

USER MANUAL



ARTURIA®
YOUR EXPERIENCE • YOUR SOUND

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1 INTRODUCTION

Arturia would like to thank you for purchasing our synthesizer model: the ARP 2600 V. We are confident it will prove to be an extremely valuable addition to your music production studio. If you've purchased our products before, you know we pride ourselves in faithfully recreating the sound and feel of the original instruments, down to the smallest detail. ARP 2600 V is no exception to this rule.

And if this is the first of our products you have owned, you are in for a treat! The synthesizer upon which this model is based was the absolute pinnacle of analog synthesizer technology at the time, light-years ahead of the competition.

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 its competitor's modular system. 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 the Bob Moog's company. The competition between the two manufacturers can easily be seen when we observe the machines: The ARP, for example, has linear potentiometers, while its competitor 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 filter made by its competitor. In 1973, this one 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 ones equipping Bob Moog's synthesizers, what he admitted 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 the Bob Moog's monosynth 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 ARP's biggest successes. The instrument allowed the combination of two polyphonic violin sounds – a great innovation for the company – and 2 monophonic bass sounds.

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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 Arturia's secret ingredient: TAE®

TAE® (True Analog Emulation) is Arturia's outstanding technology dedicated to the digital reproduction of the analog circuits used in vintage synthesizers.

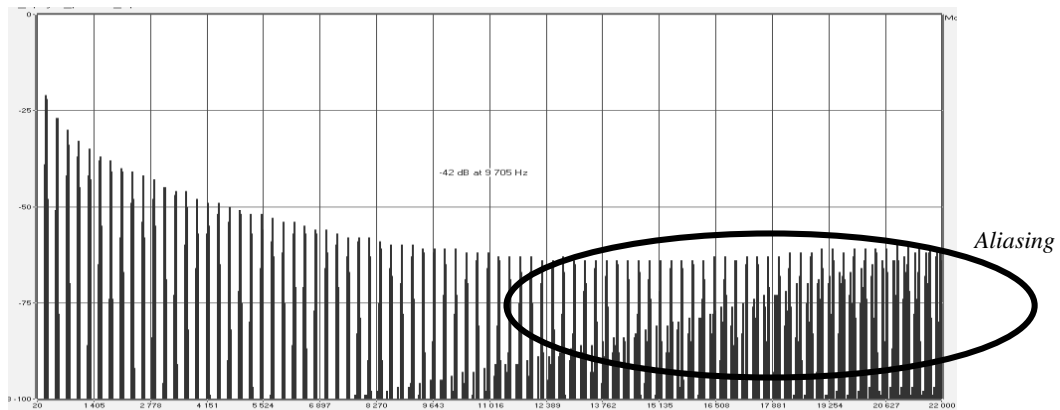
TAE®'s software algorithms result in spot-on emulation of analog hardware. This is why ARP 2600 V offers an unparalleled quality of sound, as do all of Arturia's virtual synthesizers.

TAE® combines three major advances in the domain of synthesis:

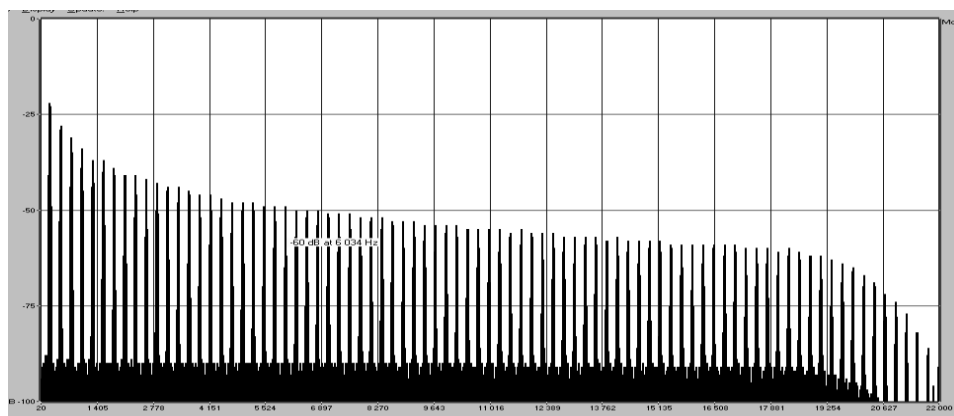
1.2.1 Aliasing-free oscillators

Standard digital synthesizers produce aliasing in high frequencies, especially when using Pulse Width Modulation (PWM) or Frequency Modulation (FM).

TAE® enables the generation of oscillators which are completely free of aliasing in all contexts (PWM, FM...), and at no extra CPU cost.



Linear frequency spectrum of a current well-known software synthesizer

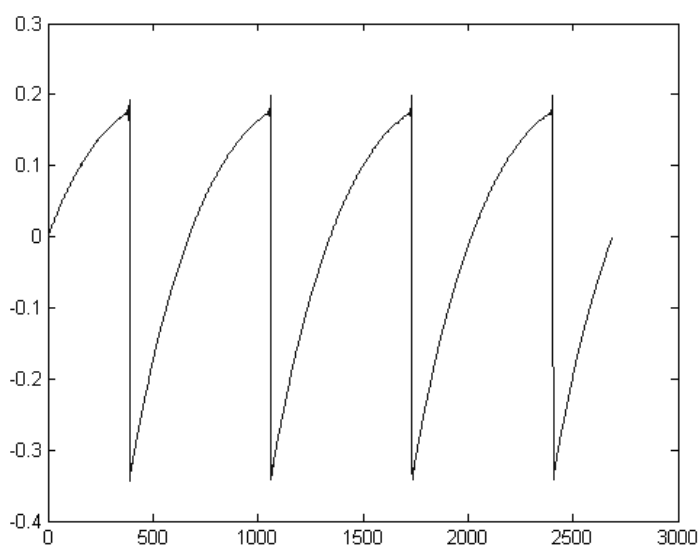


Linear frequency spectrum of an oscillator modeled with TAE®

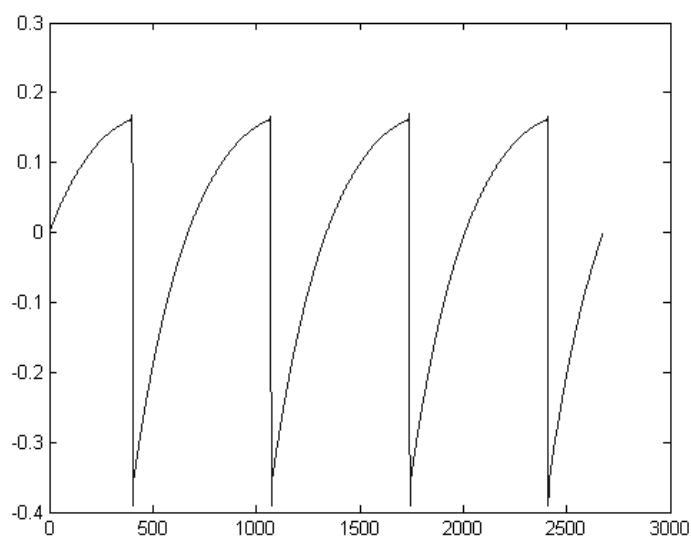
1.2.2 A better reproduction of analog oscillator waveforms

The waveforms produced by the oscillators in analog synthesizers are affected by the presence of a capacitor in the circuits. The discharge of a capacitor results in a slight 'bend' in the original waveform (most notably for sawtooth, triangular and square waveforms). TAE® reproduces the result of this capacitor discharge in software.

Below is the analysis of a waveform from one of the five original instruments Arturia's software emulates, followed by one made by TAE®. They are both equally deformed by the low-pass and high-pass filtering.



Temporal representation of the "sawtooth" waveform of a hardware synthesizer



Temporal representation of a "sawtooth" waveform reproduced by TAE®

What's more, the hardware analog oscillators were unstable. In fact, their waveforms vary 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 see why vintage synthesizers have such a typical sound.

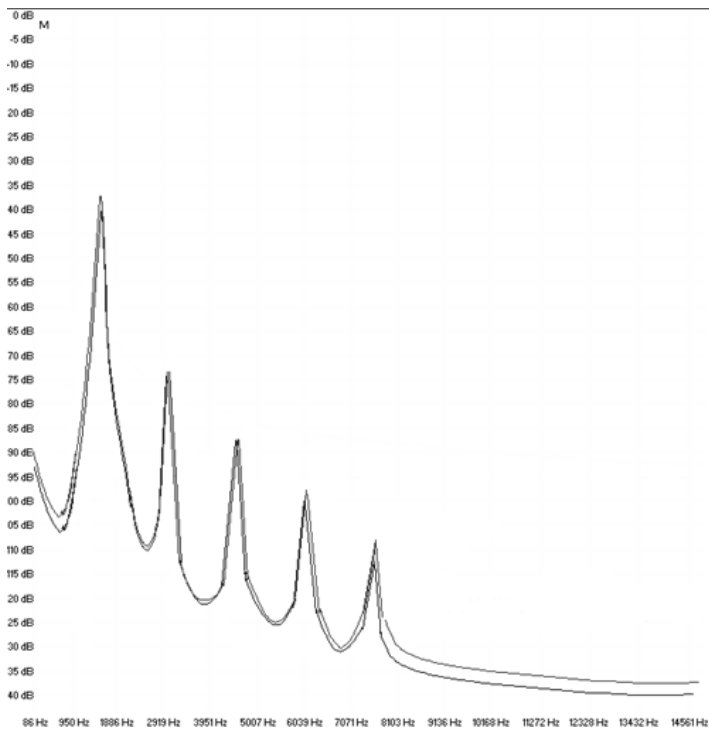
TAE® reproduces the instability of oscillators, resulting in a fatter and “bigger” sound.

1.2.3 Direct Filter Circuit Modeling

Due to advances in computer processing power, TAE® can now employ direct filter modeling techniques to achieve unprecedented accuracy in the emulation of a hardware synthesizer's filter. By modeling the operation of the individual hardware components of the filter circuit, the warm nuances synonymous with analog sounds are recreated.

The following graph shows a single example of direct circuit modeling in action. The peaks represent the generation of harmonics at multiples of the resonant frequency when a particular filter is in self oscillation mode. These harmonics are characteristic of hardware synthesizer filters and are due to the non-linear behavior inherent to their analog circuitry. Anomalies such as these add to the richness and warmth of the sound produced by the filter.

But you'll notice there are two lines on the graph: Those are the superimposed frequency domain plots for both one of Arturia's virtual instruments and the hardware filter being emulated. They are practically indistinguishable, both on the graph and to the human ear. The direct recreation of this analog circuitry causes the same characteristics of the sound to be present, thus giving the user a truly analog sound.



Comparison of harmonics generated by the filter circuits in self-oscillation of TAE® and a hardware synthesizer

So here's the bottom line: when you bring together a bunch of music lovers who also have a deep understanding of the characteristics of electronic circuits, you wind up with Arturia. And Arturia now offers you our most impressive software model yet, the ARP 2600 V.

We take great satisfaction in knowing this great synthesizer will help you explore previously unknown musical territory.

2 ACTIVATION AND FIRST START

2.1 Register and Activate

ARP 2600 V works on computers equipped with Windows 7 or later and Mac OS X 10.8 or later. You can use the stand-alone version or use ARP 2600 V as an Audio Units, AAX, VST2 or VST3 instrument.



Once ARP 2600 V has been installed, the next step is to register the software. The registration process will require you to enter the serial number and the unlock code you received with the product.

In order to proceed, go to this web page and follow the instructions:

<http://www.arturia.com/register>

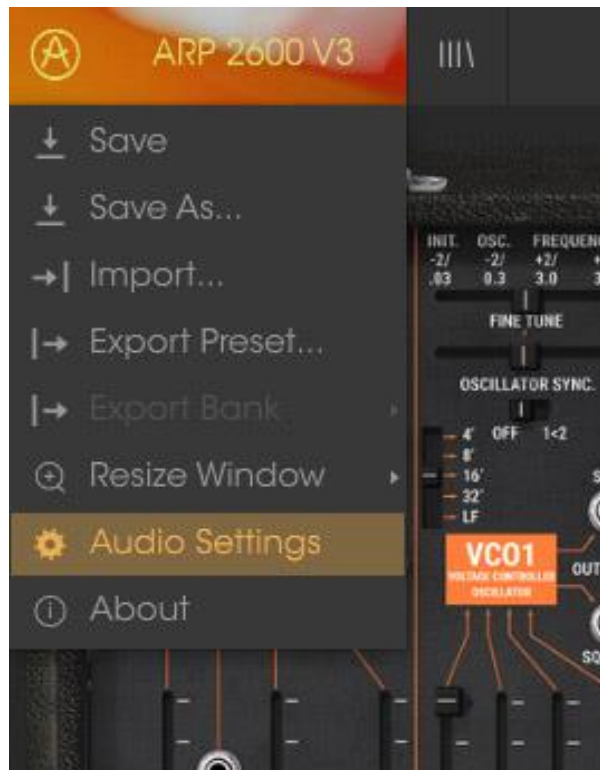
Note: If you don't have an Arturia account yet, you will need to create one. The process is quick, but it does require that you can access your email address during the registration process.

