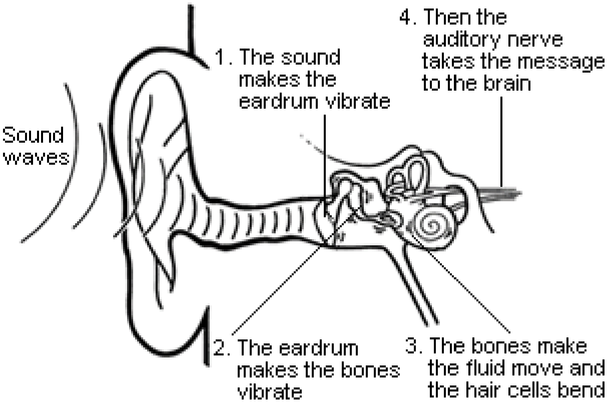
Instrument emulation, from Circuitry to Binary By Craig Logue

"Synthesizer - (music) an electronic instrument (usually played with a keyboard) that generates and modifies sounds electronically and can imitate a variety of other musical instruments [syn: synthesizer]"

(The Definition of the word Synthesizer from Dictionary.com)

Introduction to the fundamentals of sound.

Before discussing Synthesisers, it is necessary to discuss the fundamentals of Sound itself, in order to best understand how the idea of electronic audio synthesis works. Sound is a fluctuation in atmospheric pressure in our everyday environment that is sent into the air from a source and then reaches our ears. As the sound arrives at the ear, it is converted into an electrical signal that is sent to our brain giving us what we perceive as "sound". The ear is a hugely powerful tool and uses its complex internal biology to act as a decoder for these naturally occurring acoustic sound waves.



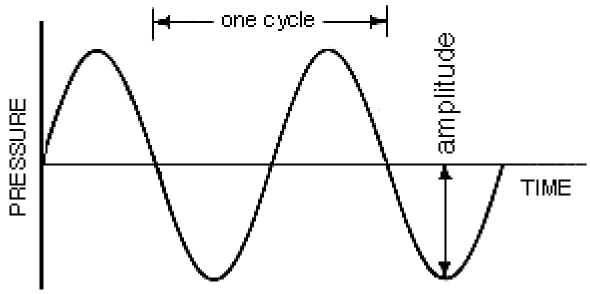
The inner mechanisms of the Ear (image taken from http://www.mydr.com.au)

David Miles Huber and Robert E Runstein, provide us with an analogy that accurately explains the movement of these acoustic waves in our environment and how they are perceived by our ears.

> *"An analogy of how sound waves travel in air can be demonstrated by bursting a balloon in a silent room. Before we stick it with a pin, the molecular motion of the room's atmosphere is at normal resting pressure. The pressure inside the Balloon is much higher, though, and the molecules are compressed much more Tightly together.....like people packed in a crowded subway car. When the balloon Is popped, the tightly compressed molecules under high pressure begin to exert an outward force against their neighbours in an effort to move towards areas of*

low pressure".(1)
"The human ear is a sensitive transducer that responds to these pressure variations by way of a series of related processes that occur within the auditory organs".(2)

With the introduction of electricity and the ability to analyse these acoustic waves in great detail, early scientists and electrical engineers began to discuss the idea that these waveforms could be generated artificially using voltage controlled oscillators (VCO). Using an oscilloscope, early audio scientists could visualize how the acoustic waves would travel through the air before they were received at the ear. An oscilloscope is an instrument that maps out variations in electrical pressure and its amplitude, against time, ultimately displaying a visual representation of the acoustic sound wave.



An example of how a sound wave is represented in the visual domain

(Image taken from http://www.sfu.ca)

Drawing on a wealth of knowledge of acoustic sound, scientists began to experiment by combining various sound modules and adding and subtracting these signals to generate unique new sounds, giving birth to a synthesiser industry that would have as much impact on the music industry as other hugely successful electrical inventions such as the electric guitar and amplified sound.

The historical development of hardware based Synthesisers

The idea of generating and controlling sound signals of varying frequencies to create music is an age old method. As far back as 3rd century BC, Long before the invention of electricity, pioneers were developing primitive methods of synthesis using only apparatus that was available to them at that time. The "Hydraulos", literally translated as "Water oboe", was a device that used water pressure and a series of valves and sliders to generate tones and is widely considered to be a distant relative of the more modern Pipe organ.

"The Hydraulos" (Image taken from http://new.taringa.net)

With the dawn of electricity the lid was lifted on the ability of early inventors to generate tones with circuitry and loudspeakers, and they soon began to find ways of combining tones of different frequencies to create interesting new sounds as well as emulating the sounds created by more traditional acoustic instruments. Inventors such as Hugh LeCaine and Robert Moog set upon a path that would help model the way we emulate sound today and would lead to early synthesisers "The Mini moog" and the "P3 Polyphonic Synthesiser", showing the potential of such technology within the musical world.

Both Instruments contained similar characteristics that would become common place in synthesisers that followed. Firstly, a number of Oscillators were used to generate tones of differing frequency and type. These Oscillators would feed directly into a multi channel mixer, allowing the user to affect the audible volume of each tone. The user then had the ability to send the tone through various filters before it reached the amplifier and loud speaker. Most importantly both instruments were controlled using a keyboard, giving the user more control over the sounds being generated.

The Mini moog (Image taken from http://onceuponawin.wordpress.com)

The P3 Polyphonic Synthesiser (Image taken from http://www.hughlecaine.com)

Inventors the world over began to develop their own Synthesisers based on the principles of early inventions such as the Mini Moog and the P3 Polyphonic Synthesizer and would design combinations of Oscillators and modules that would fill most studios, however the Mini Moog proved to be equally as versatile using large scale switches without the need for large amounts of cable.

As inventors battled with the increasing restrictions placed upon them by early electronic circuitry, a revolution was taking place in Silicon Valley that would have a huge effect on the synthesizer industry. The invention of Micro Chips and RAM memory cells would allow synthesiser manufacturers to achieve sounds that were unprecedented in comparison to earlier capabilities, as well as giving them the ability to save their sounds to memory cells that could be recalled at any time. A fore runner during this period of the synthesizers' development was the Prophet 5, which incorporated the fundamentals of earlier electronic synthesisers with the new capabilities of the micro chip.

The Prophet 5 (Image taken from http://www.vintagesynth.com)

The prophet 5 was a huge leap forward for performers and recording artists alike with its five voice polyphony (two oscillators per voice) and a white noise generator. This gave musicians the ability to combine a huge number of tones that, when processed using the analogue filters, produced a whole new world of sounds. There were three consecutive versions of the Profit 5, namely the Rev 1, Rev 2 and Rev 3. The Profit 5 was immortalised during the 1980's in the music of bands such as Kraftwerk, Depeche Mode and the artist Gary Newman and although its warm analogue tones were a huge hit, they could be unreliable and difficult to tune. The latest, Rev 3 version, would attempt to fix some of the bugs of its predecessors and included the ability to modify it to take a MIDI retrofit. MIDI (Musical Instrument Digital Interface) was a concept that connected instruments via a MIDI interface and allowed them to communicate and synchronise with each other. Although the Rev 3 version of the Prophet 5 allowed DIY electricians to install a MIDI interface, the process was difficult and often lead to damaging the instrument.

MIDI was a protocol that had exploded into the industry in 1982, and when combined with developments in computer technology, musicians were given unlimited scope to combine the capabilities of more than one instrument; unlocking the restrictions placed upon them by the number of voices and variations of sound they were allowed. A new wave of Super Star Keyboard players, such as Rick Wakeman and Keith Emerson, would have racks and racks of different instruments on stage all connected via MIDI. The development of the MIDI protocol and the ever increasing processing power of computers and their operating systems would soon embed itself in the synthesizer industry and heralded a completely new approach to composition and sound generation.

Early Synthesisers were developed that were controlled internally by digital messages, Telling the internal microprocessors what key was being hit and which knobs were being turned, however each manufacturer had a different method of encoding the messages. In a move that was groundbreaking at the time, and showed mass cooperation between competing manufacturers, MIDI (Musical Instruments Digital Interface) was developed as a form of communication between synthesisers and sequencers that were built by different manufacturers. It soon evolved to transmit key velocity information, pitch bend and modulation wheel movement as well as employing a strict MIDI clock that, when connected to another instrument such as a drum machine, would synchronise both instruments. By creating a chain between each instruments MIDI output to the MIDI input of the next device in the chain, the user could create complex combinations of instruments that could be easily controlled in a live environment. MIDI had a massive impact upon the music industry and is accurately summarised by Jon Chappell in his book “Digital Home Recording”.

> “Debuting in 1983, MIDI kicked off an unprecedented period of growth in the electronic music industry. Other factors, such as the declining cost of microprocessors, played key roles too, but MIDI was the glue that held it all together”.(3)

As MIDI evolved, like any language of digital communication, it was tweaked and developed until eventually MIDI 2 or General MIDI 2 (GM2) was introduced in 1999. Other methods of communication were invented such as Open Sounds Control (OSC) but generally fell to the wayside as General MIDI dominated the market.

