Fundamentals Of Music Technology

Volume One:

The Arp 2600 S•Y•N•T•H•E•S•I•Z•E•R

SECOND EDITION

A Pedagogic Work in Elementary Synthesis
With Sampling and Example CD

by

SAMUEL ECOFF

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Secret Society Productions

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Typeface: Times

This book is printed on acid-free paper.

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3	1.73
4	3.79
6	1.67
8	2.82
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35	1.10
36	1.10
37	1.04
38	1.06
39	1.08
40	1.01
41	1.3
42	1.1

CD LICENSING AGREEMENT

This book is dedicated to all of the people that love this wonderful instrument as much as I do; to the people who know that the patch isn't complete until every available patch cord has been used.

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Track	Page	Note(s)	Description
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24	49	---	This track has intentionally been left blank due to an error in printing in the book.
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Track	Page	Note(s)	Description
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Track	Page	Note(s)	Description
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How the CD was Recorded

The audio CD which accompanies this book was recorded by connecting the left and right outputs of the ARP 2600's mixer section to a Mackie LM-3204 mixer. The incoming sounds were compressed slightly with a Behringer Composer compressor before being routed to a Digidesign 882 audio interface connected to a Pro Tools|24 system. The ARP 2600 was then recorded into Mark of the Unicorn's Digital Performer 2.61MT hard disk recording software running on a Macintosh G3. It was then edited so that each example began and ended in exactly the right spots and was mastered with plugins from TC Works and MOTU. Other than the aforementioned gentle compression, no effects were applied to the incoming sounds from the 2600.

The 2600 was played part of the time from its own keyboard and part of the time from a Fatar SL-880 mother keyboard and Digital Performer through a Paia MIDI2CV8 converter. Many of the melodic samples were progammed into Digital Performer to insure timing accuracy and consistency. While purists may argue against the use of MIDI in controlling an ARP 2600, the author was left with no other choice as a capacitor in the keyboard's control panel went bad only a week before the final recording session for the CD, and the repair unfortunately could not be completed in time for the final recording session. (Special thanks to Tim Smith of The Audio Clinic who restored the keyboard control panel to working condition).

When melodic patches were recorded (i.e. pitched sounds) an effort was made to make them available at many different pitches for reader who may wish to sample them. These pitches can then be used to create a multisample which yields the highest amount of accuracy in sample playback.

The samples associated with Section 15, were created using the ARP sequencer rather than Digital Performer running through the Paia converter. As a result, some drift is noticable in tuning and timing stability.

PREFACE
to the first edition

This book is the culmination of years of work and study into the pedagogy of music technology, and I fear it is also just the beginning, as there will always be more to learn about this exciting new field. I have little hope of these volumes catching on as standard works, as they are highly instrument-specific. However, I feel that they have pedagogic merit, and where all else fails, they could even substitute for an owner's manual in a pinch.

This book is intentionally printed on every other page so that the student may have a convenient place to take notes, write questions about readings, and record observations during experiments.

As with any field that is in its infancy and is still rapidly evolving, it seems that there is no good way to go about writing about music technology. Either a text is so instrument-specific that it becomes outdated very quickly (within five years or so) or it is so general that it is of little merit to the beginning student. I have elected to opt for the former path, as I have consistent access to the instruments in question. While this is of the greatest value to me, it is of very little assistance to anyone else who might be interested music technology in general.

Because I have always taught these lessons in very small groups or as private lessons, I have always taught them using an outcome-based approach. I have given students a reasonable number of chances to correct their mistakes and improve their knowledge, as well as improving their grade. I have required my students to pass each quiz at a minimum of the eightieth percentile.

So, I commend this book to the reader... Get what you can out of it. For students who are about to study music technology privately and will be using these tomes as a course book, I can only say.... be prepared in every way possible! Also be forewarned that questions that are missed on quizzes have a nasty habit of showing up on the final examination.

Sam Ecoff

January Seven, 1999
Wales, Wisconsin

PREFACE
to the second edition

Over the course of two years of teaching music technology, I have stumbled (mostly blindly) upon several observations as to which students are generally successful in their studies of electronic music and which students generally fall by the wayside. It seems that it is the students who have a passion music technology are the students that are most apt to succeed. This observation would seem really rather obvious at first, but the more one contemplates it, the more ramifications it has.