Once you have acquired an Arturia account you will be able to register the product.

2.2 Initial setup

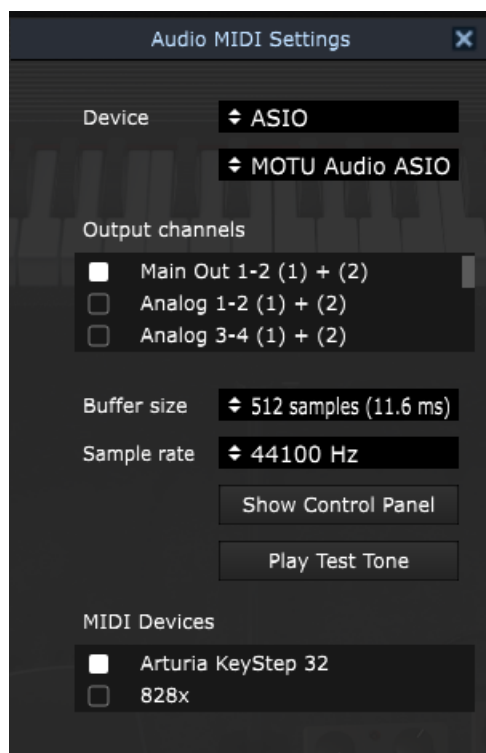
2.2.1 Audio and MIDI settings: Windows

At the top left of the ARP 2600 V application is a pull-down menu. It contains various setup options. Initially you will need to go to the menu and choose the Audio Settings option to get sound and MIDI flowing in and out.



ARP 2600 V main menu

You will then see the Audio MIDI settings window. This works in the same way on both Windows and Mac OS X, although the names of the devices available to you will depend on the hardware you are using.



Audio and MIDI settings window

Starting from the top you have the following options:

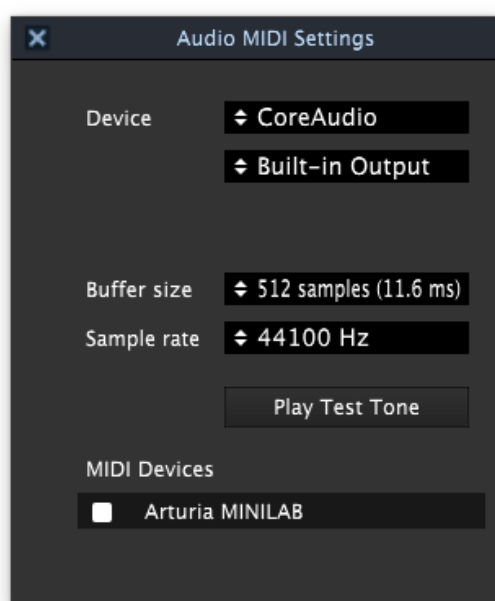
- **Device** lets you choose which audio driver you want to use to route sound out of the instrument. This might be your computer's own driver like Windows Audio, or an ASIO driver. The name of your hardware interface may appear in this field.
- **Output Channels** lets you select which of the available outputs will be used to route audio out. If you only have two outputs, only two will appear as options. If you have more than two you can select a specific pair of outputs.
- The **Buffer Size** menu lets you select the size of the audio buffer your computer uses to calculate sound. A smaller buffer means lower latency between pressing a key and hearing the note. A larger buffer means a lower CPU load as the computer has more time to think, but can result in a small latency. Find the optimum buffer size for your system. A fast, modern computer should easily be able to operate at 256 or 128 sample buffer size without creating pops or clicks in the sound. If you are getting clicks, try raising the buffer a little. The latency is displayed on the right hand side of this menu.
- The **Sample Rate** menu lets you set the sample rate at which audio is sent out of the instrument. The options here will depend on the capability of your audio interface hardware though even most computers' own hardware can operate at up to 48kHz which is perfectly fine. Higher sample rates use more CPU power so unless you have a good reason to

go up to 96kHz, then 44.1k or 48k is usually fine. The **Show Control Panel** button will jump to the system control panel for whatever audio device is selected.

- **Play Test Tone** helps you to troubleshoot audio issues by confirming whether sound can be heard through the correct device.
- Your connected MIDI devices will appear in the **MIDI Devices** area. Click the check box to accept MIDI from the device you want to use to trigger the instrument. In standalone mode, ARP 2600 V listens for all MIDI channels so there's no need to specify a channel. You can specify more than one MIDI device at once.

2.2.2 Audio and MIDI settings: Mac OS X

The process is very similar to initial setup for Windows and the menu is accessed in the same way. The difference is that OS X uses CoreAudio to handle audio routing and the audio device selection is made in the second dropdown menu. Apart from that, the options work the same way as described in the Windows section.



2.2.3 Using ARP 2600 V in plug-in mode

ARP 2600 V comes in VST, AU and AAX plug-in formats for use in all major DAW software such as Cubase, Logic, Pro Tools and so on. You can load it as a plug-in instrument and its interface and settings work the same way as in standalone mode, with a couple of differences.

- You can automate numerous parameters using your DAW's automation system.
- You can use more than one instance of ARP 2600 V in a DAW project. In standalone mode you can only use one at once.
- You can route ARP 2600 V's audio outputs more creatively inside your DAW using the DAW's own audio routing system.

3 USER INTERFACE

In this chapter we will give an overview of the features available to you with ARP 2600 V. As with every Arturia product, we have gone to great lengths to make the use of this software instrument as simple and as much fun as possible, while also striving to make sure you never run out of new things to do with it as your knowledge expands. After reading this chapter you should be ready to delve as deeply into the workings of ARP 2600 V as you would like.

3.1 The virtual keyboard

The virtual keyboard lets you play a sound without connecting an external MIDI device; just click a key to hear the active Voice. Drag the cursor across the keys to hear a glissando.



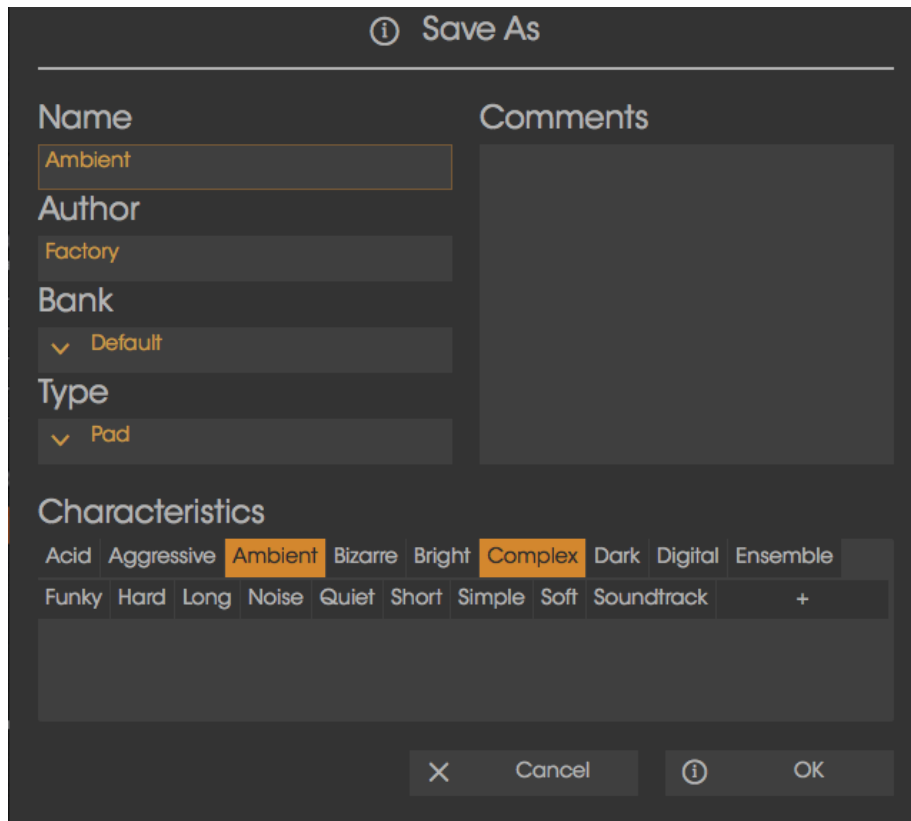
The ARP 2600 V virtual keyboard

3.2 Toolbar

The toolbar that runs along the top edge of the instrument both in standalone and plug-in mode provides access to many useful features. Let's look at them in detail. The first seven of these options can be found by clicking on the ARP 2600 V section at the very top left hand corner of the instrument window.

3.2.1 Save Preset

The first option lets you save a preset. If you select this, you are presented with a window where you can enter information about the preset. In addition to naming it you can enter the author name, select a bank and type and select some tags that describe the sound. This information can be read by the preset browser and is useful for searching the preset banks later. You can also enter freeform text comments in the Comments field, which is handy for providing a more detailed description.



The Save Preset window

3.2.2 Save Preset As...

This works in the same way as the Save command, but lets you save a copy of the preset instead of saving over the original. It's useful for creating variations on patches but still keeping individual copies of each one.

3.2.3 Import preset

This command lets you import a preset file, which can be either a single preset or an entire bank of presets. Both types are stored in the .arpx format.

After selecting this option, the default path to these files will appear in the window, but you can navigate to whichever folder you are using.



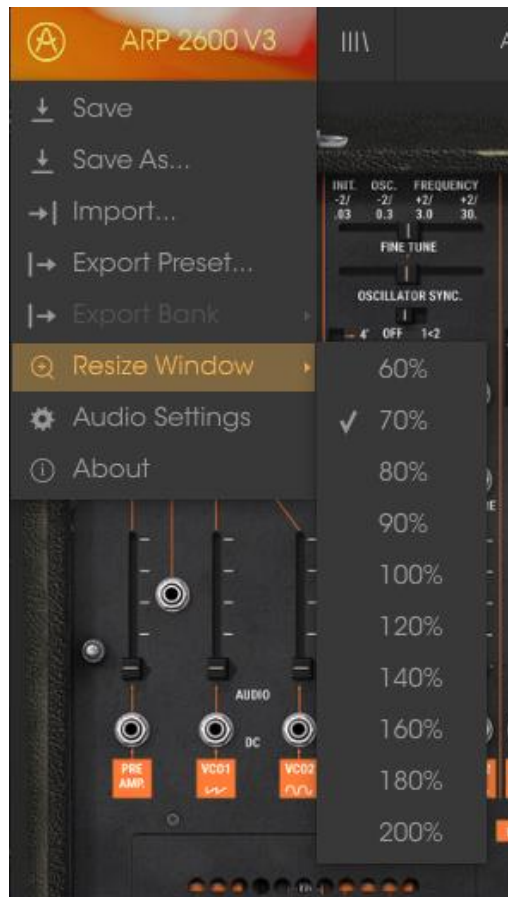
INT.	OSC.	FREQUENC	
-2/	-2/	+2/	+2
.83	0.3	3.0	30

3.2.4 Export preset

You can export and share a single preset using this command. The default path to these files will appear in the window, but you can create a folder at another location if you like.

3.2.5 Resize window options

The ARP 2600 V window can be resized from 60% to 200% of its original size without any visual artifacts. On a smaller screen such as a laptop you might want to reduce the interface size so it doesn't dominate the display. On a larger screen or a second monitor you can increase the size to get a better view of the controls. The controls work the same at any zoom level but the smaller ones can be harder to see at the smaller magnification values.