The technical aspects of Synthesis

Initially, early synthesizers were huge instruments that reflected the primitive technologies of the day, however, as electronic hardware developed, so did synthesizers. As Circuits became smaller and microchips were introduced, synthesizer manufacturers were able to make their instruments a lot more compact and portable without compromising on their ability to create unique sounds. The introduction of Memory modules contained within the circuitry of the synthesizer also made it easier to sample audio that could be played back at will. These developments proved popular with musicians as the synthesizer became a new and interesting addition to many bands Such as the Beatles and Pink Floyd. Developers began to invent new methods of synthesising sound that would utilise the ever increasing versatility of new Synthesizers.

The various methods and technologies involved in synthesising sound include subtractive or additive synthesis, wavetable synthesis and sample based synthesis among others. The basic definition of Synthesis is to combine two or more things together to form something new, and when we apply this to the world of sound, the creative use of these combinations produced new and exciting results. Earlier synthesisers were restricted by the ability to only generate simple tones using primitive oscillators and were restricted to ADSR (Attack Decay Sustain Release) controls to manipulate those tones, but as systems became more complex manufacturers could come up with new methods to recreate instruments.

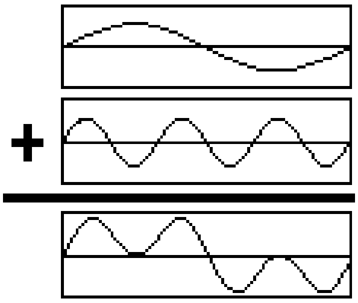
Subtractive Synthesis

Subtractive synthesis is the method of removing harmonics from the signal being generated to produce new sounds or to replicate the timbre of an acoustic instrument. If the user was to generate a Saw tooth signal and then remove the lower end of the sound using a Low Pass Filter, with some fine adjustments it could sound like a bowed violin. This method normally employs a number of oscillators that create a repeating wave of sound, Filters that do the harmonic subtraction, a voltage controlled amplifier (VCA) that amplifies the sound to an audible level and Envelope Generators such as an ADSR (Attack Decay Sustain Release) that allow the user to shape the notes volume and harmonic content over time. Subtractive synthesis is generally associated with earlier electronic synthesisers such as the mini moog and the ARP 2600 which actively competed against each other during the 1970's and early 1980's.

The ARP 2600 (Image Taken from http://www.vintagesynth.com)

Additive synthesis

It is known that an instruments timbre is made up of multiple harmonics and the process of additive synthesis allows the user to control the frequency and amplitude of each of these harmonics. The timbre of an acoustic instrument such as a violin or a cello shows that their harmonics change over time and the ability to control the frequency and amplitude of each of the synthesizers' harmonics would result in more accurate recreations of those instruments. This method of synthesis added more dimension to the sounds being created. Rather than rigid tones with filters applied to them, sounds created using additive synthesis had more movement and offered more potential to the user. Additive synthesis creates complex waveforms by mixing multiple simple waveforms.



Additive Synthesis (Image taken from http://www.sonicspot.com)

An example of an early additive synthesiser that was used successfully in the music industry was the RMI Harmonic synthesiser that was made popular by Jean Michelle Jarre. Released by Rocky Mountain Instruments (RMI) in 1974, The Harmonic Synthesiser was the one and only synthesiser released by the company and incorporated the fundamentals of additive synthesis, giving it the ability to generate complex sounds.

The RMI harmonic Synthesiser (image taken from http://www.flickr.com)

Wavetable Synthesis.

Wave table synthesis was originally developed by Wolfgang Palm in the Late 1970's and has since been adopted as the main method of synthesis by many International synthesizer manufacturers. Wavetable Synthesis is the process of generating sounds using digital signals by storing small samples of musical instruments which can then be combined, altered and enhanced to completely remodel the signal being generated. A single period of a periodic waveform is stored to a circular buffer where it continuously loops, allowing the user to control the speed at which the loop is played back and creating new output waveforms. The process of Wavetable synthesis is well summed up by Ben Milstead in his book "Home Recording Power".

> *"Instead of synthetically manufactured oscillators creating source waveforms,*
> *Wavetable makes it possible to digitize a waveform from the natural world*
> *Directly into memory or a storage device. This "sampled" waveform can then*
> *Be modulated and amplified in a regular synthesiser-like fashion". (4)*

Sample Based Synthesis

As computer technology further developed and storage space and processing power became more affordable, Synthesizer manufacturers began to employ sample based synthesis methods to more accurately recreate instruments and to allow musicians to develop their own sounds. Whereas wavetable synthesis was restricted to sampling a single period of a periodic waveform, new technological advances allowed users to sample and store much longer samples. The instrument being recreated could either be sampled at a single note and then allow the synthesiser to modulate the sample for all other notes on the keyboard, or every note could be sampled creating a more accurate remodelling of the instrument. Although previous synthesis methods had produced credible recreations of various instruments, the age old argument of man over machine once again proved that there are certain nuances to an acoustic instrument being played that machinery simply cannot capture. The sample based synthesis method provides a more accurate progression from lower frequencies to higher frequencies. The lower notes are not overly dull whilst the higher notes are not overly bright, allowing the user to create a far more accurate recreation of an instrument with a more Natural feel when it is played.

A classic example of an early Sample based synthesiser was the Fairlight CMI (Computer Music Instrument). Designed in 1979 by Peter Vogel and Kim Ryrie, founders of Fairlight, it displayed the possibilities of sample based synthesis, however, it was highly restricted by the processing power and technologies of the time. Widely respected by enthusiasts as one of the earliest sample based synthesisers, it would quickly be overshadowed by newer inventions, but would always remain an icon in the music industry.

The Fairlight CMI (Image taken from http://120years.net)

An in-depth analysis of Two Pioneering Hardware

Synthesisers the Minimoog and the ARP2600

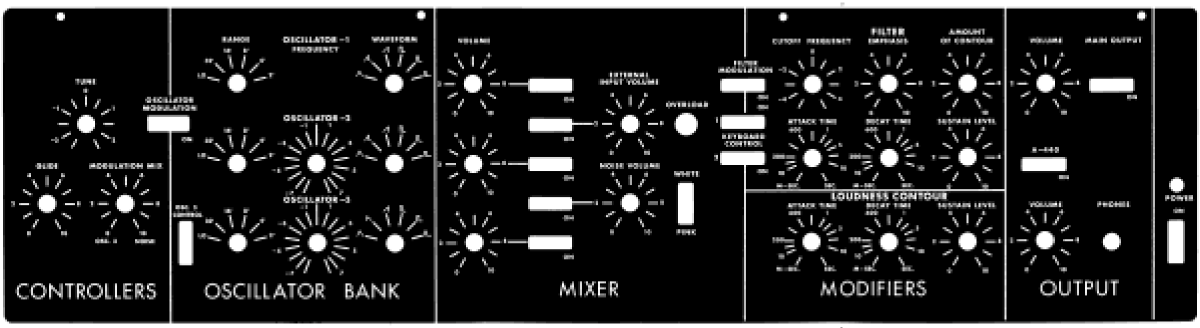
Developing further upon previous research into the historical development of electronic synthesisers, The following sections will take a detailed look at the technologies and capabilities of two specific synthesizers that are widely considered to be the first commercially successful instruments available to the general public and were an early example of the potential of electronic synthesizers in both composition and live performance.

The Mini Moog

In 1954 a young engineering graduate from New York was working from home with his father building and selling Theremins, but in the years that followed he would have a ground breaking impact on the world of electronic synthesis. Robert Moog began to build his own Modular Synthesizers that could be connected and controlled by a monophonic keyboard. Moog unveiled his early inventions at the 1964 Audio Engineering Society Convention and it immediately sent ripples throughout the industry. However Moog's modular synthesisers would be truly immortalised in the seminal album "switched on Bach" by Wendy Carlos, where listeners got to hear the true potential of such an instrument.

The mini moog was an amalgamation of years of design tweaks and developments in technology and epitomised the fundamentals of electronic Synthesis. Moog managed to squeeze the capabilities of his earlier modular synthesizers, into a compact and portable casing, making it the first synthesizer that musicians could take on tour without too much effort.

The mini moog was made up of a number of components that were grouped on the front panel of the synthesizer. Each group would offer the user different methods of affecting the tones that were being generated.



The front panel of the mini moog (image taken from http://www.arpeggi8.com)

Controllers

The controllers located to the left of the mini moog's front panel are generally used in the tuning and set up of sounds and gives the user control over how the signal should act when a note is played. The three controllers are Tune, Glide and Modulation mix and each provides a different effect on the sound.

- Tune – The tune controller allows musicians to "tune in" to the band they are playing with, or, to tune to the A440 factory reference tone.

- Glide – the glide control allows users to specify the amount of portamento or "glide" between notes as a sequence is played.

- Modulation Mix – In addition to functioning as an audio signal, Oscillator 3 could be used as a control signal. As a control signal it could be mixed with the output of the Noise generator. The user could then introduce the modulation mix using the modulation wheel and would allow them to create deeper sound textures.

Other controllers were available to the user that could be implemented live to create exciting new sounds that had never been heard before. Obviously, the mini moog's Keyboard was its self a controller as it could be used to control the filters, but there were also three controllers to the left of the keyboard that could be used effectively in live performance. These controllers were:-

- Pitch Bend – The pitch bend allowed the user to alter the frequency of the tone that was being generated by the oscillators by rocking the wheel backwards and forwards as required and was applied in real time.

- Modulation wheel – As previously mentioned the modulation wheel was used to introduce the setting created the modulation mix process to control the output of the noise generator with oscillator 3.