First, students need to make a commitment to music technology if they are to study it. Although there is a great deal to know about other musical instruments, piano for instance, relatively little has changed in the design and playing technique of the piano in the last ten years. In the music technology industry, the last ten years have seen one revolution after another including the rise of the home MIDI studio, digital audio recording for the average musician, and finally, the rise of the complete home project studio which is actually able to compete in terms of quality with major production facilities. Because technology is evolving at such a rapid pace, students must be even that much more dedicated to the task of mastering as much information possible. In this wonderful day of instant information, gathering information is no longer the great challenge to the student, but rather taking time and finding the energy to master all of the information which is at the student's fingertips.

The second observation I have made is that some students wish to learn about music technology in the 'better-faster-cheaper' mode, which accomplishes little. To these students, understanding the mechanics and theory of one oscillator frequency modulating another is a complete waste of their time, and they would much rather just call up a preset on a modern synth which will in their minds do that work and thinking for them. One must understand that there are always greater possibilities when one can understand the theory of synthesis which stands behind the sounds, and when a musician is given full access to all of the parameters of sound available instead of three knobs for 'realtime control' and a bunch of ROM presets sporting today's latest flavors.

Indeed, there is nothing wrong with using preprogrammed musical patterns and combining them with other sounds to create a new kind of music, but there is a fine line between a musician and a technician. While the technician assembles premade parts and works logically with machines to produces sound, a musician will actually create new loops and adds the dionisian element of the creation of new sound. As synthesists, computer operators, composers, arrangers, and music technologists, it is important to keep both the hat of the technologist and the hat of the musician at the ready so that we may freely and readily switch between the two. Perhaps that is the most important part of music technology: It is not about being one-dimensional or about confining oneself to a single role. It is about exploring all of the possibilities and about trying all of the parameters. When access to parameters is denied, either by companies who produce equipment advertised to fill the role of pro gear or by people who shut out different possibilities in music technology, it is the music that suffers.

Thank You

This book is and has been a collaborative effort, as many such large undertakings are. I would be truly remiss if I missed this opportunity to thank the following people for their assistance in completing this text. It is, I feel, important to note that many of them performed their services entirely *gratis* because of their love of the subject.

I would like to thank Dr. Michael Cunningham who introduced me to the ARP 2600 Synthesizer during my undergraduate degree at the University of Wisconsin-Eau Claire. He also deserves credit for coining the term “redundant patch.”

I also owe a great debt of thanks to my loving fiance, Kara for all of the hours she spent proofreading and inputting corrections on a subject which she cares about only for my sake. She was also incredibly helpful during the recording sessions for this book, ‘wo’-manning the digital audio workstation to leave me free to concentrate on the creative aspect of creating patches.

I must also thank my internet friend Roger Lesinski whom I have never met, but has provided wonderful insights and new thoughts into the technical side of this book, and for his great proofreading skill.

This book would still be sitting gathering dust on a shelf as a twenty-one page outline if it were not for the many students whom I used as guinea pigs while I was developing this book. I owe them a great debt of thanks for their continued patience and also their assistance in proofreading. (It is sometimes embarrassing to admit that 10-year old students found many errors that I and the rest of my proofreading team missed!)

I would be remiss if I forgot to mention Ihor “E” Tanin of “E” Lectronix Rock ‘n Roll Hospital in New Berlin Wisconsin. Not only did he restore my ARP 2600 at a fantastically low price, he also put up with my phone calls three to four times per week for several months. I also owe a great debt to Timothy Smith of The Audio Clinic/Weyer-Smith Labs in Billings, Montana. He did a wonderful job of repairing my broken 3620 keyboard, and his knowledge of the 2600 was truly amazing and invaluable.

Finally, I would like to thank my uncle, David Reed who ever so kindly supplied me with the paper I needed to print the very first copy of this book when I was too poor to purchase paper myself, and to my parents who have always supported my efforts, and who put up with years of bleeps and bloops coming from their basement while I learned how NOT to program synthesizers. To all of these people I am grateful!

About the Format of This Book

This book has many facets and serves many purposes to many people. While it is primarily geared towards an academic setting where the basic concepts of subtractive synthesis may be introduced, it can also be of value in other ways, which are best left to be discovered by the reader.

This book does not start from ground zero. It assumes that the reader has a small amount of knowledge in the area of basic acoustics. It is important to understand how sound travels, the concept of harmonics, frequency and how it is measured, basic waveforms and their harmonic content. It is common practice to begin a book such as this with a short chapter on acoustics, but since there are so many excellent books which cover these topics on a very accessible level, these topics have been omitted from this book. For persons interested in reading these books (it never hurts) a short list can be found in Appendix One.