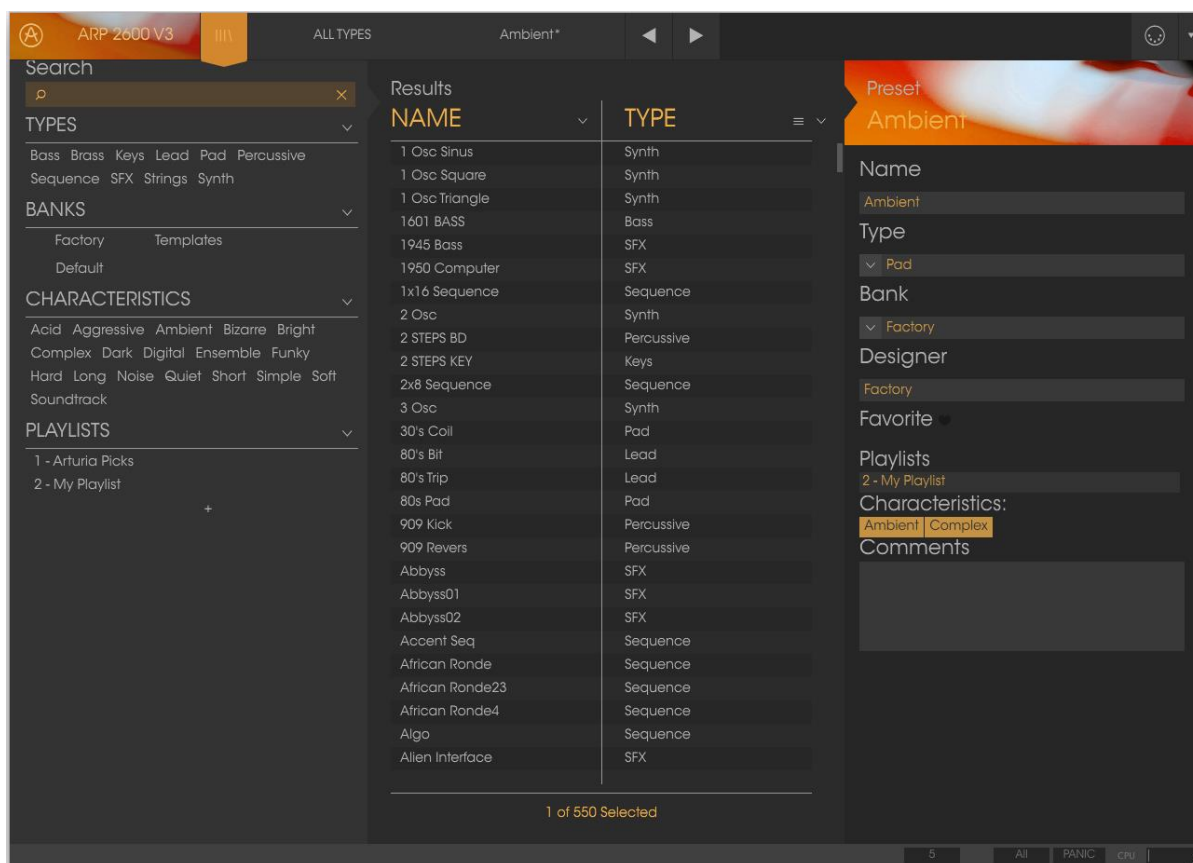
The Resize Window menu

3.2.6 Audio settings

Here you manage the way the instrument transmits sound and receives MIDI. See section 2.2 of the manual for full details on this.

3.2.7 Preset browser overview

The Preset browser is invoked by clicking the toolbar button that has four vertical lines. See section 3.3 of the manual for full details on this. The Filter, name field and left / right arrows in the toolbar all assist with preset selection.



The Preset Browser

3.2.8 MIDI Learn assignment

The MIDI plug icon at the far right side of the toolbar places the instrument into MIDI learn mode. Parameters that can be assigned to MIDI controls will be shown in purple, and the idea is that you map physical buttons, knobs, faders or pedals from hardware MIDI controllers to specific destinations inside the instrument. A typical example might be to map a real expression pedal to the virtual volume pedal, or buttons on a controller to the effect switches so you can change the sound from your hardware keyboard.



MIDI Learn mode

3.2.8.1 Assigning / unassigning controls

If you click on a purple area you'll put that control into learning mode. Move a physical knob or fader and the target goes red, indicating that a link has been made between the hardware control and the software parameter. There's a popup window that displays which two things are being linked and a button to unassign the two from each other.



Preamplifier knob selected and assigned

3.2.8.2 Min / Max value sliders

There are also minimum and maximum value sliders that you can use to restrict the parameter change range to something other than 0%-100%. For example, you might want the filter cut-off be controllable via hardware from 30% to 90%. If you made this setting (Min set to 0.30 and Max set to 0.90) your physical knob would be unable to alter the volume lower than 30% or higher than 90%, no matter how far you turned it. This is very useful for making sure you can't accidentally make the sound too quiet or too loud when performing.

In the case of switches which only have two positions (on or off), those would normally be assigned to buttons on your controller. But it is possible to toggle those with a fader or other control if you like.

3.2.8.3 Relative control option

The final option in this window is a button labelled "Is Relative". It is optimized for use with a specific type of control: one which sends only a few values to indicate the direction and speed at which a knob is turning, as opposed to sending a full range of values in a linear fashion (0-127, for example).

To be specific, a "relative" knob will send values 61-63 when turned in a negative direction and values 65-67 when turned in a positive direction. The turn speed determines the parameter response. Refer to the documentation of your hardware controller to see if it has this capability. If so, be sure to switch this parameter on when setting up its MIDI assignments.

When configured this way, movements of the physical control (usually a knob) will change the software parameter by starting at its current setting, rather than

being an “absolute” control and snapping it to some other value as soon as you start to move it.

This can be a great feature when controlling things like volume, filter, or effect controls, since you won't usually want them to jump massively out of their current setting as soon as you start to modify them.

3.2.8.4 Reserved MIDI CC numbers

Certain MIDI Continuous Controller (MIDI CC) numbers are reserved and cannot be reassigned to other controls. These are:

- PitchBend
- Ctrl Mod Wheel (CC #1)
- AfterTouch
- Ctrl Expression (CC #11)
- Ctrl All Notes Off (CC #123)
- Ctrl Omni Mode Off (CC #124)
- Ctrl Omni Mode On (CC #125)
- Ctrl Poly Mode Off (CC #126)
- Ctrl Poly Mode On (CC #127)

All other MIDI CC numbers may be used to control any assignable parameter in ARP 2600 V.

3.2.9 MIDI controller configuration

There's a small arrow at the far right hand side of the toolbar that deals with MIDI controller configurations. This allows you to manage the different sets of MIDI maps you may have set up for controlling the instrument's parameters from MIDI hardware. You can copy the current MIDI assignment setup or delete it, import a configuration file or export the currently active one. This is a quick way to set up different hardware MIDI keyboards or controllers with ARP 2600 V without having to build all the assignments from scratch each time you swap hardware.



3.2.10 The lower toolbar

3.2.10.1 Current control value

At the left hand side of the lower toolbar you will see a readout showing the value or state of whatever control you are modifying. It will also display the current value of a parameter without editing it: just hover the cursor over the related control and the value will appear as pictured below.



3.2.10.2 *Midi Channel Setting*

At the right hand side of the lower toolbar are three small windows. The first one on the left indicates the current MIDI Channel setting. Click on it and it will expand to show the full range of values you can select (All, 1-16).

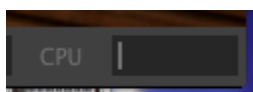


3.2.10.3 *Panic button and CPU meter*

The Panic button can be pressed to reset all MIDI signals in the event of stuck notes or other issues. The Panic button is also MIDI-assignable.

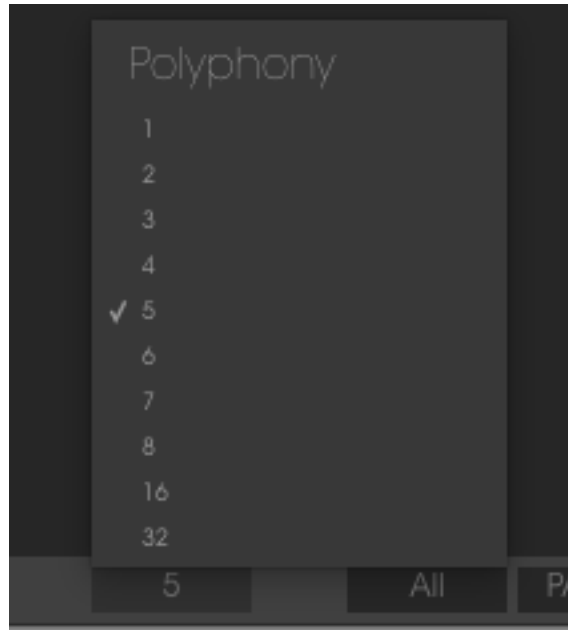


The CPU meter is used to monitor how much of your computer's CPU is being used by the instrument.



3.2.10.4 *Maximum Polyphony*

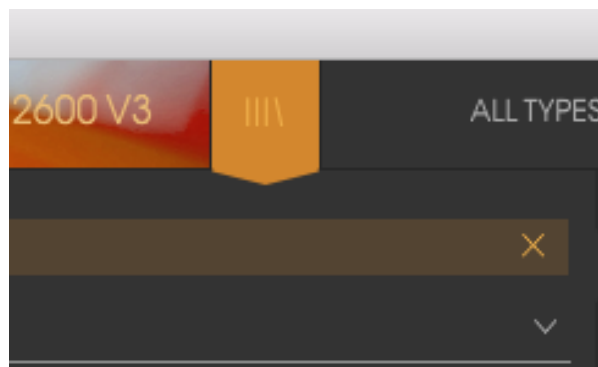
By clicking this button, you will be able to adjust the upper limit for the number of voices played the ARP 2600 V. It can be set from 1 to 32. Having lower setting will result in less CPU being used. Setting the number too low can create situation where the voices cut off and create unnatural sustains. The key is to find a balance that you and your computer can both live with.



3.3 The Preset Browser

The preset browser is how you search, load and manage sounds in ARP 2600 V. It has a couple of different views but they all access the same banks of presets.

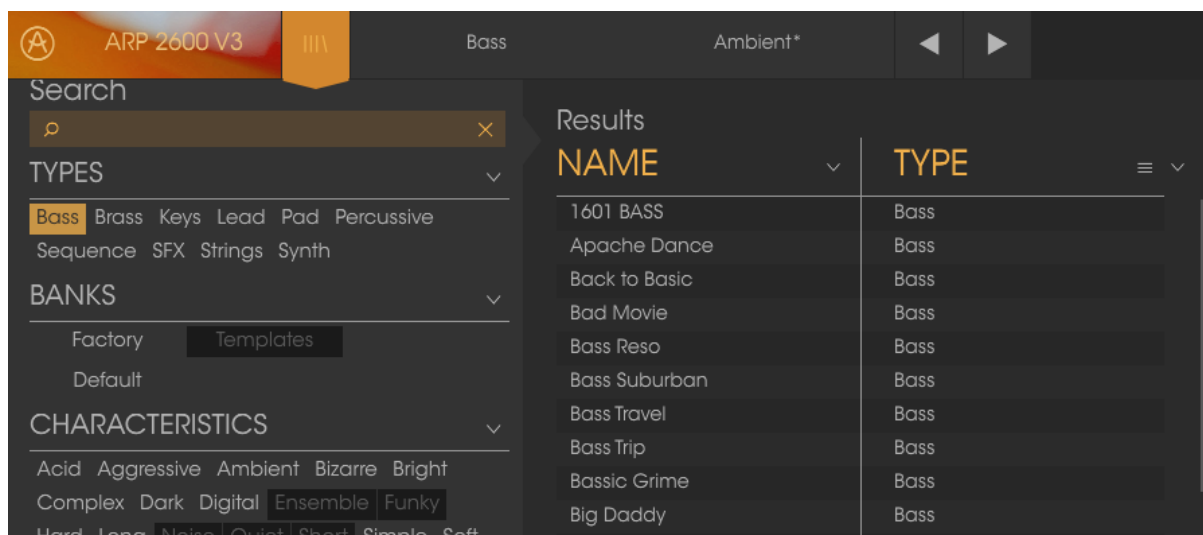
To access the search view, click on the browser button (the icon looks a bit like books on a library shelf).