Ben Milstead summarises Controllers in his book "Home recording Power".

> *"A controller is any component, analogue or digital, that is applied to one or*
> *More of the primary sound generating elements: source, modulator, or*
> *Amplifier. Controllers give you great flexibility by extending a keyboards*
> *Dynamic and sound shaping potential. In addition, a controller can also*
> *Be the electronic keyboard or other input device. It "controls" the source*
> *Waveforms frequency". (5)*

Oscillator Bank

The oscillator bank consists of three separate and independently operated oscillators that are capable of generating six different waveforms, triangular, saw tooth triangular, saw tooth, square, wide rectangular and narrow rectangular. The Oscillator bank also has a number of dials and switches that can affect the waveform being generated. Each oscillator has individual controls that can alter the following:-

- Range – The range switch allows the user to set the pitch of the tone being generated by the oscillator. By hitting a note on the keyboard and cycling the switch from left to right, the frequency of the tone would step up and down in octaves, except the switches lowest setting that create a sub sonic click that could be used to create rhythms.

- Waveform – The waveform switch allowed the user to select between the six types of waveform that the Oscillator was capable of producing.
- Volume – The volume control affects the amount of the oscillator signal that is being fed to the mixer. Each oscillator signal can be turned on or off independently using the mixer switches.

- Oscillator 3 control – The oscillator 3 control switch is used to disconnect oscillator 3 from the control of the keyboard and give Oscillator 3's frequency control a much wider range.

- Oscillator Modulation – The oscillator modulation switch was a way of stopping the oscillators from generating waveforms and could be used as a way of resetting sounds.

The Mixer

The mixer was the most important part of the mini moog as it was the output signal of this component that was sent to the filters. There were a number of controllers and switches that would allow the user to create an overall mix of the oscillator signals being generated.

- Volume – The volume dial allowed the user the control the level of the signal being generated by the oscillator. Each oscillators dial had a corresponding on/off switch that could be used to introduce and remove each of the oscillators from the mix individually.

- External Input Volume – This dial controlled the level of input from any external instruments, such as a microphone or guitar that might be connected to the mini moog via the input on the rear panel of the instrument. This dial has a corresponding on/off switch that can add or remove the external audio signal from the mix.

- Noise volume control – This dial controlled the level of noise being generated and had a corresponding switch to select the type of noise to be generated (White or Pink). It also had a corresponding on/off switch that allowed it to be added or removed from the overall mix.

Modifiers

The modifier group contained two subgroups that had controls for the two sound modifiers, the filter and the Loudness contour. These sound modifiers affect the overtone content of the sound that is being channelled from the mixer. The user had the ability to control separate elements of each sound modifier.

Filter

- Cut off Frequency – This dial controlled the filters cut-off frequency allowing users to sweep from dull muted sounds through to high shrill sounds.

- Amount of Contour - This dial allowed the user to control the amount of contour that was applied to the filters cut off frequency.

- Attack Time – This dial controlled the attack time of the note being played. By hitting a note and observing the difference as the attack time is increased, it was possible to hear the effect the attack time was having on the sound being generated. The movement from a dull sound to a bright sound was rapid when the attack time was low and then slowed down as the attack time was increased.

- Decay Time – This controller basically did the reverse of the attack time control. By increasing Decay time the user affected the drop off of the waveform being generated. The movement from a bright tone to a dull tone would gradually slow down as the Decay time was increased.

- Sustain level – This dial allowed the user to control the frequency at which the contour levels off as the tone is generated. As the user hits a note they could affect how long the tone remains bright. Setting the dial to 0 would completely cut the sound and setting the dial to full meant the note would continue to play at the frequency of its initial peak. Setting this dial somewhere in the middle would allow the user to control the length of the notes as they were played.

- Emphasis – The emphasis control dial added a sharp resonance in the response of the filter at the cut off frequency giving users scope to create interesting new sounds.

Loudness Contour

This Modifier was an overall controller for elements of the signal being sent from the mixer and being processed by the filter before it reaches the amplifier. There are three separate controllers contained within the Loudness Contour group:-

- Attack Time – similar to the process employed by the filters attack time, the attack time the initial rise in volume of the waveform to a peak.

- Decay Time – Again similar to the effect produced by the filters Decay time controller, this dial controls the drop in volume from the initial peak to the sustain level.

- Sustain Level - This controller allowed the user to determine the volume level at which the contour levels off after the attack and decay.

The Output

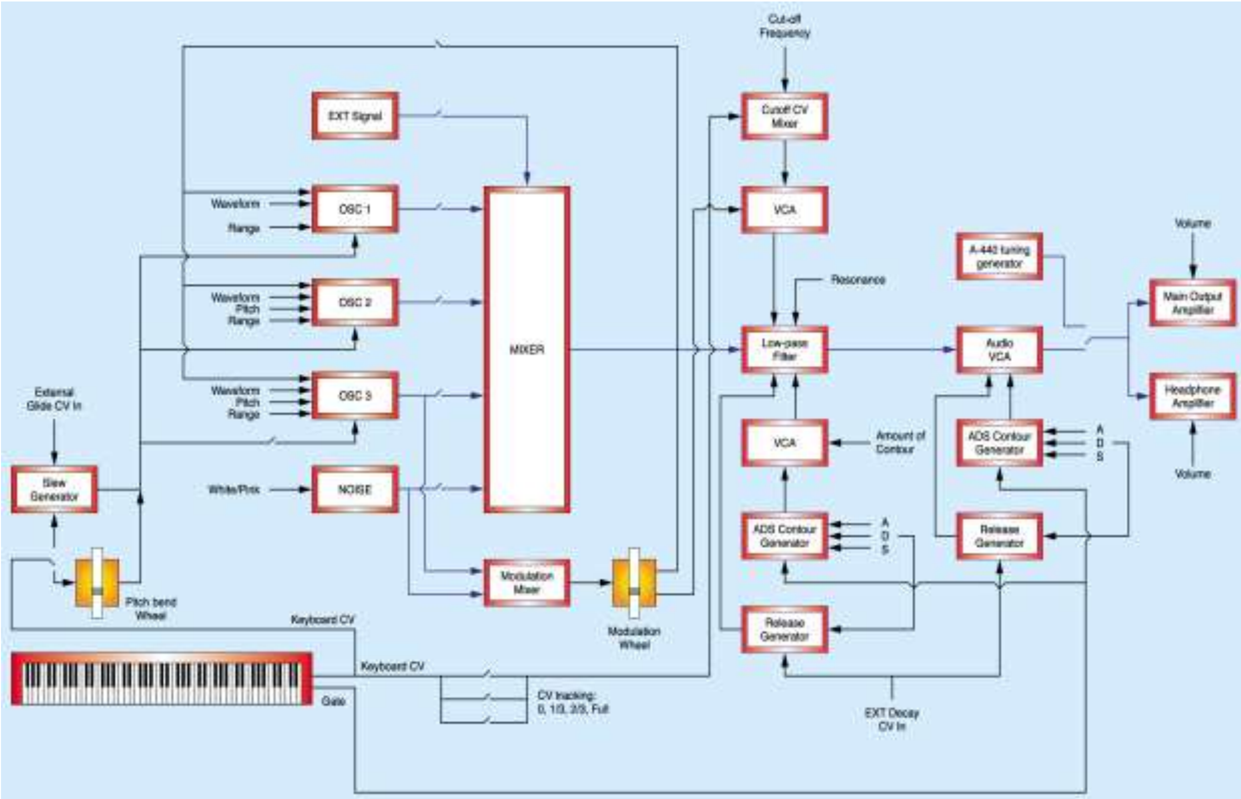
Once the Oscillators had generated their signals and a mix had been achieved and effected using the filter and Loudness contour modifiers, the resultant signal was sent to the mini moog's amplifier and output. This section of the front panel is mainly used to power the instrument and connect it to a loud speaker. There are a number of components to be noted in this group:-

- Volume – There were two volume controllers on this section of the front panel, one for the main output and another for the headphone volume. This dial determined the level of the volume being sent to a loudspeaker as well as headphones. The volume controller had a corresponding on/off that would affect the overall signal fed to the amplifier.

- A-440 – This switch would generate a reference tone that would allow users to tune the oscillators and filters manually. This was a laborious process that involved a certain level of electronic expertise.

