the parameters (without having saved them under the name “P2”), the next time the song is opened, the ARP2600 V will load preset “P1” *with the modifications*.

8.4.5 Automation

Automation with Sonar functions simply by the reception and recording of Control Change type MIDI messages. You can configure the ARP2600 V MIDI message reception just as you do in standalone mode. The sequencer takes care of recording the continuous control transmitted to the ARP2600 V, and allows you to edit them.

8.5 Audio Unit

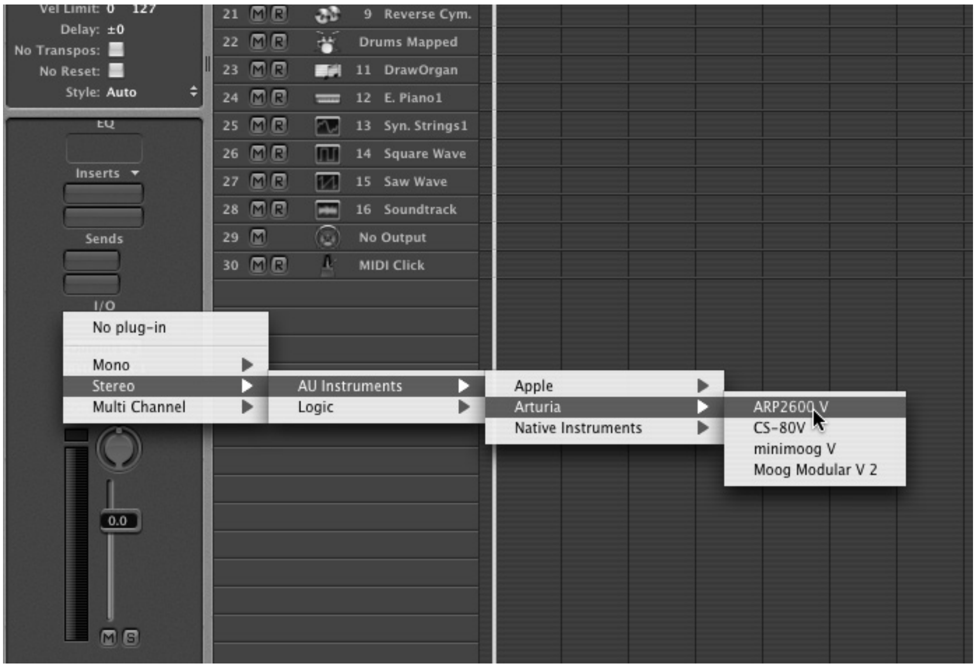
8.5.1 Installation

Audio Unit plugins are automatically installed in “/Library/Audio/Plug-Ins/Component”.

8.5.2 Using the plugin in Logic Audio

8.5.2.1 Adding an instrument

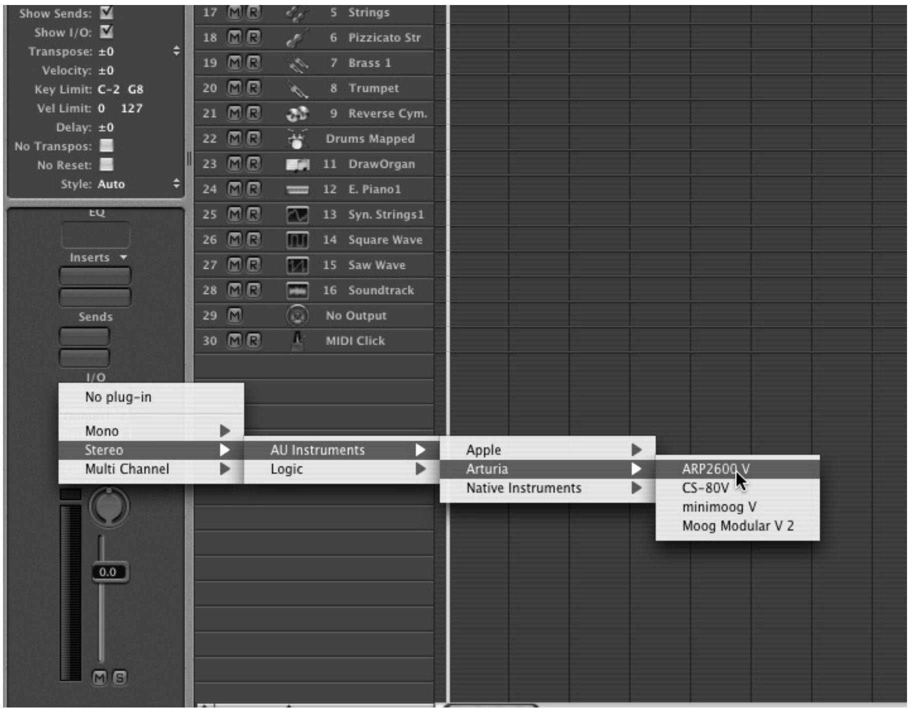
Select an instrument track. On corresponding the mixing table slice clic on the “I/O” button to get the plugins list, and select “Stereo -> AU Instruments -> Arturia -> ARP 2600 V”.



Adding an instrument in Logic Audio

8.5.2.2 Adding an effect

In the mixing table slice you wish to add an effect to, click on “Effect” to get the effects list, and then select “Stereo -> Audio Instruments -> Arturia -> ARP2600 V Efx”.