The book itself is grouped into five units. These units are then split into parts called *sections*. I felt that this was a more appropriate term than *chapter* since modular synthesizers are sectional devices by nature. Each section has several subheadings and illustrations. Following each section of text is a set of experiments that should be performed on the instrument. There is no substitute for hands-on experience. Following the experiments are a set of review questions and a list of all of the important terms which were introduced in that section. These will primarily be of interest to persons in an academic setting, but can also serve as a memory refresher for the casual reader.

The rear of the book features a glossary of terms, including some background terms which are not included in the text itself. An index is also present for easy reference of terms and concepts.

This book includes an audio CD which contains sounds played on an ARP 2600. This disc serves three purposes. First, it allows people to check the results of their experiments to see if they have come up with the correct sound. Secondly, it allows people who do not have access to a 2600 to hear the results of each audio experiment and some examples in the text. It will also allow them to hear what this marvelous instrument can do. Finally, it can be used as a source of analog synthesizer samples for a sampler. (Please read the sample use agreement on page ii if you intend to use the CD for this purpose. The license granted to you is fairly unrestrictive, but there are certain legal obligations which must be met if the disc is to be used for this purpose.)

One final note about this book is that in many of the examples, the subject in the experiment is referred to as 'Bob' or 'Wendy.' This is in honor of Dr. Robert Moog and Wendy Carlos. Dr. Moog invented the first commercially available synthesizer and invented many of the modules described in this book. Wendy Carlos is an excellent musician/composer/inventor whose wonderful recording "Switched-On Bach," performed on Bob Moog's Series IIIp synthesizer, still holds the record as the best selling classical album of all time.

SECTION 1

General Controls

Introduction and Background

The ARP 2600 was designed and manufactured at a point in time when synthesizers had just emerged as a musical instrument (the late 1960's), and most people had no idea how to use and program them. Because of this, the ARP company designed a synthesizer whose primary purpose was to teach people about synthesizers. The ARP 2600 was manufactured from 1970 to 1980, which is a very long production run for a synthesizer by today's standards. Its designers did everything they could to make it easy to understand. For instance, all of the controls are laid out so that when creating sounds, they start at the left side of the synthesizer and move towards the right. This is the way most sounds are created, just letting the electronic signals flow from left to right. The ARP 2600 is much like an assembly line in this way. Each part adds to or changes the sound a little bit until a finished sound emerges at the end. The 2600's designers also used white diagrams on the instrument's front panel to attempt to show users where signals were flowing within the instrument.

Figure 1-1 The ARP 2600

In Figure 1-1, one can see that the ARP 2600 is actually two separate parts: A keyboard unit and a cabinet unit. The keyboard must be connected to the cabinet in order for the keyboard to function, because the keyboard draws power from the cabinet. However, it is entirely possible to use the ARP 2600's cabinet without the keyboard attached. It still functions perfectly well. In fact, many of the experiments in this book do not require the keyboard.

Figure 1-2: The keyboard's multipin connecting cable

The connection to the keyboard is established with a single cord. The cord is permanently attached to the keyboard at one end, and has a multipin plug at the other end. This design was changed several times by ARP, and it is entirely possible to find keyboards made for the 2600 which do not follow this design. (e.g. some earlier models have cables which can be unplugged from both ends.) A model 3620 keyboard was used for purposes of this book. Notice in Figure 1-2 how many connectors there are on the plug which connects the keyboard, and keep this information in mind. For now, it is just necessary to know that the keyboard receives power from the cabinet through this cable.

The Balance of Power

The ARP 2600 gets power from a household electrical outlet via a three-prong cord which plugs into the right side of the cabinet. The pins are aligned in such a way that the cord cannot be plugged in upside down. However, the plug that connects with the AC outlet is not polarized and can be connected in either direction.

The main power switch interrupts incoming electricity so that the ARP cabinet and its keyboard can be switched on and off. It is located at the lower right hand corner of the cabinet, just above the headphone jack. (See Figure 1-3) Notice that when the synthesizer is switched on, the red light above the switch goes on. This is the only visual indication that the power is on. There is not a separate on/off switch for the keyboard; it is switched on when the cabinet is switched on. When turning the synthesizer on, it is always a good idea to make sure that the synthesizer has been zeroed (see below) and that there are no additional cables connecting the 2600 to other devices in the studio. This insures that no damage will be done to the synthesizer or other studio devices, and that the synthesizer isn't gong to make some sort of a terrible squealing sound or something worse.