Block Diagram of a Mini Moog's circuitry

(Image taken from http://www.soundonsound.com)

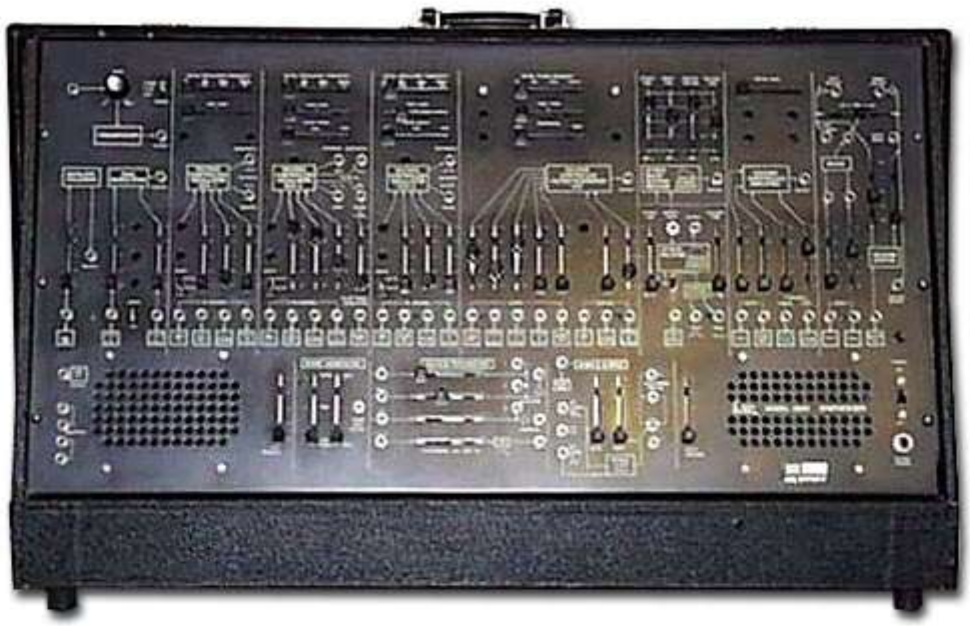


The design ingenuity of the mini moog was way ahead of its time and the instrument would become a proud ancestor to some groundbreaking instruments that were to come in future years that would still carry the fundamental processes employed in Robert Moog's early creations.

ARP 2600

During the Mid to late seventies, as the mini moog was making huge in roads into the music industry, another ground breaking synthesiser had been developed by ARP instruments and was ready to be released to an eager public. Named after its designer, Alan R Pearlman, the ARP2600 stood in direct competition to the already popular mini moog. Pearlman, a 57 year old engineer, was more familiar with classical music than rock or pop, and although his synthesisers were taken to heart by these industries, he himself did not understand them and felt they were pre occupied with drugs and groupies and were made up of mainly inarticulate musicians. His engineering background had seen him design amplifiers for the Apollo and Gemini space programs and despite being far removed from the rock and pop music industries his instruments would eventually dominate more than 40% of the synthesiser market, superseding the share held by the mini moog.

Both instruments were monophonic and used Voltage Controlled Oscillators to generate sound, however the ARP 2600 was considered to be more versatile than the mini moog when it came to creating complex sounds. Both instruments were based on similar design ideas in that they both incorporated three oscillators that could be manipulated independently and a voltage controlled filter, however the ARP 2600 would introduce unique features that would stand it ahead of its competitors.



The ARP 2600 front panel (Image taken from the http://www.vintagesynth.com)

In today's market, the original ARP 2600 synthesiser is extremely rare and early versions regularly exchange hands for large sums of money. This image is a rare glimpse at the early ARP2600, and although crudely taken, gives us a snapshot of the capabilities of the instrument. The ARP 2600 was similar to the moog in that they were both hardwired with fundamental devices needed to successfully synthesise sound, however, the ARP 2600 was "semi modular" as it also allowed the user to reprogram the electronic circuitry using patch cables. This capability would make it popular with artists and it was widely embraced by the music industry at large.

The hard wired modules contained within the ARP 2600 are displayed in the following table that was taken from the original ARP 2600 owner's manual written by ARP Instruments inc. in 1977.

SOURCES	MODIFIERS	CONTROLLERS
VCO 1	VCF	ADSR
VCO 2	VCA	AR
VCO 3	RING MODULATOR	KEYBOARD
NOISE GENERATOR	ELECTRONIC SWITCH	ENVELOPE FOLLOWER
	REVERB	SAMPLE AND HOLD
	INVERTERS	VCO's IN LF RANGE
	LAG PROCESSOR	INTERNAL CLOCK
	MICROPHONE PREAMP	

<u>Sources</u>

The "Sources" are the circuits that actually generate the waveforms that are combined to create the overall sound. This is the area of the synthesiser that would generate the initial tones that could then be routed through other pre wired circuitry that was then used to affect the sound in many different ways.

<u>VCO 1</u>

The first of the Voltage Controlled Oscillators (VCO), this module was only capable of generating a Saw tooth wave and a square wave. This allowed the designers the freedom to hard wire the keyboard control voltage, sample and hold, ADSR (Attack Decay Sustain Release) controllers and the sine wave generated by VCO 2, to the frequency control of VCO 1. This would show how versatile the ARP 2600 was and show off its superior "Semi Modular" design that when used intelligently could create sounds that were out of this world.

VCO 2

The second of the Voltage Controlled Oscillators was capable of generating saw tooth, pulse, triangle and sine waveforms. The oscillator would generate all tones simultaneously and the user would simply connect a patch cable to the output of the wave form they wanted to hear. This could then be patched through to the filter and manipulated using its features before being sent to the voltage controlled amplifier and main audio output.

VCO 3

The third of the Voltage controlled Oscillators, this module was similar VCO 1, however, a pulse wave generator replaced the square wave generator; and the symmetry generator slide was replaced by a slide that allowed the user to manually control the variable pulse width. By using this oscillator to generate saw tooth waves and combining them with the wave forms being generated by the other two Oscillators, the user could create huge complex sounds that would be easily recognised in recordings from artists such as Jeane Michel Jarre on his 1978 album “Equinoxe”, as tones created using the ARP 2600.

Noise Generator

The Noise generator provided the user with the ability to apply a noise signal to the overall tone being generated by the Voltage controlled oscillators. The noise generator could create a signal of +20 volt peak to peak with around 10 octaves of bandwidth. Using the first of the two sliders that controlled the Noise generator, the user could select the type of noise being generated. The slider could be moved from “white” noise, to “pink” or to a position that created a low frequency noise. The second of the two sliders controlled the amplitude of the noise signal being generated and when used creatively, could add noise that would sit subtly in the overall mix of the tone being generated.

Modifiers

The “Modifiers” group of the ARP 2600 contained all of the circuitry that would directly manipulate the sound being generated by the “sources”. Using various modulators the user could affect the sound with filters and effects in a creative manner, producing results that would stand it ahead of its main competitors in a fast developing industry.

VCF

The Voltage controlled filter (VCF) was where the ARP 2600’s versatility began to shine. Providing the user with a 24db/octave Low pass filter with a variable cut-off frequency and resonance, where the cut-off frequency could be moved between 10 Hz to 10 kHz adding interesting effects to the overall sound being generated by the "sources".

VCA

The Voltage Controlled Amplifier (VCA) controlled the amplitude of the signal passing through it, in this case the signals being generated by the "sources" and how they are affected by the VCF. This is the point in the circuit where the signal is sent to the speakers or to the headphone output.

Ring Modulator

The ring modulator is a device that takes an input from two separate sources and multiplies them to create complex and interesting waveforms. The ring modulator was commonly used to create percussive sounds such as a bell or a gong, and would add a whole new dimension to the sound being generated.

Reverb

A sign of the ARP's advancing technology; the reverb unit was pre wired to the output of the mixer. As the signal entered the reverb generator it was sent through a reverb spring driver and output to either the left or right reverb sliders, which independently applied the reverb effect to its corresponding output channel.

Microphone Pre Amp

The microphone pre amplifier was used as an input for external signals such as a microphone or any other device that could be connected via a ¼ inch jack input. There were three input gain ranges available to the user, 20db, 40db and 60db and a rotating controller that determined the input attenuation of each range.

Controllers

The "controller" group of the ARP 2600 was the area of circuitry that allowed the user to control the individual parameters of a variety of envelopes that could be applied to the signal being generated by the "sources". The "controllers" would add a new dimension of control for the user in their quest to find that new unique sound and can be heard used to their full potential in the music of artists like David Bowie and Depeche mode who helped to introduce the ARP2600 to the record buying public.

ADSR

The Attack Decay Sustain Release (ADSR) controller would allow the user to control the amount of each of its parameters was applied to the sound being generated. The main operation of the ADSR controller was the same as the AR, however, it also introduced control parameters for Decay and Sustain. These parameters also generate control voltages with variable rise and fall that can be used creatively to add further dimension to the sound being generated.

AR

The Attack Release (AR) was an envelope generator that would produce a control voltage with variable rise and fall times. By applying this control envelope to the amplitude of the signal being generated, the user could experiment with fading the tone in and out, altering the speed at which the rise and fall times occurred.

Envelope follower

The envelope follower was a circuit designed to measure the amplitude of the positive peaks of any wave form being generated. This measurement was then converted into a control voltage that could be used to control a variety of the "modifiers".

Sample and Hold

The sample and hold feature that was introduced on the ARP 2600 was another sign of its ability to create sounds that were unique to it. The sample and hold circuit receives a variable trigger pulse that causes it to sample and store the input voltage at that exact point. This sample is then held until another trigger pulse is received, where it then plays back. The amplitude and rate of the sample can be controlled using the two sliders available with the rate slider also controlling the internal clock speed.

Internal Clock

The internal clock, although crude in design, would introduce the idea of creating an internal time based signal with a low frequency oscillator that could be output to a separate device and used as a time reference. This would come in handy when users wanted to create rhythmic sounds.

The ARP 2600 showed a huge leap forward in the capabilities of synthesisers in the late 1970's, and although not as compact and easy to transport as the mini moog, the ARP 2600 was far superior in its ability to generate multiple waveforms and control and affect them in ways that would result in pleasing sounds to the listener. It could be said that the mini moog was designed more with "musicians" in mind, whereas the ARP 2600 was designed more with "scientists" in mind. The mini moog can be used to quickly generate sounds that could be incorporated into a live performance where as the ARP 2600 offered the user so much control, that it would often require more time to create equally pleasing sounds that could be incorporated into a live performance.