The Preset Browser button

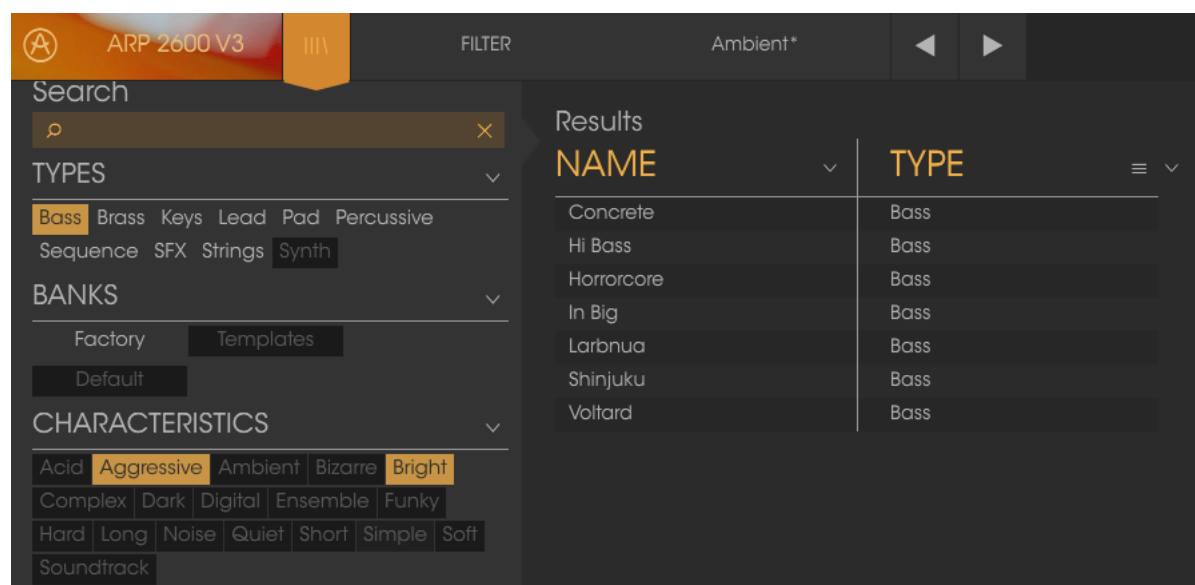
3.3.1 Searching presets

The Search screen has a number of sections. By clicking on the Search field at the top left you can quickly enter any search term to filter the preset list by patch name. The Results column is updated to show the results of your search. Press the X button in the search field to clear the search.



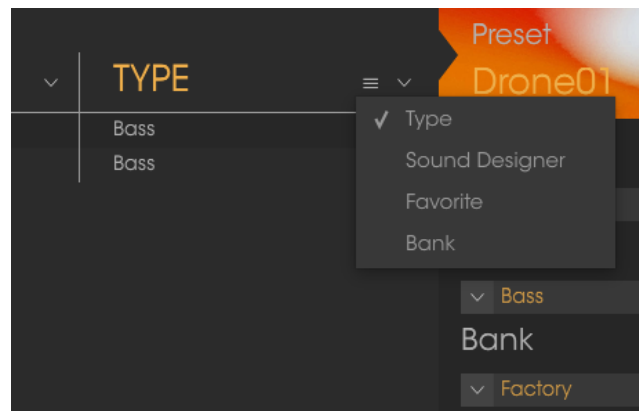
3.3.2 Using tags as a filter

You can also search using different tags. So for example by clicking on the Bass option in the Types field you can show only presets that match that tag. The tag fields can be shown or hidden by using the small down arrow buttons in their title fields. Results columns can be sorted by clicking the same arrow button in their own section.



You can use multiple search fields to perform narrower searches. So by entering a text search and also specifying type, bank and characteristics options you could see only the presets that match those exact criteria. Deselect any tag in any area to remove that criteria and widen the search without having to go back and start again. Using “Ctrl + click” (Windows) or “Cmd + click” (Mac) will allow you to select multiple elements in the same area.

The second Results column can be switched to show Type, Sound Designer, Favorite or Bank tags depending on how you like to search. Click on its options menu button just next to its sort arrow.

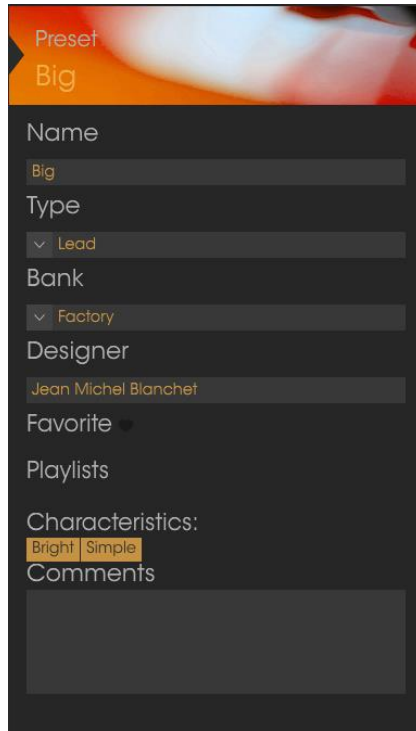


3.3.3 The Preset Info section

The Info column on the right of the search field shows you information about any preset. The information for User presets may be changed here: Name, Type, Favorite, etc.

However, if you want to alter the information for a Factory preset you must first use the Save As command to re-save it as a User preset. After this the Info section will gain Edit and Delete buttons at the bottom of the window.

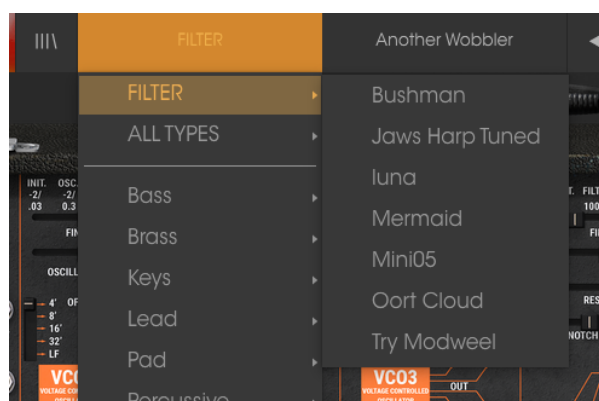
Click Edit and then make the desired changes, either by typing in one of the fields or by using a pull-down menu to change the Bank or Type. You can even add new Characteristics by clicking the + sign at the end of that list. Click Save when you are done.



3.3.4 Preset selection: other methods

The pull-down menu to the right of the Search menu provides a different way to select presets. The first option in this menu is called Filter, and it will display the presets that fit the search terms you used in the Search field. So if you searched for “Love” in the main search area, the results of that search will appear here.

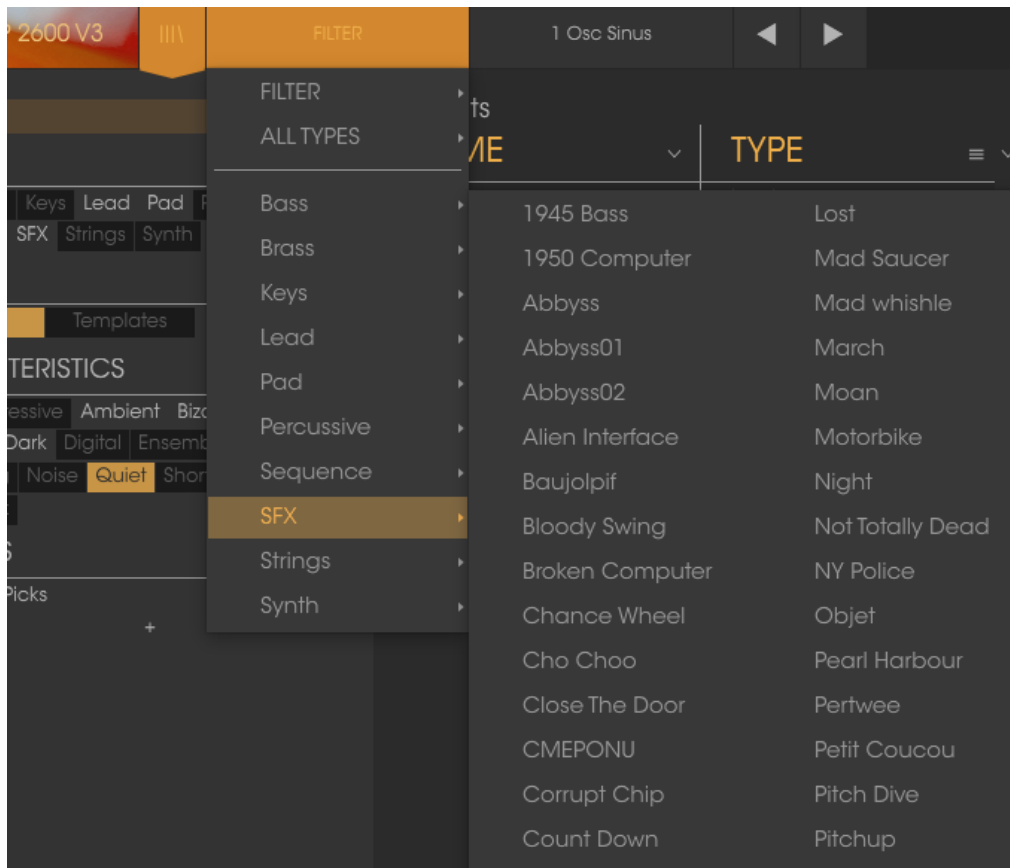
Similarly, if you previously selected Type: Lead in the Search field you would see the results of that search in this area instead.



Filter results may differ based on Search criteria

Selecting the All Types option in the pull-down menu will bypass the Search criteria and show the entire list of presets.

The Categories below the line also ignore the Search criteria and display the presets based on their Type.



3.3.4.1 Selecting a preset by its Type

Clicking on the name field in the center of the toolbar will show you a list of all available presets. The list will also take into account any selections you have made in the Search field. So if you have pre-selected a Characteristic this shortcut menu will only show you presets that match that tag.

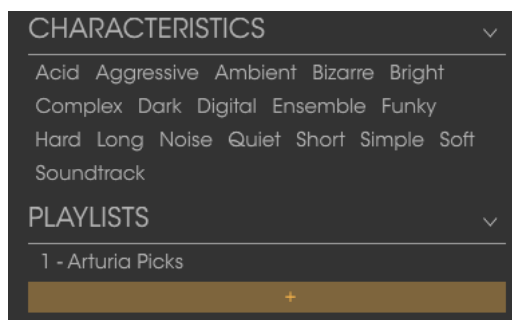
The left and right arrows in the toolbar cycle up and down through the preset list: either the full list, or the filtered list that resulted from the use of one or more search terms.

3.3.5 Playlists

In the lower left corner of the Preset Browser window is a feature titled Playlists. This is used to collect presets into different groups for different purposes, such as a set list for a particular performance or a batch of presets related to a particular studio project.

3.3.5.1 Add a playlist

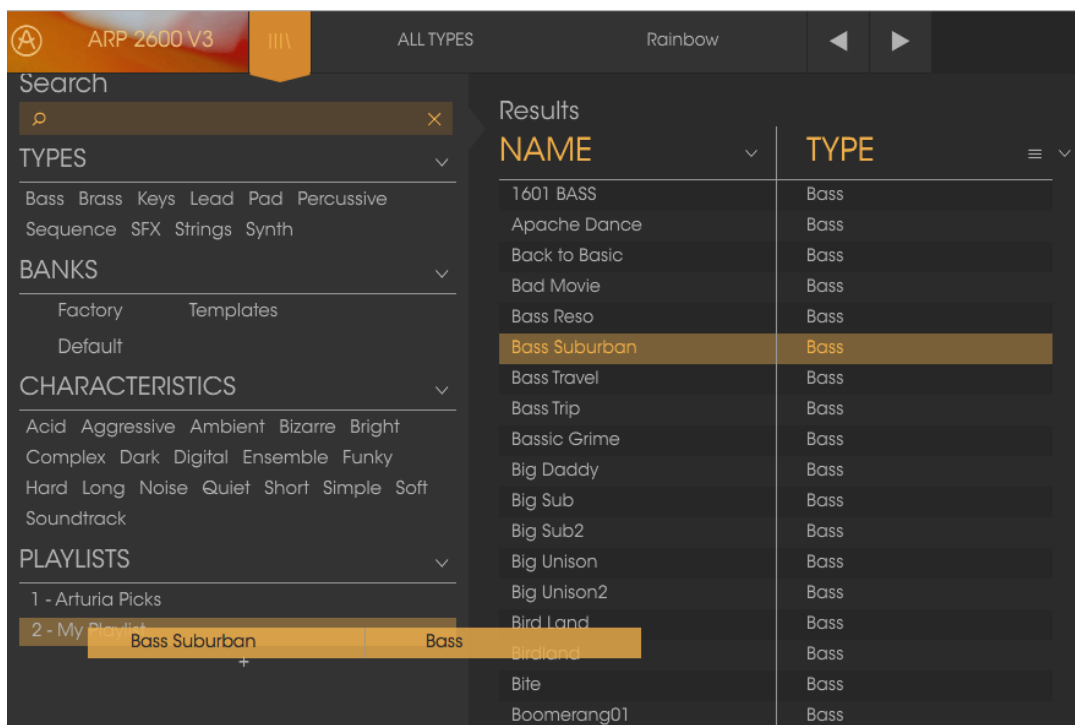
To create a playlist, click the plus sign at the bottom:



Give the playlist a name and it will appear in the Playlists menu. You can rename the playlist at any time; just click the pencil icon at the end of its row.