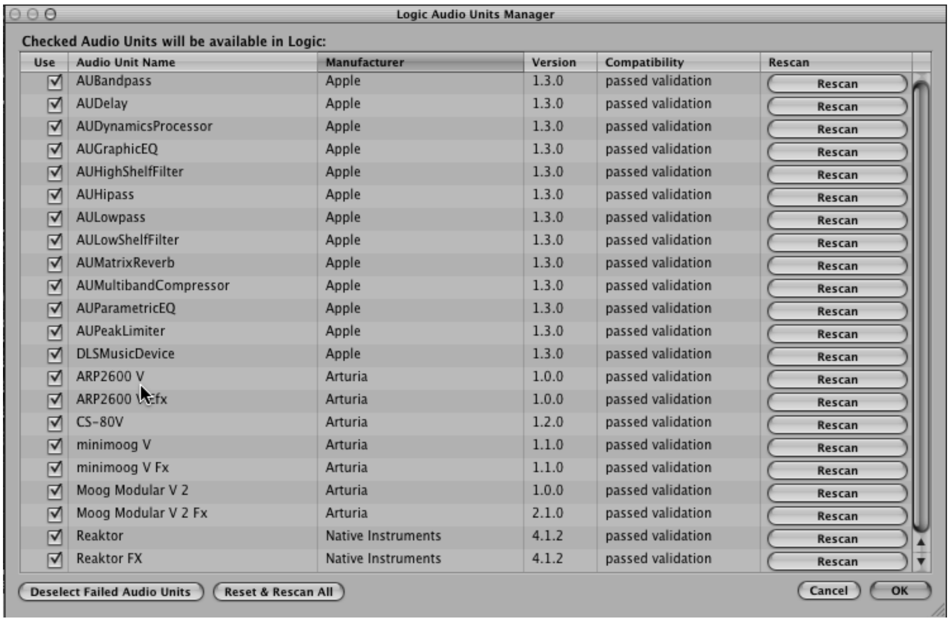
Adding an effect in Logic Audio

Since version 7, Logic has an AU Manager. To launch it, click on “Preferences -> Start Logic AU Manager”.

Starting Logic's AU Manager

This manager lets you see the list of available plugins, test their compatibility with Logic, and activate or deactivate them.

If one of your plugins is creating problems in Logic, check that it has passed the compatibility test and that it is enabled.



The AU Manager

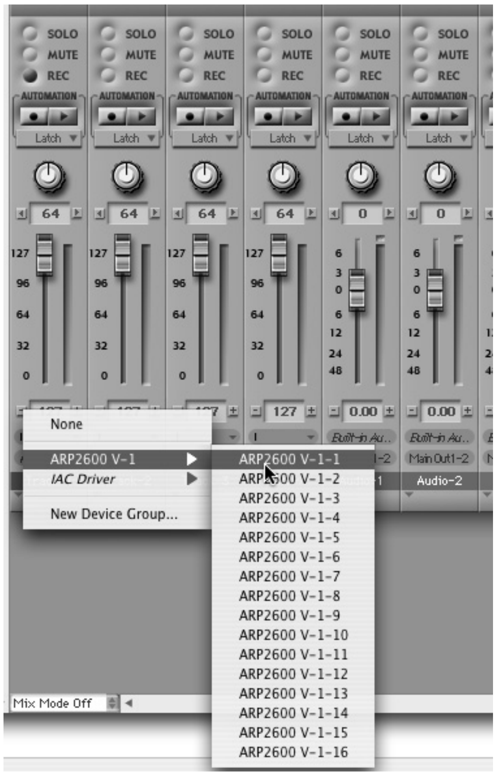
8.5.3 Using the plugin in Digital Performer

8.5.3.1 Adding an instrument

Clic on the menu item "Project -> Add Track -> Instrument Track -> ARP 2600 V".

Loading a new instrument

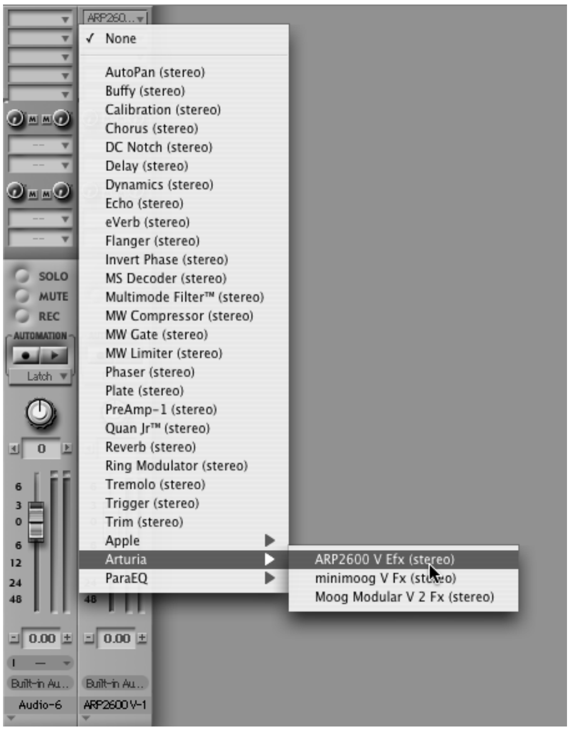
Once you have added this instrument, you can redirect a midi track toward it. In the connections menu of the midi track, select the instrument as well as the midi channel.



sélection du canal MIDI

8.5.3.2 Adding an effect

You can insert the ARP2600 V Efx effect on an audio track or another instrument track. To do that, just click on the effect menu of the corresponding mixing table slice, and select “Arturia -> ARP2600V Efx”.



Selecting the AR2600 V Efx

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