The historical development of Software based Synthesisers

The mini moog and the ARP 2600 were of an electronic age, which relied heavily on the physical generation of waveforms using oscillators and analogue circuitry; however, the dawn of the computer age would have a huge impact on the synthesizer industry and banish these instruments to the dusty store rooms of studios and educational establishments. Apart from the fact that it became common place for designers to incorporate micro chips into their instruments circuitry, Software programmers began to experiment with the idea of "virtual" instruments that used algorithms and computer programming to accurately recreate classic hardware synthesizers. As computer systems became more powerful and processing power more abundant, programmers were able to place bigger demands on systems leading to a range of "virtual" instruments such as the Rebirth program by Propeller heads, which accurately remodelled a Roland TR-303 groove box into the virtual domain. Ben Milstead says in his book Home recording power,

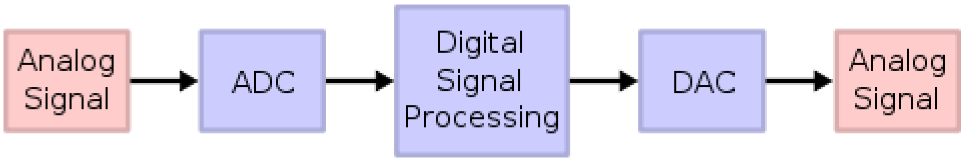
> *"One of the most important audio applications of 1997 was a program call Rebirth by Propeller head software. Through physical modelling, Rebirth was able to emulate and mimic the sounds and behaviours of a couple of the most sought after and expensive bass synths and drum machines for dance music, the Roland TB-303 and TR-808".(6)*

One of the most important developments in the "virtual Synthesizer" industries was the introduction of Digital Signal Processing (DSP). DSP was a process of analysing and extracting information from signals. This information can then be used to synthesise replica signals and manipulate them. This opened up a world where computers were able to analyse the sonic characteristics of classic electronic instruments and recreate them using binary code. DSP was a revolution in the synthesiser world and meant that producers could now include software versions of classic instruments to their recordings, at a fraction of the price of the hardware version of the instrument.

Digital Signal Processing (DSP)

DSP implements a series of algorithms that analyse and extract acoustic information from the instrument and convert it into Binary data that can then be handled by a computer. The power of the DSP processing technique was showcased with the release of the Mini Moog V and the ARP2600V by Arturia. These were hailed as groundbreaking "virtual" instruments and showed the true power of DSP technology and how it could provide authenticity at a fraction of the cost. Douglas Self tells us in his book Audio Engineering Explained,

> *"Over the past 40 years, the field of digital signal processing (DSP)has grown from its origins as a collection of techniques for simulating the behaviour of analogue systems on digital computers into one of the most widely studied and universally used tools in modern technology".(7)*



This block diagram of a simple DSP process illustrates how the signal is received and how it is processed. (Image taken from http://www.absoluteastronomy.com)

Not only did DSP technology revolutionise the synthesiser industry, but it was incorporated into many everyday devices and allowed mobile phone designers and mp3 player manufactures to experiment with new ways to analyse, manipulate and recreate sound. Other programming languages designed for sound were developed such as Csound and Steinberg's SDK format for creating Vst instruments, however, DSP is widely considered to be the most used. For this report we will concentrate on DSP's impact on the "virtual" instrument industry and following on from research into the hardware versions of two instruments the mini moog and the ARP2600, we will look at the mini moog V and the ARP2600 V, Their "virtual" counterparts, and how accurately they recreated their analogue ancestors.

The minimoog V

The Mini moog V was Arturia's attempt at re modelling the classic original minimoog synthesiser. When first introduced onto the market it was hailed as a massive success, sounding as authentic as the original instrument, without the issues of unwanted noise and the inevitable instability that often came with analogue circuitry. It is an often made misconception that it was the analogue circuitry that would give the minimoog its unique tone and warmth, however, the minimoog V proved that it could provide equally "Fat" sounds without having to incur the drawbacks of analogue technology.

The first difference that was immediately apparent is that the original hardware synthesiser was a monophonic instrument; meaning only one note could be played at a time. This was mainly due to the technology available to the designers at the time; however, when the mini moog V was released it was Polyphonic allowing the user to play more than one note simultaneously. Although Arturia were aiming to recreate the exact authenticity of the original synthesiser, the decision was taken to make the re modelled version polyphonic as the technology was now available to make this possible, and it was also felt that Moog himself would've chosen this option had he had the capability at the time.

The second major benefit of the "virtual" version of the minimoog, the minimoog V, was its ability to store an unlimited number of patch settings to memory. Although the original Hardware version of the synthesiser was capable of producing amazing, groundbreaking sounds, they had to be noted and recreated manually which would place restrictions when it came to using it in a live performance environment. The minimoog V, however, had access to the storage capabilities of the computer it was being run on, which in today's market can be almost limitless. As if to prove this unique feature of the minimoog V, it came with a vast array of pre programmed patches, some designed by world famous artists such as Arthur C. Colombo and Jean Michel Blanchet, that provided the user with a good "jump off point" to create their own patches from.

(Image taken from http://www. vintagesynth.com)

The Arturia minimoog V front panel is an exact replica of the original hardware synthesiser and provides modern users with a nostalgic snapshot of what early users of the analogue instrument would have been confronted with. As with the original mini moog, the minimoog V provides the user with three oscillators capable of generating triangle, saw tooth, square, Saw/triangular, wide rectangular and narrow rectangular waveforms. The mixer is also a homage to the original instrument providing on off switches for each of the three Oscillators and a volume control knob for each. The noise generator is identical to that of the hardware version of the synthesiser and arturia have even included an external stereo audio input.

The minimoog V has a 24db/octave Low Pass Filter which shows the power of DSP technology as it creates an extremely accurate version of the Low Pass Filter contained in the original Synthesiser. Although the capabilities of modern "virtual" would allow them to create a low pass filter that far superseded those of the hardware version, Arturia has spent a lot of time trying to recreate the original sound manipulation offered by the hardware version. This was widely considered to be where the original minimoog's power lied and it was important that Arturia got the details right for the minimoog V as it was this Low Pass Filter that gave the minimoog a lot of its distinctive tones and any digital re creation would have to offer the same tones. The Low pass filter of the minimoog V offers the same six parameters as the original minimoog, namely Cut off, Emphasis (resonance), Amount, Attack time, Decay time, and Sustain time.

The Voltage Controlled Amplifier\ (VCA) also contained the same controls for Attack, Decay and sustain as the original hardware version which were accurate in the way they affected the tone being generated. Apart from the inclusion of certain modulation controllers such as Pitch wheel, Mod wheel, Glide, Legato and Pitch bend and a digital replication of the original built in A-440 tone generator for tuning the instrument.

The ARP 2600V

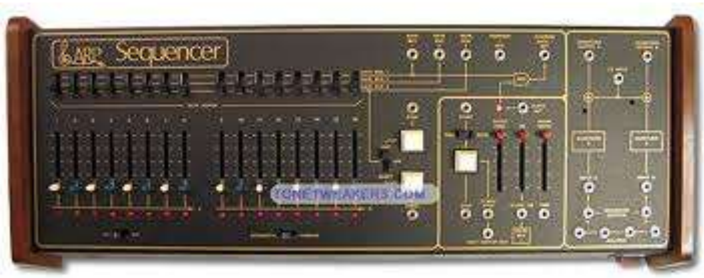
The original analogue version of the ARP2600 was considered to be one of the finest synthesisers made, and much in the same way that they developed a software version of the mini moog, Arturia released a "virtual" version of the ARP 2600, namely the ARP2600 V.

Arturia's ARP2600 V.

(Image taken from http://www.vstidownload.com)

The ARP2600 was a huge success, and its unique sound was made famous by top selling artists such as Herbie Hancock, Stevie Wonder and Jean Michel Jarre. The impact that the ARP 2600 had on the music industry as a whole meant that its "virtual" descendant had a lot to live up to. During a time when computer technology had only just been discovered and the power that the micro chip could bring was utilised, the ARP2600 raced miles ahead of its competitors in the methods it offered to generate and manipulate sound and its semi modular design would increase its ability to synthesise sound. Although the original instrument was decades old, the task that faced Arturia in their quest to remodel the ARP2600 "virtually", was made considerably more difficult by its use of patch cables that could be used to re route sound throughout the different modules, giving the user an unprecedented amount of combinations to experiment with.

Again, Arturia had a lot more technology available to them than the designers of the ARP2600, and had to work hard not to improve too much upon the original design. They did however include a couple of additions that they felt still held true to the fundamental processes employed by the original synthesiser. Four extra digital tracking generators would improve upon the one tracking generator that was part of the hardware version, as well as extra delay and chorus effects available to the use. However, the biggest addition to the ARP2600 V would be the introduction of the "virtual" ARP sequencer that would have been bought separately when purchased as a hardware device in the 1970's.