3.3.5.2 Add a preset

You can use all of the options in the Search window to locate the presets you want to have in your playlist. Once you have found the right preset, click and drag it onto the playlist name.



Click and drag from the Search Results list onto one of the playlists
To view the contents of a playlist, click on the playlist name.

3.3.5.3 Re-order the presets

Presets may be reorganized within a playlist. For example, to move a preset from slot 2 to slot 4, drag and drop the preset to the desired location.

This will move the preset into the new location.

3.3.5.4 *Remove a preset*

To delete a preset from a playlist, click the x at the end of the preset row.
Click the X to remove a preset from a playlist

3.3.5.5 *Delete a playlist*

To delete a playlist, click the x directly to the right of the playlist name.
Click the X to delete a playlist.

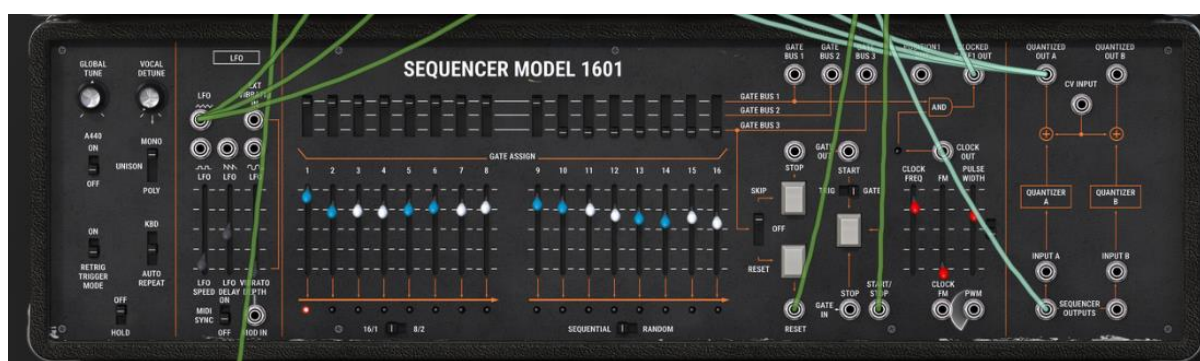
3.4 Overview of the 3 sections of the ARP2600 V

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

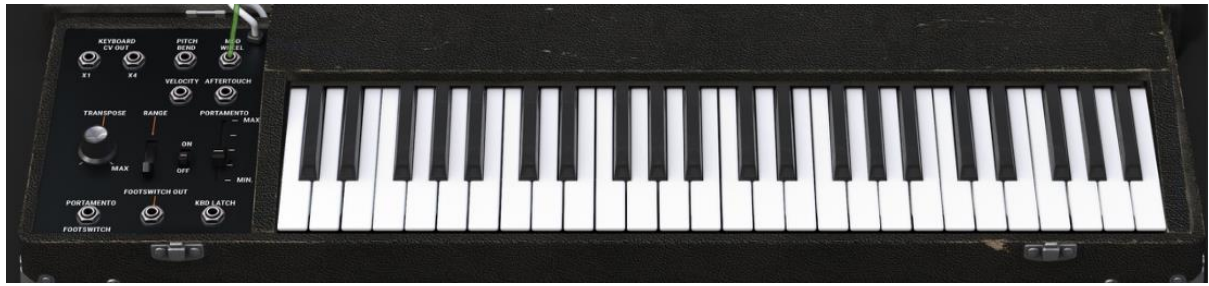
From top to bottom:



The synthesizer section



The Sequencer / LFO / General settings section



And the keyboard section

You can scroll up and down the ARP 2600 V interface thanks to the wheel of your mouse or by using the up & down arrows of your computer keyboard.

3.5 Overview of the Synthesizer section

3.5.1 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 waveforms (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.

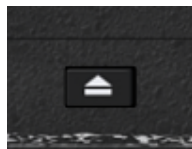


Synthesis parameters

3.5.2 The effects

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

To open it, click on the “open” button under this grid.



Open the effects grid

3.5.2.1 Chorus

Chorus is used to copy your sound, and slightly detune the copy, to give it more depth.



The chorus effect settings

3.5.2.2 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.



The Delay effect settings

3.5.1 “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.

It is located behind the grid in the bottom-right corner of the synthesizer section. You can make it appear by using the open button below the grid.



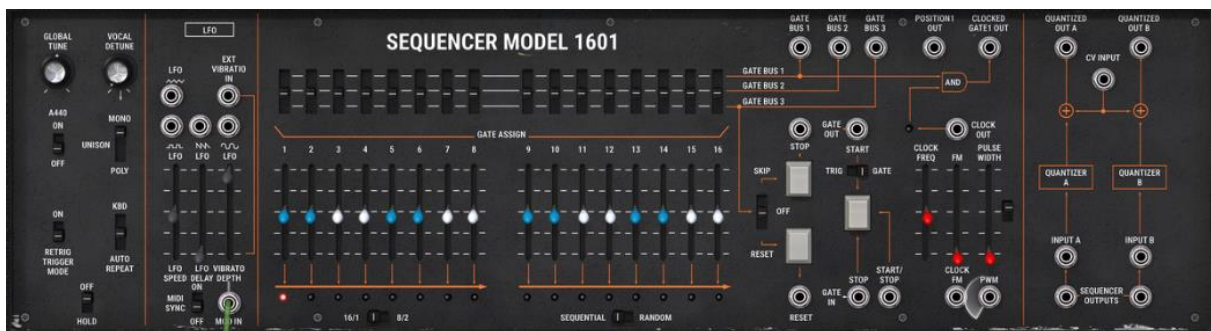
Open the tracking grid...

3.1 Overview of the Sequencer / LFO / General settings section

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 Synthesizer 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.

3.1.1 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.

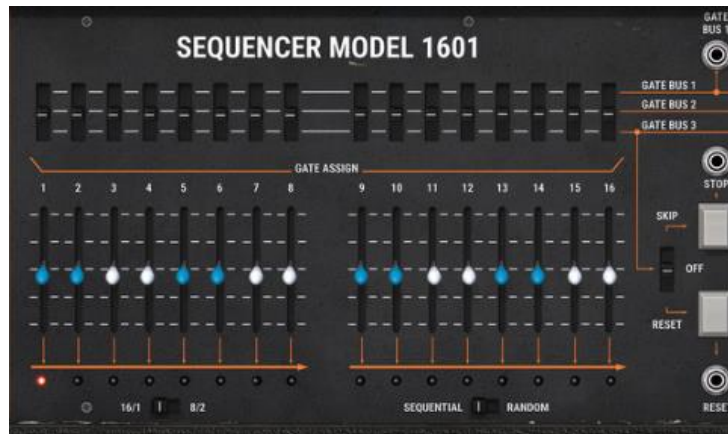


The ARP sequencer

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

3.1.2 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

4 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.

4.1 Sound programming cabinet

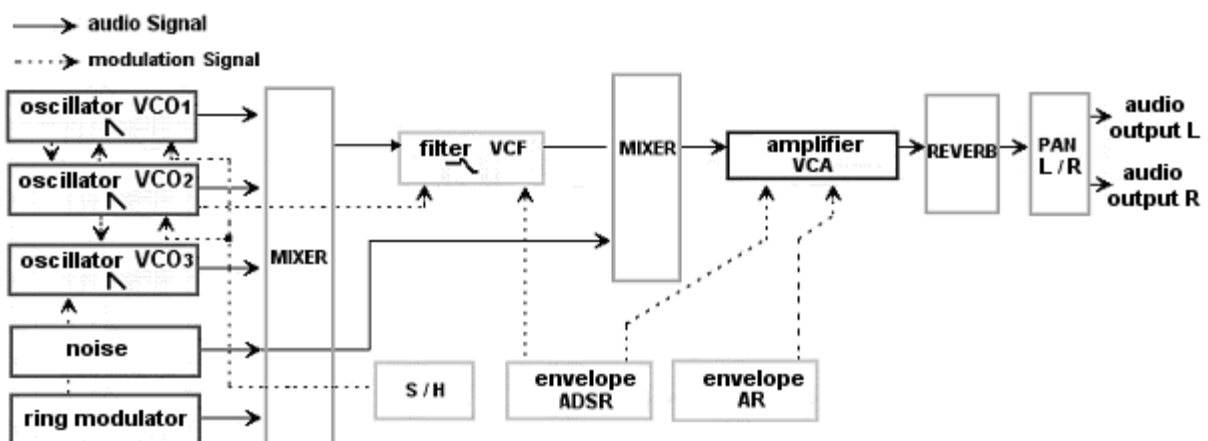
4.1.1 Description

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:

- oscillators, which can also be used as modulation sources. (VCO)
- 1 multimode filter. (VCF)
- 1 amplifier (VCA)
- 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
- mixer / lag module
- 2 effects (chorus / delay)



4.1.2 The 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 waveforms. 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 waveforms that can be used simultaneously.

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



The 3 oscillators settings

4.1.2.1 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 waveform 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
- **Oscillator 2 Sin:** pre-cabled for the manual setting of the modulation by sine waveform of oscillator 2 (VCO2 Sin)

4.1.2.2 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 waveform outputs: triangle, sine, sawtooth and square

Impulse width: Impulse width for "Sawtooth", "Square", "Triangle" waveforms

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 waveform 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)

4.1.2.3 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 waveforms

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 waveform 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” waveforms of oscillators 2 and 3, can be modified with the “pulse width” potentiometer.

Oscillator 1 possesses two audio outputs for the sawtooth and square waveforms. 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 2600V2. 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

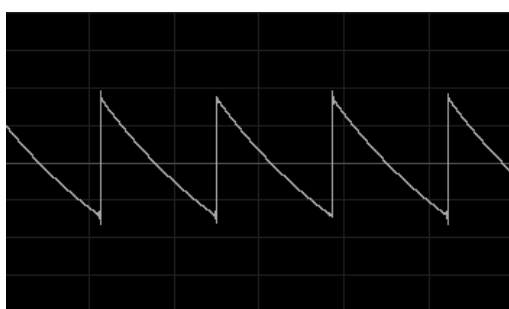
perform modulation using the lower CPU power compared to the other positions.

To set the modulation rate, use the linear potentiometer situated above the corresponding jack.

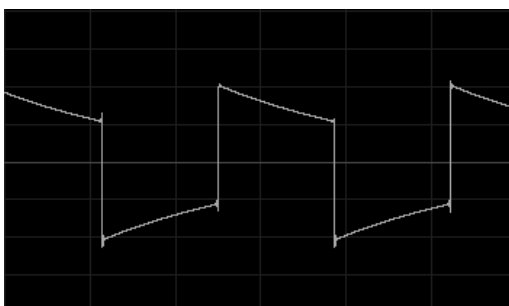
You can also connect another source of modulation to each input. This considerably widens the possibilities for sound creation.

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

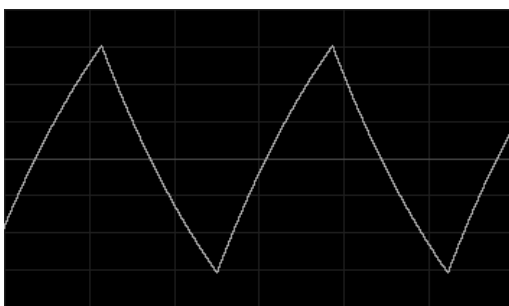
Here are graphical representations of the different waveforms used by ARP 2600V oscillators:



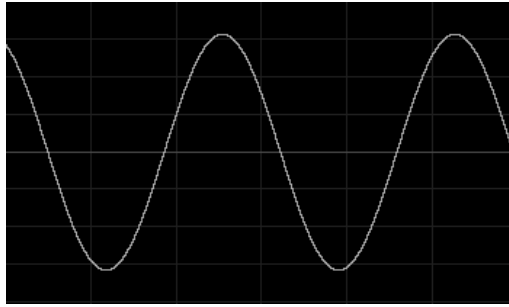
Sawtooth



Square



Triangle



Sine

4.1.3 The 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 waveform of oscillator2
- **ADSR Env:** re-cabled for the manual setting of the modulation by ADSR envelope.

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.

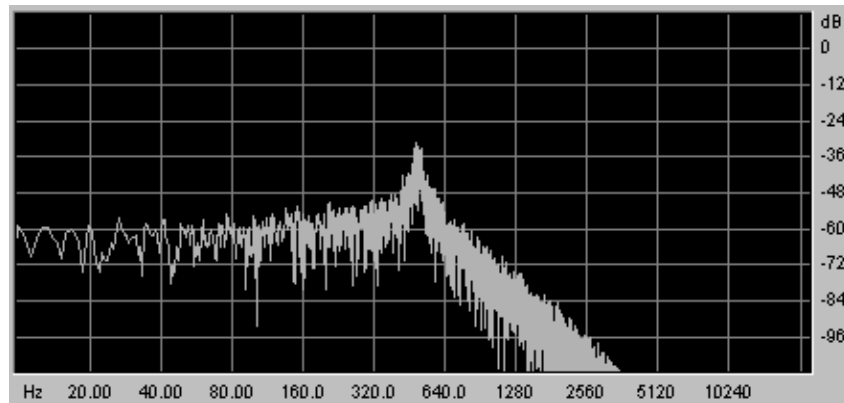
Only the cut-off frequency can be dynamically modulated by one of the 3 modulation inputs.