Figure 1-3: The ARP 2600's headphone jack, power switch, and indicator light

Speaking of Which

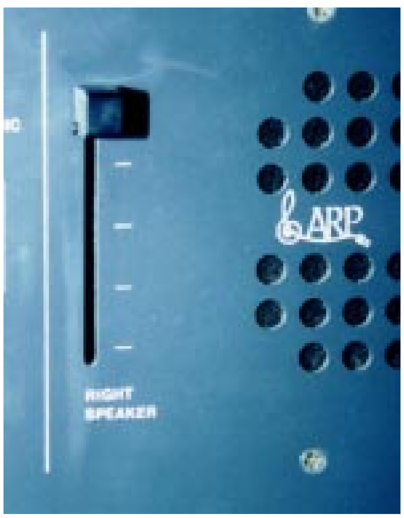


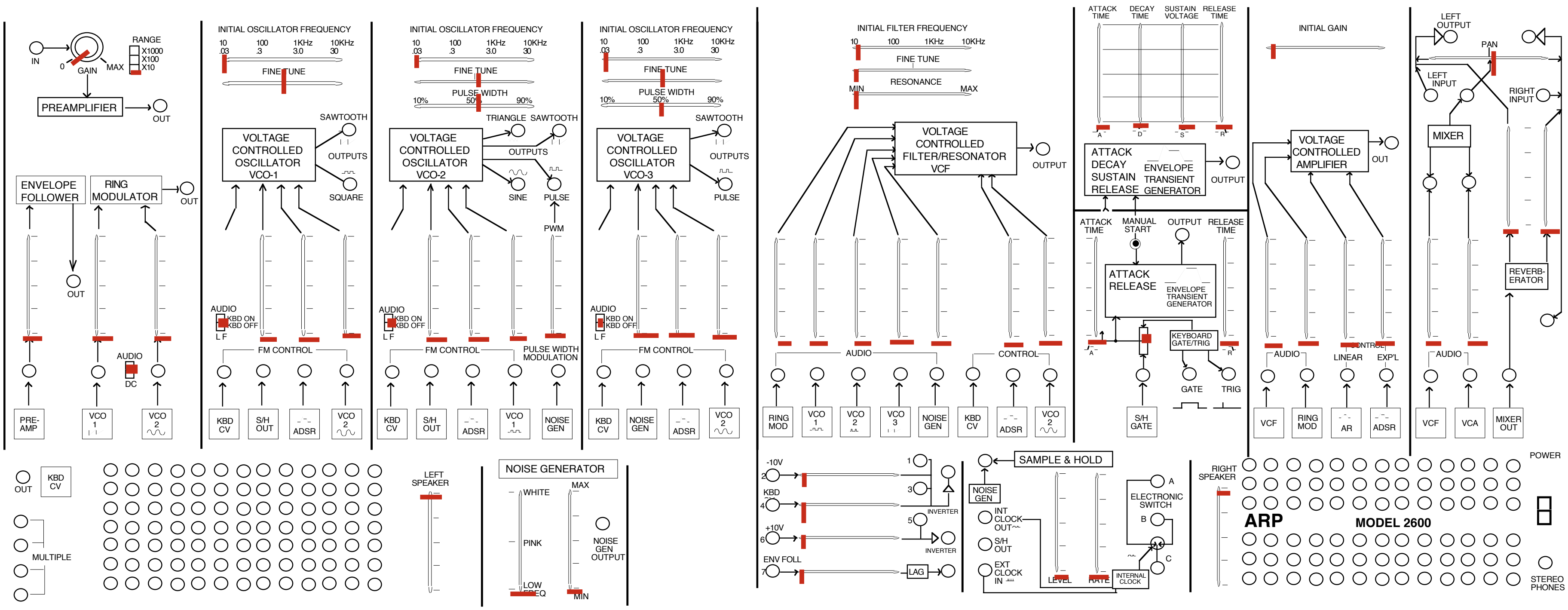
Figure 1-4: A speaker volume control

One can see that the ARP 2600 has built-in speakers. Each speaker has its own volume control. This control is pictured in Figure 1-4. The ARP also has a quarter-inch jack into which one can plug a pair of stereo headphones. The headphone jack is located just below the main power switch on the cabinet. (See Figure 1-3) Although it accepts stereo headphones, the ARP 2600 is a monaural synthesizer. (I.e. the same signal is fed to both the right and left earphones) The only exception will be explained in Section 9.