The original ARP Sequencer as it looked in the 1970's

(Image taken from http://www.vintagesynth.com)

The ARP sequencer employed a number of crude analogue processes to provide the user with sixteen sequencer steps that could control other synthesisers. This technology would be a breakthrough in how synthesisers were programmed and would have a direct reflection on how sequencing would develop in the years that followed. When used in conjunction with the ARP2600 keyboard players were given a lot more control and it was this that inspired Arturia to include this sequencer in the ARP2600 V.

The ARP2600V stayed faithful to its predecessor and employed three Voltage Controlled Oscillators the second of which could be used to modulate the other two Oscillators or the

Filter. A Low Frequency Oscillator was available, that could be synchronised to the ARP sequencer that had been added. When it came to the main features of the ARP2600V, Arturia again remained true to the original design and included a simple Low pass resonant filter, a sample and hold generator and a Ring Modulator. These features used modern DSP technology to accurately recreate the processes involved in generating sound with a specific combination of hardware, and were highly acclaimed by critics who said it captured the essence of the original classic synthesiser. Also included were digital recreations of the original ADSR and AR control envelopes as well as a version of the original envelope follower that was found on the ARP2600.

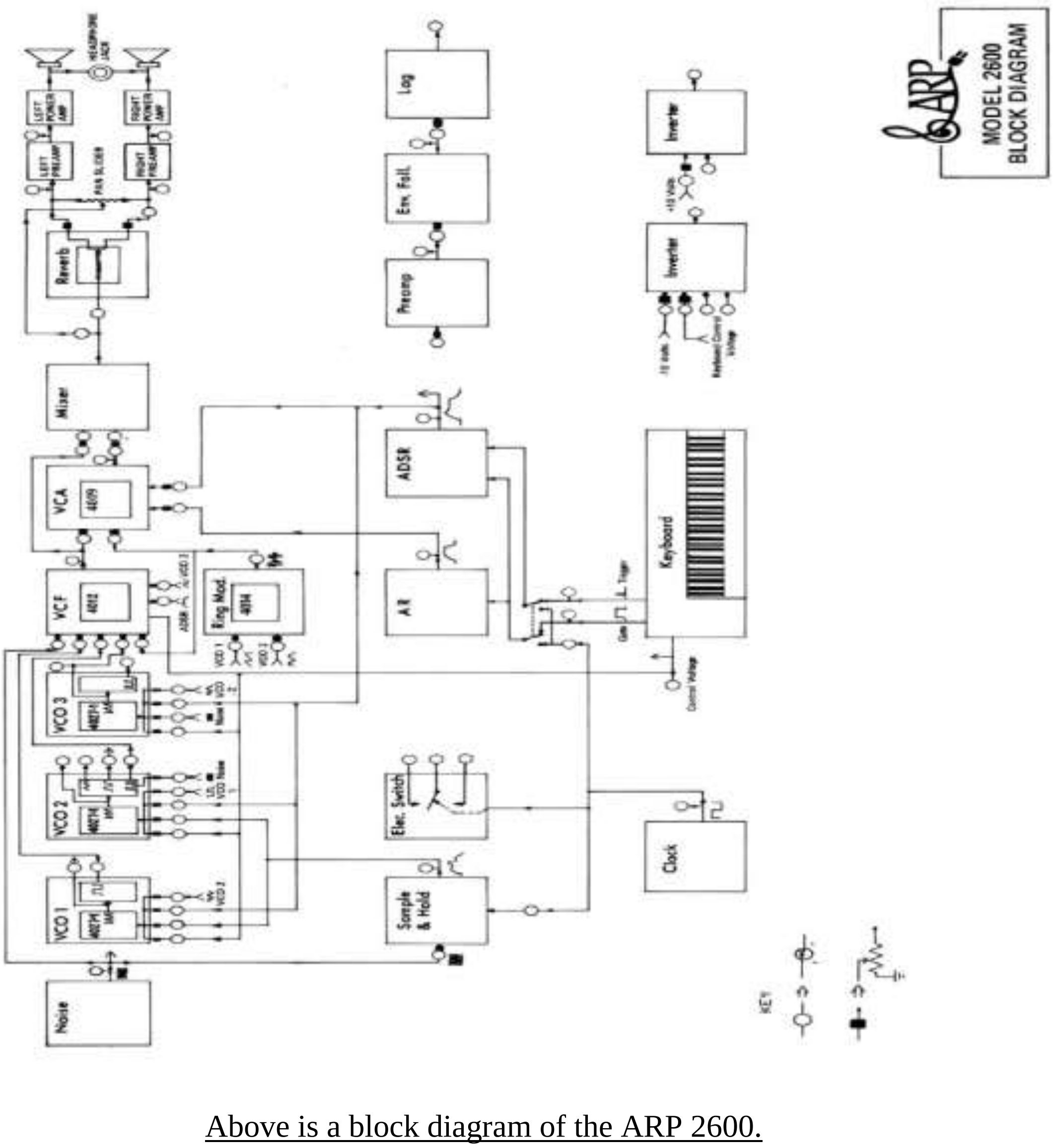
Although the original mini moog is considered to be a groundbreaking instrument in the history of electronic synthesisers, it is hard not to concede that the “virtual” or digitally recreated version, the mini moog V, is more versatile and a lot more capable than the original instrument. The mini moog V has benefited from decades of computer technology development and was able to outperform the original Hardware mini moog tenfold, however, they have remained faithful to the original design in order to generate sounds that accurately capture the warmth and analogue qualities of the hardware mini moog. In conclusion, the mini moog V has done an amazing job at recreating the mini moog synthesiser without the technological drawbacks that original designer, Robert moog, had to deal with during the 1970’s. The “virtual” instrument is a lot more stable and obviously easier to transport than the original, however, it wasn’t for the moog’s creativity and engineering genius, Arturia wouldn’t have had an instrument to re model in the first place.

The original ARP2600V caused a ripple throughout the Music industry and is widely considered one of the pioneers of synthesiser technology and the “virtual” version, the ARP2600 V was also well received by modern artists and producers. In an age where producers rarely leave the desktop, the ARP2600V would give them the opportunity to get their hands on equipment that is rare in today’s market. Again critics argue that digital methods cannot accurately recreate the warmth that analogue technology provides, however Arturia again succeeded in emulating the analogue characteristics of the early synthesiser whilst reducing the gremlins such as noise and instability that were common place in the ageing analogue instrument. Although the interface of the ARP2600V can be confusing to anyone not knowledgeable about synthesisers, Arturia still managed to capture the signal flow from left (oscillators and audio inputs) through the mixer and filter, before leaving through the effects and output amplifier, giving the user a more immersive experience of the ARP2600.

Analysis and appraisal of remodelled ARP2600 synthesiser and processes involved in creating it.

Introduction

Developing on research into the classic Synthesisers the mini moog and the ARP2600, I have chosen to remodel one of them as an ensemble in Native instruments Reaktor. Of the two instruments, it was felt that the ARP2600 offered the biggest challenge and therefore would be perfect to remodel. Armed with the original circuit diagram from the ARP2600 original user manual and all of the knowledge I had gained as part of this report, I was able to get an idea of what circuits were contained within the instrument and how they were wired together. This information could then be used to build my own “virtual” version of the synthesiser.



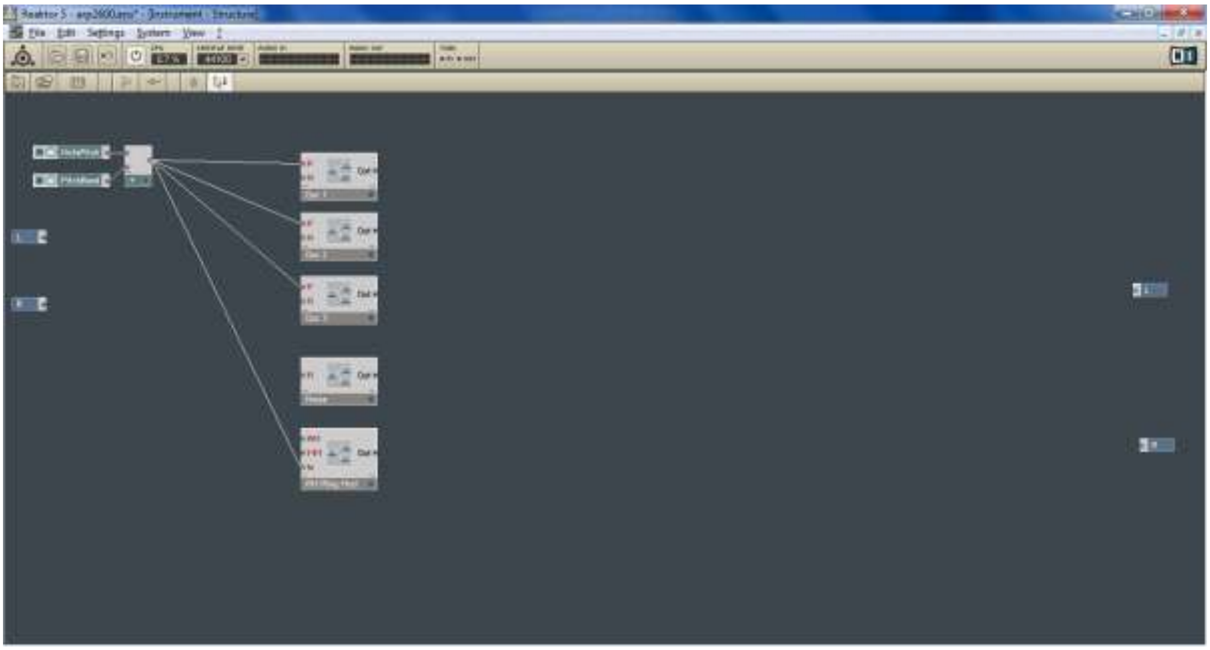
Above is a block diagram of the ARP 2600.