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

4.1.3.1 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

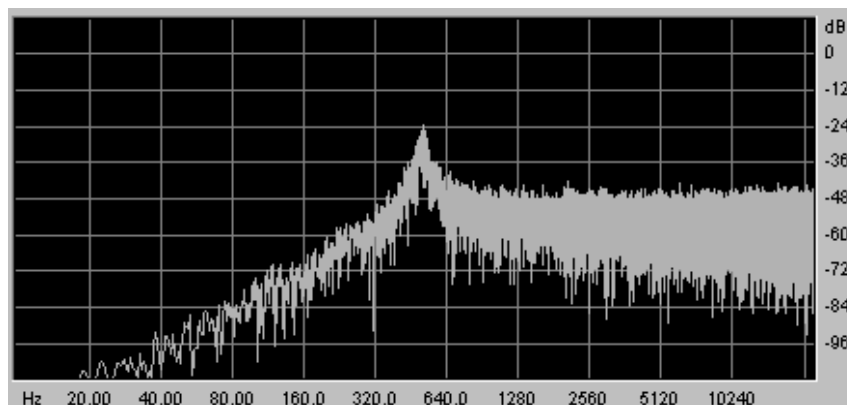
The four other filtering modes didn't exist on the original ARP2600 but existed on the ARP2500 modular systems. They all use a filtering slope at 12 dB/ octave. These modes were added to increase the possibilities for sound creation on the ARP 2600 V 2.5.

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

The low pass 12 dB filter 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)

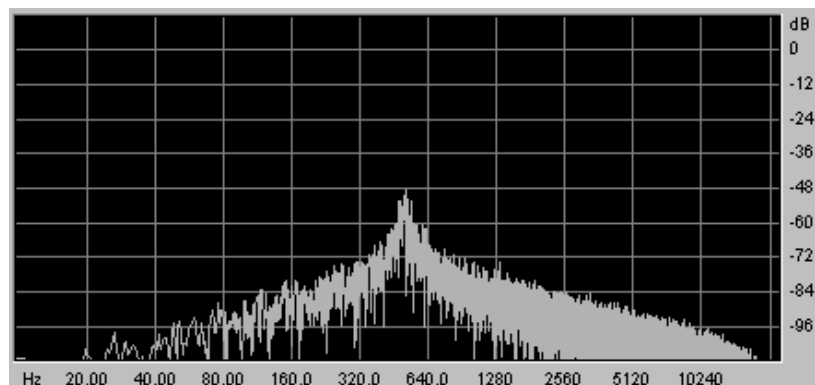
The high pass filter is the opposite of the low pass filter. It eliminates the frequencies above the cut-off frequency.



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.

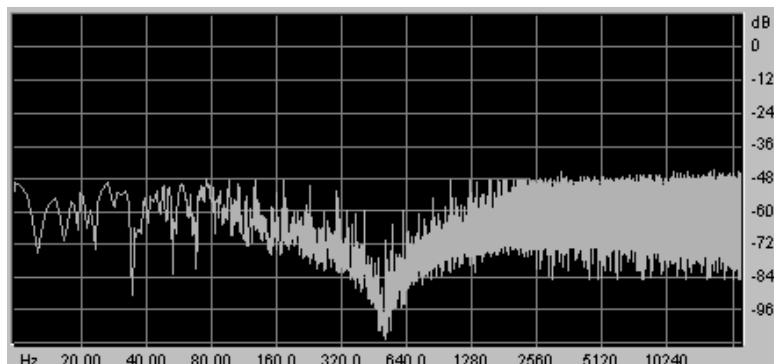


The band pass filter

- 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.



The notch filter

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.

4.1.4 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

4.1.4.1 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

4.1.4.2 AR

Attack (Attack time): Sets the attack time

Release (Release time): Sets the release time

4.1.4.3 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.

4.1.5 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).

4.1.6 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.

4.1.7 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: Smooths 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” potentiometers: Potentiometer to set the volume of 2 input signals (A and B for example)

4.1.8 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 waveform 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”) : Audio or modulation external signal input connection jack. (pre-cabled to the noise module)

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.

4.1.9 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 waveform 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

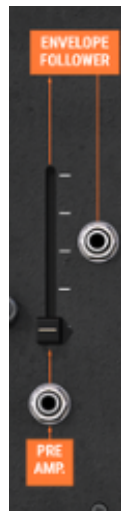
External clock input (Ext clock In): External; clock connection jacks (pre-cabled to the Sample and Hold clock)

4.1.10 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 for the audio signal

Level (Level): Sets the amount level of the envelope follower

Gate out (gate out): Trigger signal output connection jacks

4.1.11 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

4.1.12 “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

4.1.12.1 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.

Curve viewer: Displays the 4 tracking curves. Clicking on it opens the Edit interface.

4.1.12.2 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.

4.1.13 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

4.1.14 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.

4.1.15 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 x4 (KYBD CV output): Output connection jacks for key follow x1 (tempered key follow) and x4

4.2 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 / retrigger

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

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

4.3 Low frequency oscillator (LFO)

The use of 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 waveform.

Square output (LFO Square): Audio output of the LFO square waveform.

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

Sine output (LFO Sine): Audio output of the LFO sinusoid waveform

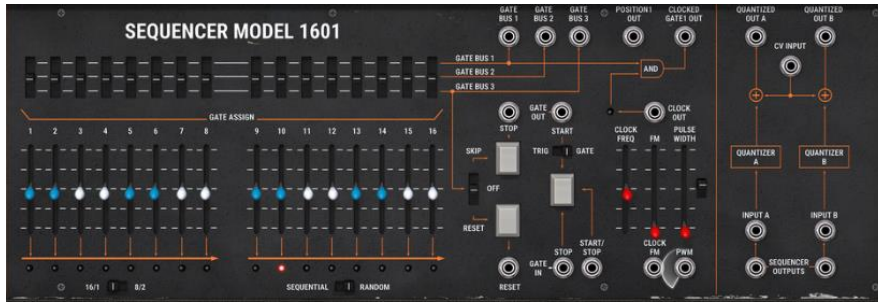
External vibrato input

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

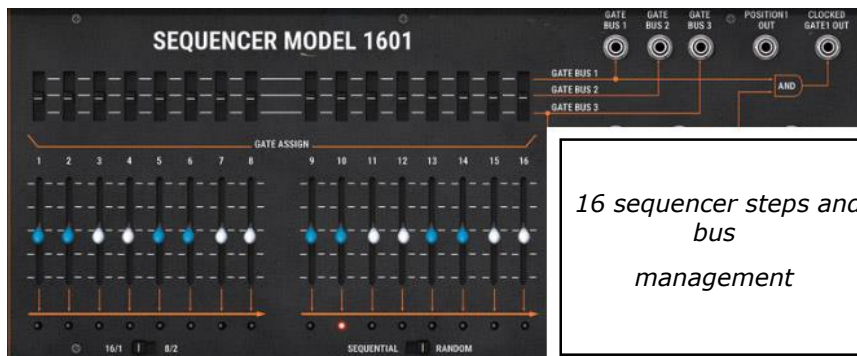
4.4 The ARP sequencer

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.



2x8 output levels (1 ... 16 faders): Sets the modulation or tuning level for sequence steps.

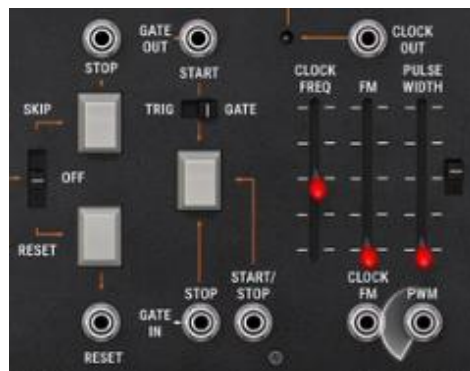
Bus 1, 2 and 3 (switches Gate bus 1, 2, 3): Trigger selector switch for steps by bus 1, 2 or 3

Bus inputs 1, 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.

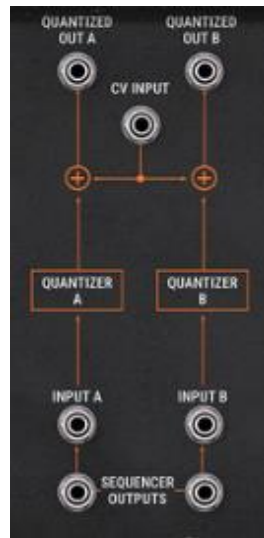
Clock FM (FM): Sets the modulation rate for the clock frequency.

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)

5 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, Oberheim, Yamaha, Buchla, Sequential Circuits (Prophet series), Roland, Korg (MS and PS series), Bob Moog's creation 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.

5.1 The three main elements

5.1.1 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.



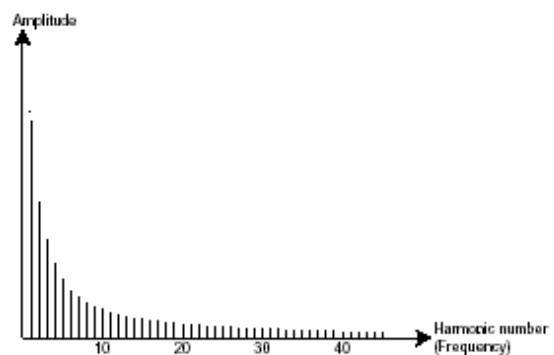
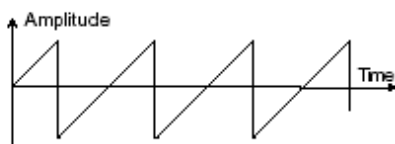
The oscillator settings on the ARP2600 V

The main oscillator settings are:

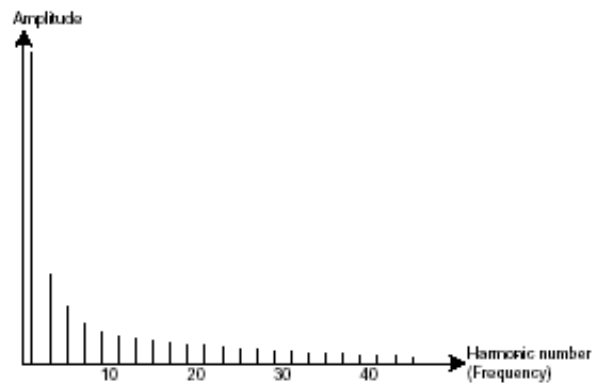
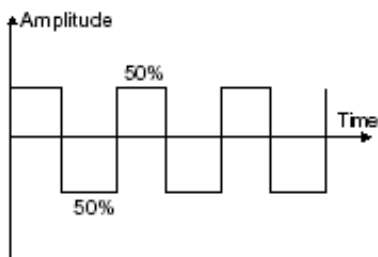
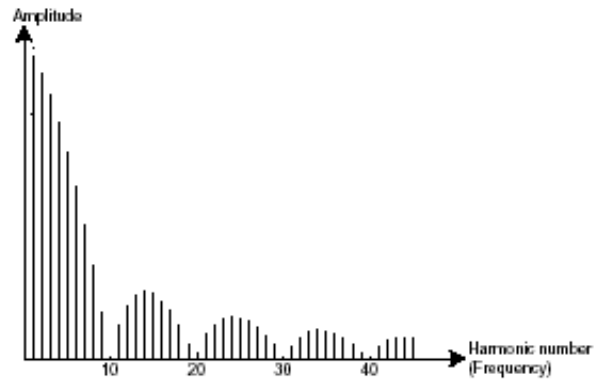
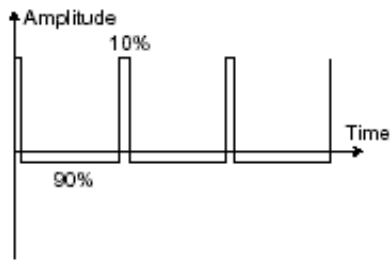
- **The pitch** is determined by the oscillation frequency. You can set the frequency of the oscillator with 2 controllers: first, the “RANGE” selector which determines the fundamental frequency – it is expressed in feet- : Low, 32, 16, 8, 4, 2; the highest number (32) brings the deepest tone, inversely, 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:

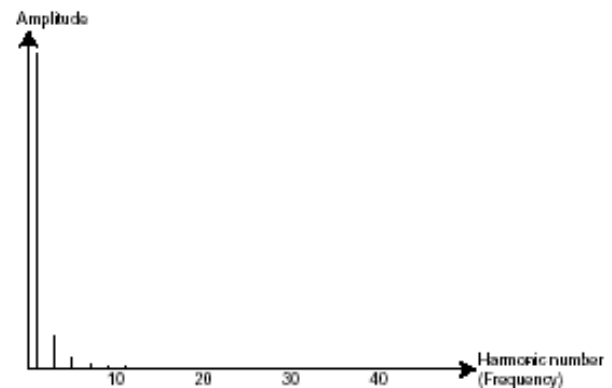
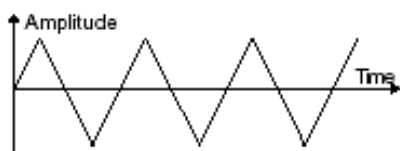
- The **saw tooth** presents the richest audio signal of the four waveforms (it contains all of the harmonics at decreasing volume levels in high frequencies). Its sound is ideal for brass sounds, percussive bass sounds or rich accompaniments.