On some synthesizers, plugging headphones in will interrupt sound going to the speakers. On most professional-level synthesizers, this won't happen, (most pro-level synthesizers don't have speakers) but the ARP was designed before many of the professional standards were developed, and plugging in headphones cuts off the speaker's output entirely, even if the volume level is set as loud as possible.

Zeroing the Synthesizer

Sometimes when a student starts to use the synthesizer, someone else has been using it before them. This can make working on the synthesizer very frustrating, since one doesn't know how the last person was using it, and some switch or fader might be set in a way that would keep the synthesizer from functioning the way it normally would. It is best to return all of the knobs, faders, and switches to their original position, and to remove all patch cords (see page four) from the synthesizer to prevent this sort of frustration. This is called *zeroing the synthesizer.* The synthesizer should be zeroed each time one begins using it. When attempting a new sound, it is also wise to zero the synthesizer, as the instrument might not behave the way one expects because of some earlier setting. Diagram 1-1 on page 3 illustrates the proper settings of each knob, switch, and slider when zeroed. Notice that all patch cords have been removed.



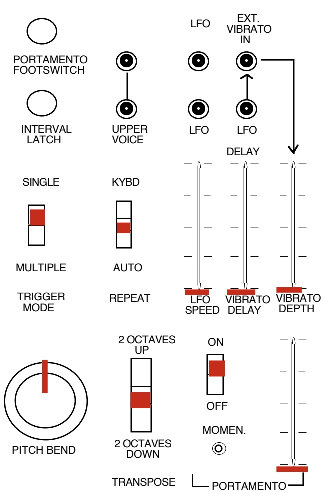


Diagram 1-1 indicates the proper setting of each knob, switch and slider when the ARP 2600 is zeroed. The upper diagram represents the 2600's cabinet while the small lower diagram represents the controls on the keyboard.

ARP made several different versions of the 2600. These are easiest to tell apart by the markings on the cabinets. The earliest models featured blue metal cabinets and a long wooden handle across the top. This model also lacks fine tune controls on VCO-1 and the VCF. While these models were very stylish, they were not particularly road-worthy. Later models featured a gray cabinet face in a wooden box covered in black Tolex (a vinyl-like substance which is very durable). These models also had a small plastic handle on the top of the cabinet and on the keyboard. These models are the most common version of the 2600, and one can be seen in Figure 1-1. The last 2600's ARP produced had a dark gray face with orange and white lettering, again in the Tolex case.

ARP also produced several different models of keyboards. The last ones produced have significantly more features than the early models (more on this in Section 13). The keyboard controls shown in the diagram are those from the model 3620 keyboard, which was the last model ARP produced.

Diagram 1-1

Parameters and Values

Soon the synthesizer's functions will be explained, but it is important to first understand the concept of a parameter. A *parameter* is simply something that one can change. A *value* is one of the possible settings of a parameter. For instance, if Bob looks at a light switch, he can see that the switch itself represents the parameter. It is something whose value he can change. This parameter has two possible values: On and Off. A fader, on the other hand, is said to have an infinite number to values, although its range of values may be measurable.

Patch Cables

Thin cables called *patch cords* or *patch cables* (See Figure 1-5) are used to connect different parts of the synthesizer together. They consist of two plugs which have been soldered to either end of a length of wire. This wire can be of any length. Some setups offer cables of just one length, while other setups have many different lengths of cables. The cables ARP included with the 2600 were all of the same length, but few of them are still around today as the wire has usually deteriorated to the point where the cables are unreliable. Many owners of 2600's today either purchase cables from companies which specialize in cables or make their own from parts acquired from electronics stores and supply houses.