(Image taken from the ARP2600 user’s manual)

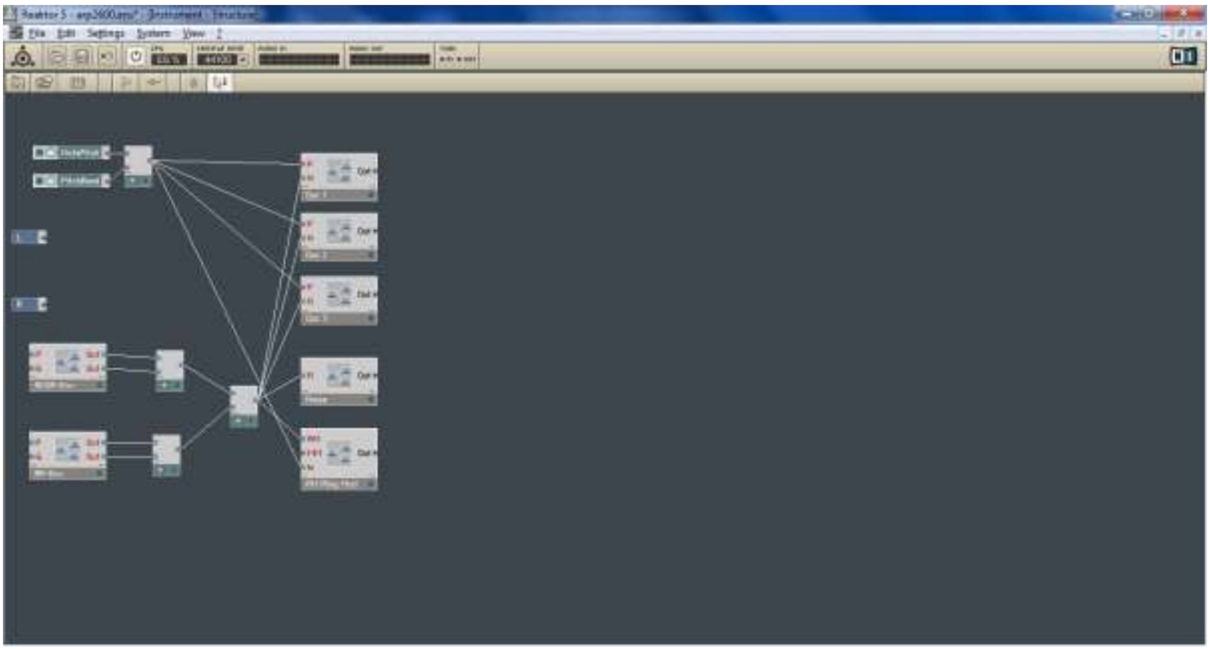
Although the aim of the exercise was to create my own version of the ARP2600, I knew it would be impossible to recreate an exact digital replica and therefore decided early on that I would try to create an instrument that contained all the fundamental elements contained within the original hardware design, however not be an exact visual representation. As Reaktor is such an advanced piece of software, its capabilities have surpassed those available to the designers of the original instrument and therefore I had to achieve certain aspects of the original design using modern methods. For instance, it is extremely hard to create an exact replica of the original Voltage Controlled Filter (VCF) as the most basic of the VCF's available in Reaktor can still outperform the original. Therefore I found the need to add the VCF's available to me in Reaktor and then remove some of their parameters until they more resembled the original.

It was felt that the best place to start would be to create the three "sources" that would generate the waveforms that would then be routed through the rest of the circuitry. I already knew that the ARP2600 was designed with three oscillators, the first and third of which capable of generating both saw tooth and pulse waves and the second which was capable of generating triangle, saw tooth, Sine and Pulse waves. I began by setting up my instrument in Reaktor until I was faced with an empty structure window. The structure window is where I would add all of the devices that made up my instrument and wire them together to achieve the signal flow I wanted. By right clicking on the empty structure window I could access all of the different modules and building blocks available to me in Reaktor that would make up my finished instrument. I added three Oscillators from the "Macro" menu and selected which waveforms each was capable of generating in relation to the original design of the ARP2600. Next I created a ring modulator and a noise generator that would complete the "source" group and give me basis for my "virtual" instrument.

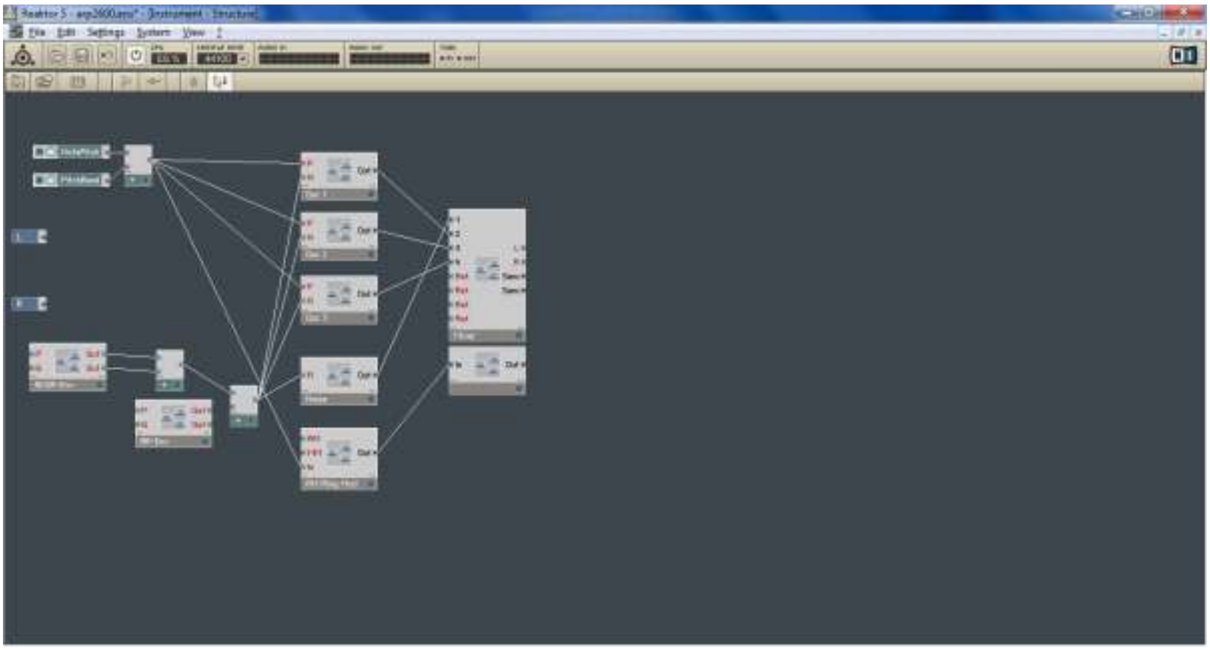
In order to allow the keyboard to send information to the Oscillators, I needed to create note pitch and pitch bend MIDI parameters. Using a Mathematic parameter I could then combine the two separate outputs of the MIDI controllers to a single output that could be wired to the Pitch Control input of each of the Oscillators and the Noise generator. This meant that every time a note was hit on the keyboard or the pitch bend was rotated it would send this information to the oscillators and the noise generator.



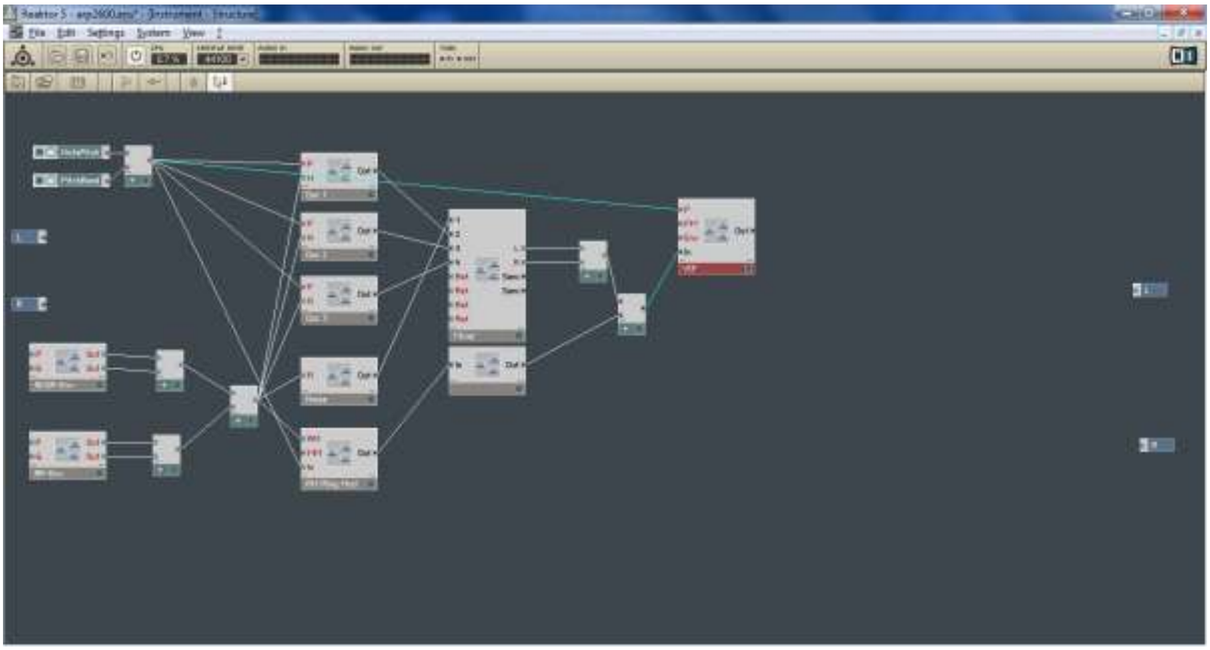
The next job was to add the envelopes that would control the amplitude of each of the Oscillators, the Noise generator and the ring modulator. Two envelope controllers, Attack Decay Sustain Release (ADSR) and Attack Release (AR), were added to the structure window by right clicking and selecting them from the “macro” menu. Using a sequence of the mathematical “add” parameter, I could combine the output of both envelope controls to a single output that was then connected amplitude controller of each of the Oscillators, the ring modulator and noise generator, as shown in the following screen dump.