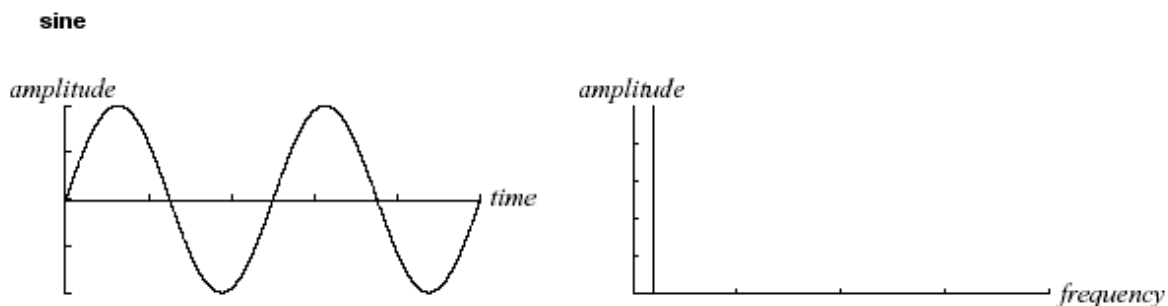
- The **square** possesses a more “hollow” sound than the saw tooth (it only contains odd harmonics) but none the less, its rich sound (notably in low frequencies) can be used for sub-bass sounds that will come out well in the mix (the square oscillator is often set an octave below that of the saw tooth),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

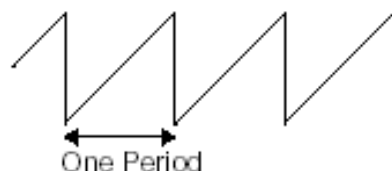


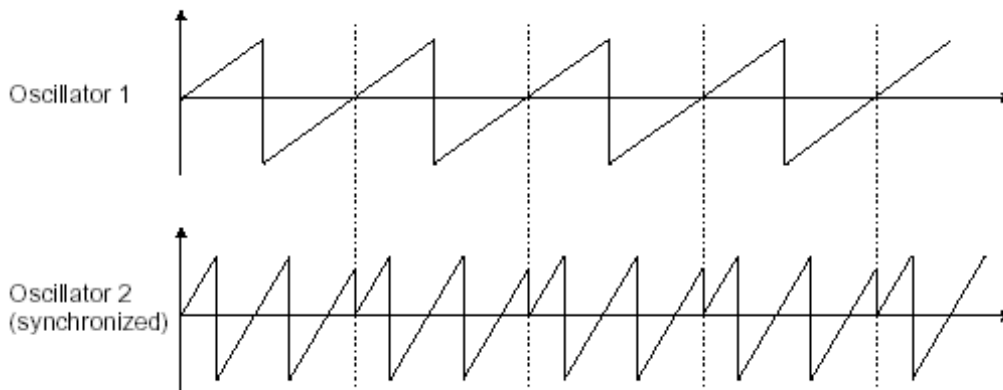
PWM (Pulse **W**idth **M**odulation) is a setting that allows you to modify the waveform cycle (or wave length). This can be done manually with the help of a knob “**PW**” or by modulation (with an envelope or LFO). This pulse width variation translates to a spectrum modification, resembling a waveform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



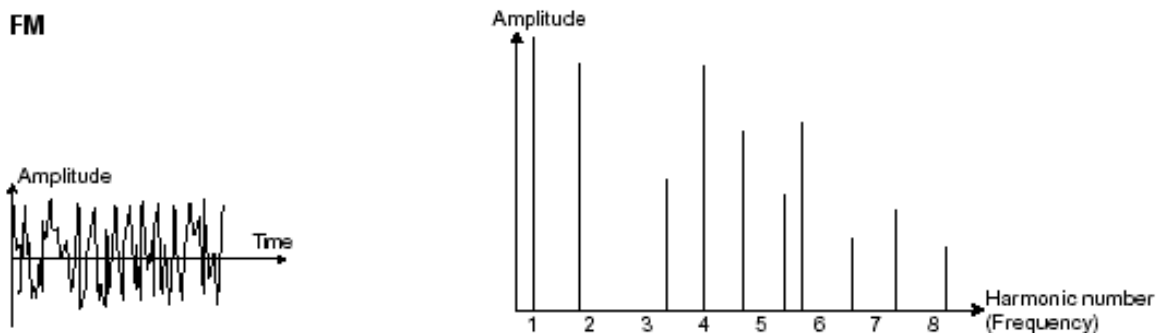


In the above image, oscillator2 is synchronized with the first and tuned to twice its frequency.

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.

FM



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

5.1.2 The filter or VCF

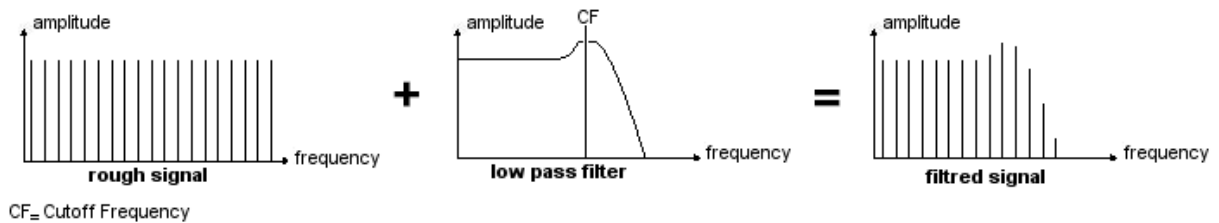
The audio signal generated by an oscillator (the waveform) is next generally directed to a filter module (**Voltage Controlled Filter**). It is this module that we use to control the sound by filtering (by subtraction, which explains the name given to this type of synthesis) the harmonics situated around 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

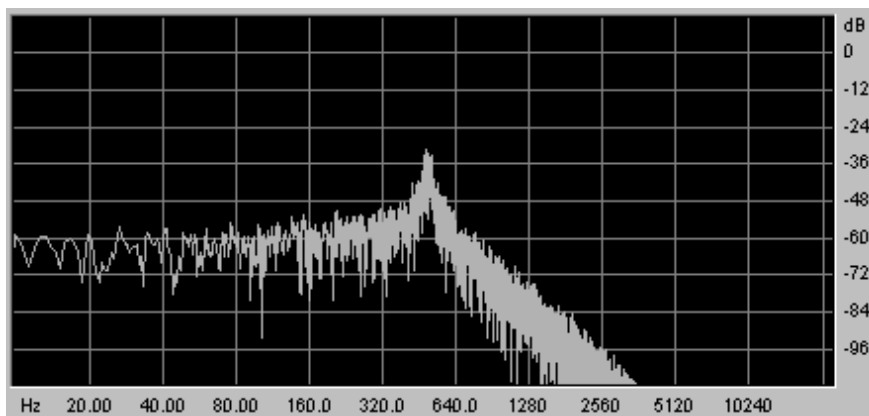
The ARP 2600 V offers two types of slope: 24 and 12 dB/Oct slopes.



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

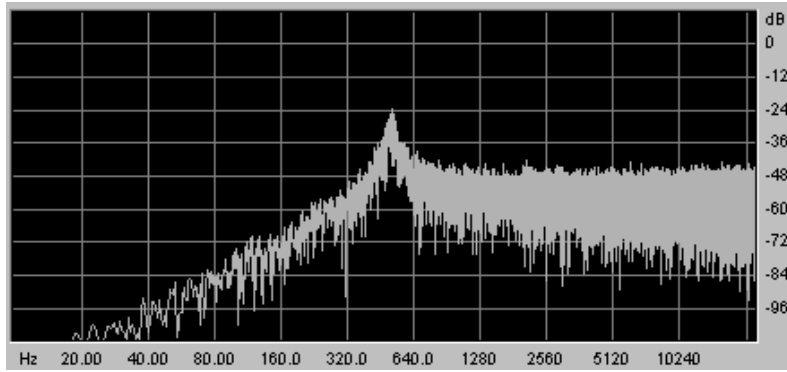
The low-pass filter (LPF) progressively removes high frequencies above the assigned frequency limit (the cut-off frequency) and allows the sound below the cut-off to pass through unchanged. Depending on the setting we will hear the sound becoming more or less “brilliant”, more or less “dampened”.

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.

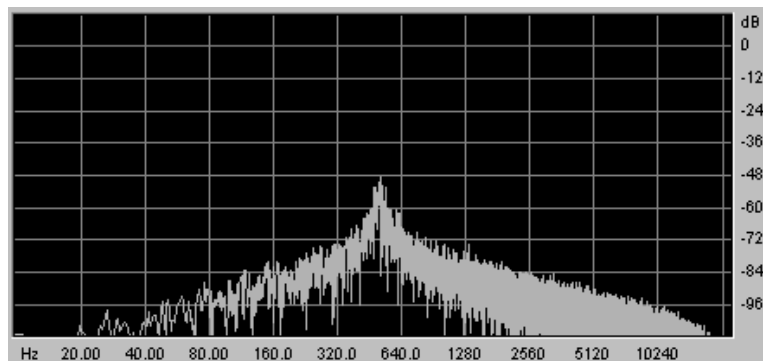


Spectrum of a noise signal proceed with a low-pass filter

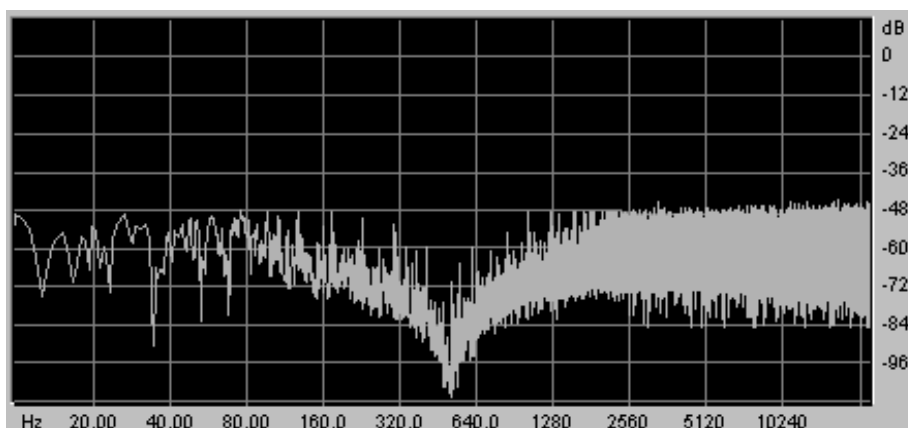
The high-pass (high-pass filter or HPF), as opposed to the low-pass, eliminates low frequencies and only allows high frequencies past. The sound will thus become “thinner”. It is very useful for removing redundant low frequencies.



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.

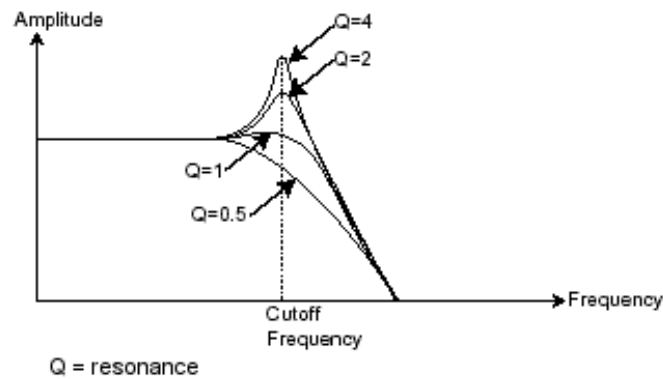
A second setting to compliment the cut-off frequency: the resonance.