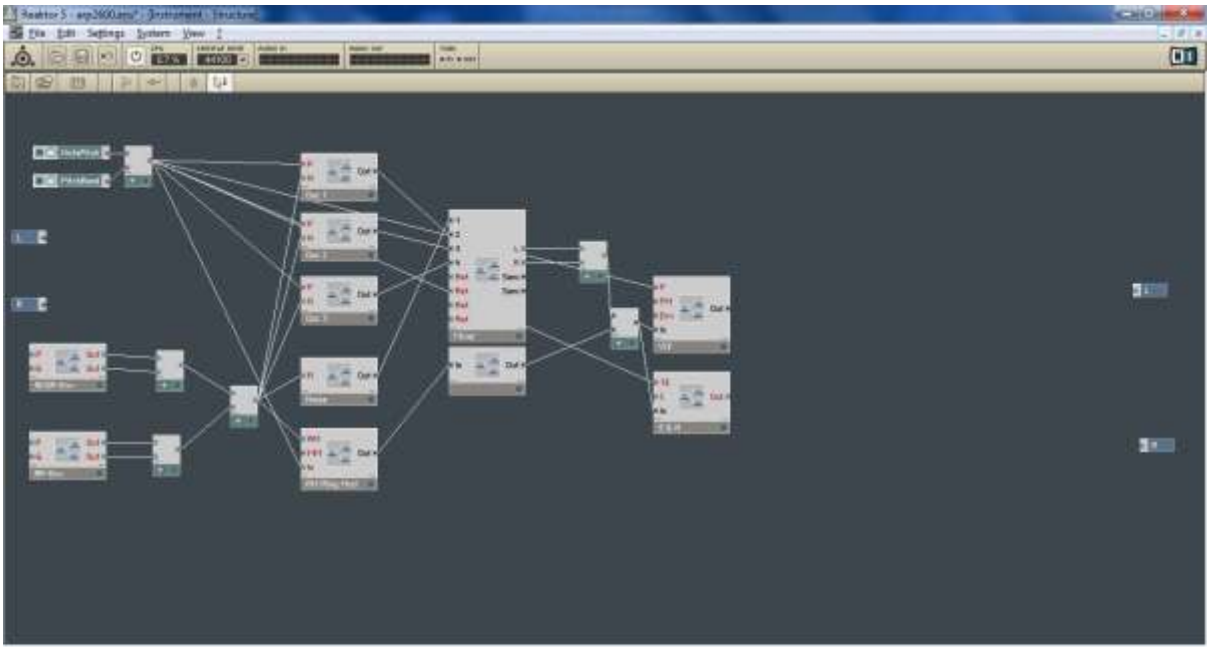
Next was the Instruments mixer that would allow the user to balance the amount of each of the “sources” that made up the overall sound being generated. As there were five “sources”, the three oscillators, the noise generator and the ring modulator, I needed to add a mixer with at least five inputs. The most simultaneous inputs available in Reaktor part of a four channel mixer, therefore I needed to create a single channel mixer that would sit next to the first mixer, giving me the extra channel I needed. The output of the noise generator was wired through to the mixers first input with the three oscillators' occupying the following three inputs. The output of the ring modulator was then wired to the input of the single channel mixer giving me the complete five channel mixer I wanted that could be altered using the controls in the instruments interface.



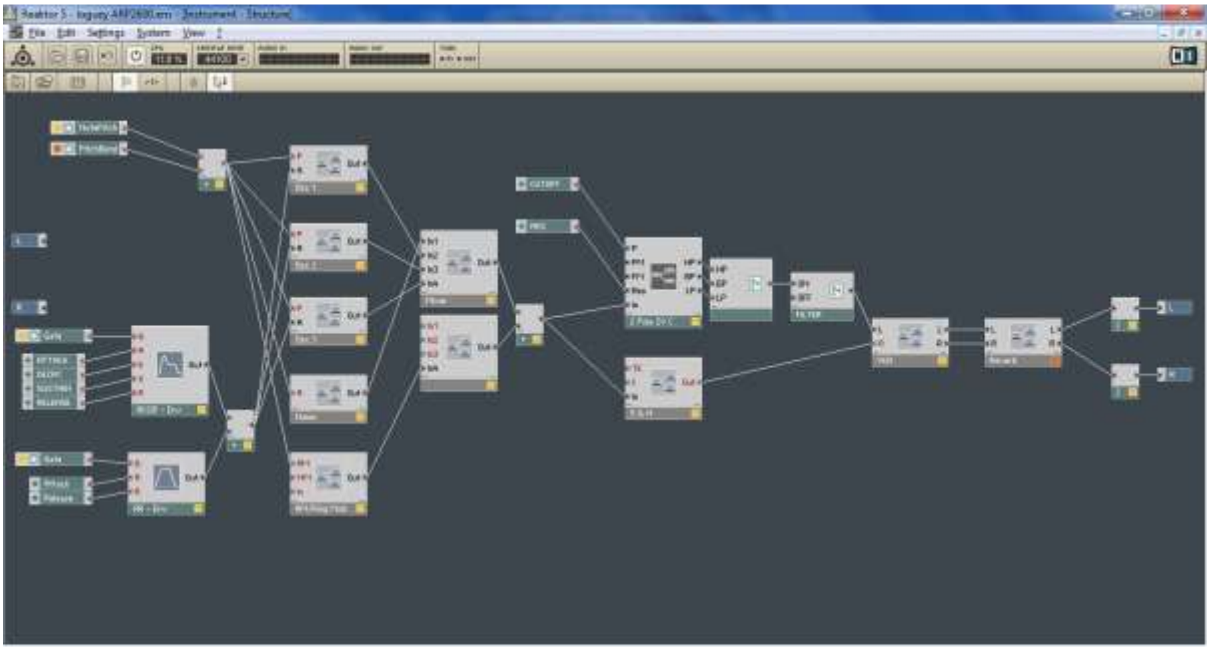
Now it was time to concentrate on the Voltage controlled Filter that was made popular by the original ARP2600. Using a combination of mathematical addition parameters I was able to combine the stereo output of the four channel mixer with the mono output of the single channel mixer to create an overall mono signal that would be wired to the input of the Voltage Controlled Filter. A VCF was added to the structure window from the "macro" menu and the mono output signal generated from the mixer channels was wired to its input. The output of the MIDI note pitch controller was then wired to the note pitch input of the Voltage Controlled Filter. This now meant that every sound being generated and every envelope being applied, was now being routed through the Voltage Controlled filter and could be manipulated using its powerful band pass filter and resonator.



“Sample and Hold” was a big feature of the original ARP2600 and using reactor I would be able to add this component to the structure of my instrument simply from the “macro” menu. I needed a source to connect to the input of the sample and hold generator that would actually be sampled. The output of the mixer channels was routed to the input and the output of the MIDI parameters was connected to the Trigger Event input. This then allowed the keyboard to trigger the sampling of the sound that was then held on loop and could be modulated using the Voltage Controlled Filter.



Using the original ARP 2600 users’ manual I could see that the final stage of the signal flow was first amplified using a Voltage Controlled Amplifier and then sent through a primitive reverb circuit. Using a mathematical addition parameter I was able to combine the overall signal being sent from the Voltage Controlled Filter with the output signal from the Sample and Hold generator into a single signal that was then attached to the input of the Voltage



<u>Appraisal</u>

Firstly, and most importantly, the interface does not resemble the original ARP2600 interface, however, it contains the modules and controllers that you would expect to see on the original. The ARP 2600V by Arturia was a monumental piece of software development, but with the resources available to me I felt it best to simply concentrate on the circuitry and signal flow of the original ARP 2600 and not its aesthetic value. The Sample and hold generator and ring modulator were directly inspired by the original instrument and when combined with oscillator waveforms and the noise generator could produce interesting sounds reminiscent of the early analogue instrument. The need to incorporate a reverb circuit into my instrument was an interesting exercise and helped me appreciate the ground breaking but primitive reverb that was used by the original ARP2600 and how these kinds of effects have developed with the dawn of computers and "virtual" instruments.