(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.

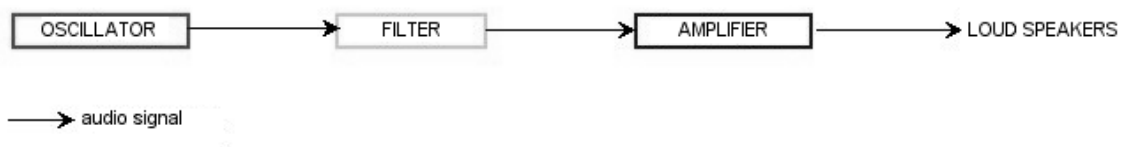
5.1.3 The amplifier or VCA

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:



5.2 Complementary modules

5.2.1 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 waveform) 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

5.2.2 The envelope generators

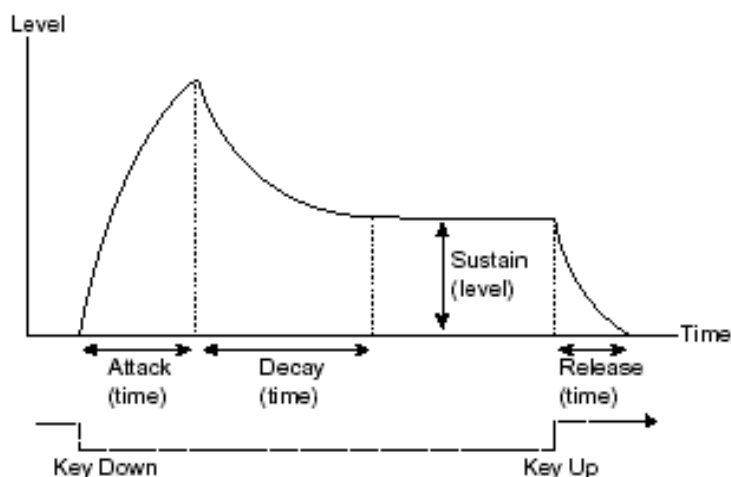
The envelope generator, connected to the amplifier, is used to “sculpt” the form of a sound during a cycle, which begins when we press a note on the keyboard, and ends when we release.



The envelope module

The most common envelope modules use 4 settings that 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.

5.2.3 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.

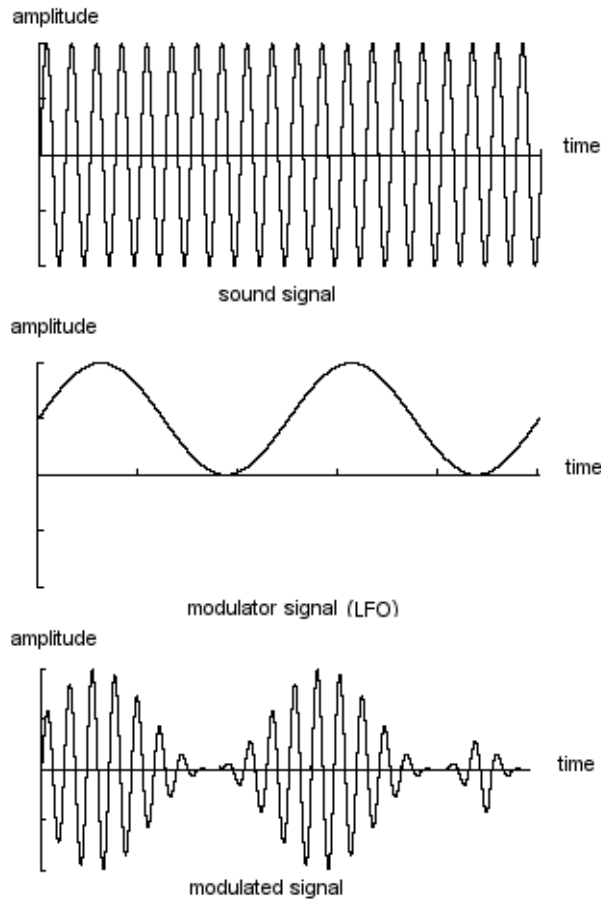
On the ARP2600 V, the LFO module is situated on the left of the sequencer .



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

5.2.4 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

5.2.5 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 saw-tooth waveform coming from a VCO to the sample and hold input.

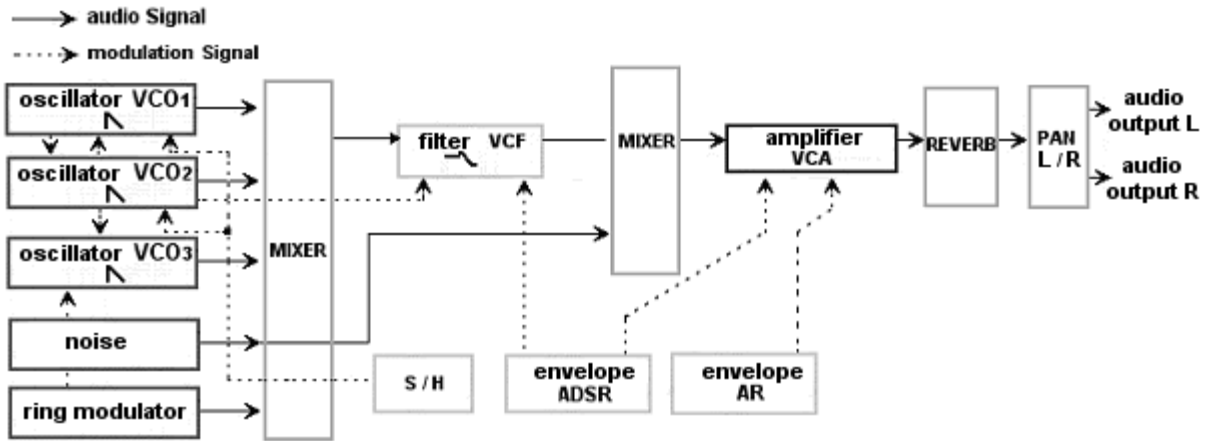
The famous sound of R2D2 in Star Wars was created with the sample and hold module on the ARP2600!



The sample and hold module of the ARP2600 V

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

- oscillators (VCO)
- 1 noise module
- 1 ring modulator
- 1 sample and hold
- 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 ADSR envelope
- 1 AR envelope



6 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

6.1 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 around 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).

6.2 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 ARP2600V2.

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 waveform 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 waveform 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 waveform 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

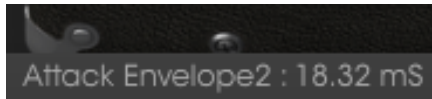
If it has a tendency to “disappear” too much (you don’t hear sound anymore), gently raise the cut-off frequency potentiometer.

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.

6.3 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

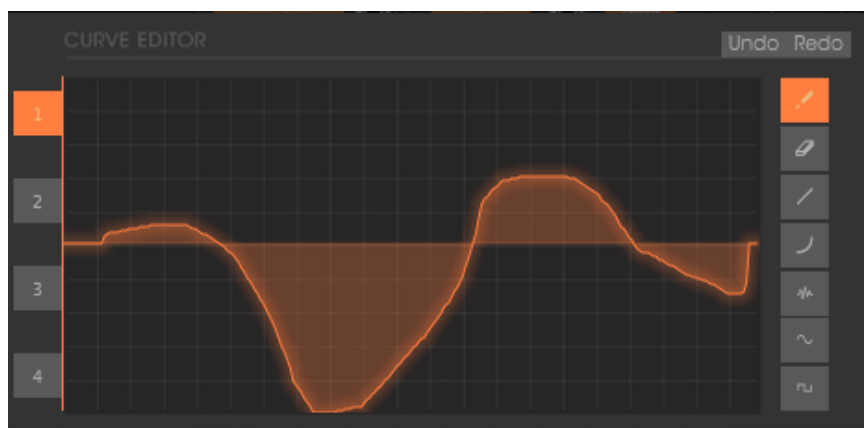
Raise the potentiometer above the modulation input jack. This is the modulation rate.



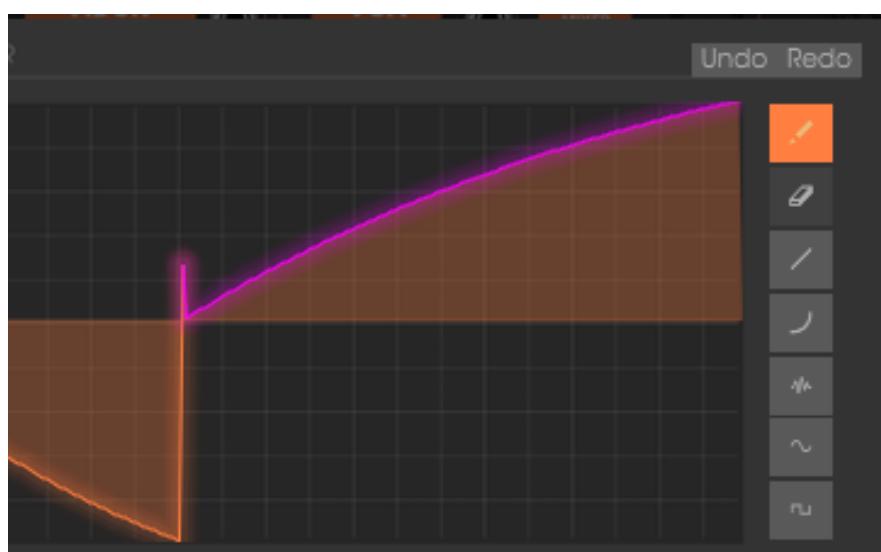
The modulation rate

Next click in the central part. A secondary window appears. It will allow you to graphically draw the waveform that will modulate the cut-off frequency.

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



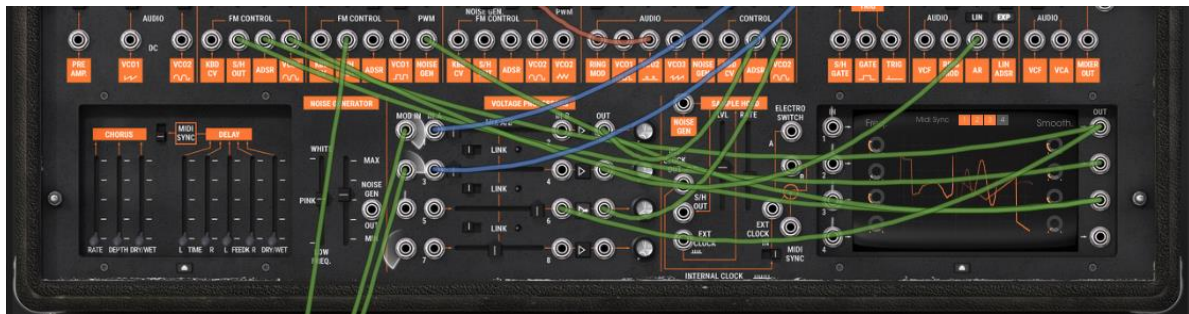
By default, the “pencil” tool is selected. Start by drawing a freehand curve. The form is not important. Once this is done, play a note to hear the result of this modulation. You can set the oscillation speed with the “Freq” potentiometer which is on the left of the tracking generator.



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.

6.4 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.

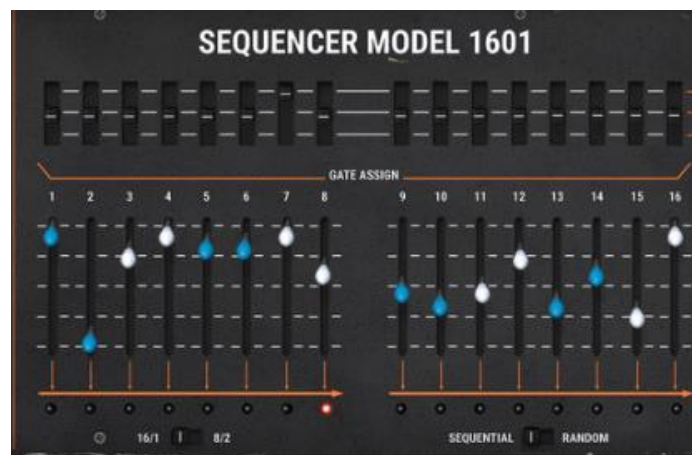
So that your sequence is tuned by following the semi-tones, connect the “Quantized output A” on the sequencer to the input oscillator1 key follow (“VCO1” > “CV KBD”).



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

6.5 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 waveform width for oscillator square waveform, 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 ARP2600V2. But don't hesitate in making your own attempts at programming; this is the best way to progress and find certain originality.

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