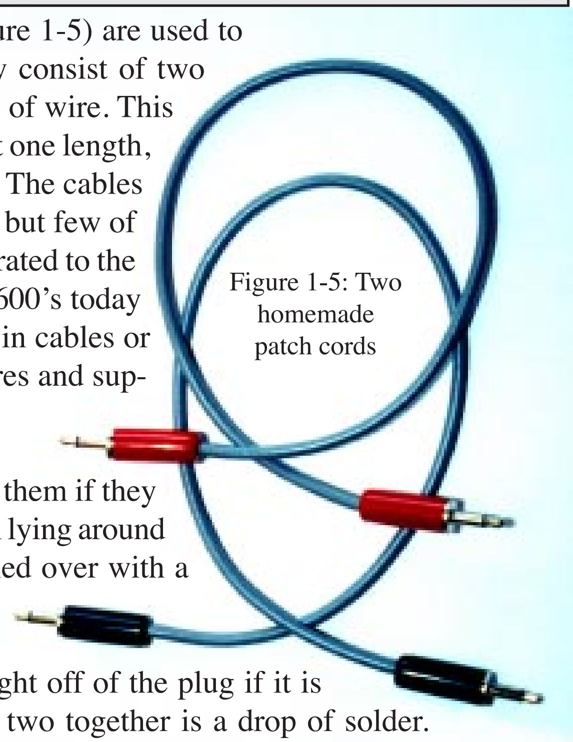


Figure 1-5: Two homemade patch cords

Patch cables are pretty durable, but one must take care of them if they are expected to last a long time. First, don't ever leave them lying around on the floor as they can be stepped on or worse yet, rolled over with a chair. Second, whenever a patch cord is removed from a jack in the ARP's cabinet, pull it out by the plug rather than by the cord. It is entirely possible to rip the cord right off of the plug if it is pulled hard enough, because the only thing holding the two together is a drop of solder. Third, do not bend the cable itself at tight angles, as doing so can actually sever the wire inside the casing. Finally, when finished with them, patch cords should be stored in a safe location, away from extreme heat and off the floor where they could become damaged. A simple hook mounted on a wall or the side of a table is a great place to store patch cables.

Many studios use two different colors of patch cables when patching the ARP 2600; red and black. The cables are identical other than the color of the plug and/or wire casing, and don't function any differently, but they are used for different purposes to make it easier to understand the way the synthesizer works. For audio signals, black cables are used. *Audio signals* are signals that are the raw sound that one eventually wants to hear. Red cables are used to carry control signals. *Control signals* are signals which one doesn't intend to hear and which will be used strictly to effect change on some other part of the synthesizer. (The difference between audio and control signals will become clearer in time.) The next section contains a great deal more information about control signals. For now, just remember that black is used for audio signals, and red is used for control signals.

Modular Synthesizers and Connectivity

A *modular synthesizer* is a synthesizer that is made up of several different discreet devices which can easily be seen and can be connected to each other in any order the user pleases. These devices are called *modules*. Almost all of these modules are housed in the synthesizer's cabinet. On the ARP 2600, it is possible to actually see the individual modules. They are separated on the front panel of the cabinet with heavy white lines. With larger modular synthesizers, companies often allowed users to pick and choose which modules they wanted to make up a particular synthesizer, and as such, the modules were entirely separate devices which didn't share a common front panel. On a truly modular synthesizer, these different modules are not connected to each other, and the user must connect them together using patch cords to create sounds. This last point is very important, so keep it in mind.

The patch cords are plugged into little holes on the modules called *jacks*. These jacks grip the ends of an inserted patch cord and make an electrical connection. The ARP 2600 uses 1/8 inch phono jacks (see Figure 1-6) and as such, patch cords must have 1/8 inch phono plugs. Although they all look the same, it is very important to understand that not all jacks are the same. Some jacks are inputs, and some jacks are outputs. A jack which allows signals to come in is called an input, and a jack which puts out signals is called an output. An input must be connected to an output. Likewise, an output must be connected to an input. Connecting an input to an input or an output to an output won't do anything at all. This is analogous to holding the handset of a telephone upside down. Before patching two jacks together, it is very important to make sure that one of them is an input, and one of them is an output. Otherwise, the connection being made won't do anything.

Figure 1-6: two 1/8" jacks

The ARP is really a good teacher in that it is very forgiving. If a silly connection is made, such as connecting an input to an input, or connecting an output to an output, it will not hurt the ARP at all. Just remember: signals can only come out of an output; they can not go in. Signals can only go into an input; they do not come out.

Modular: The Pros and Cons

There are some great advantages to modular synthesizers. First and foremost, one could connect the modules in any order. It is possible to come up with some pretty wild combinations which are not possible when dealing with a non-modular synthesizer (called a *fixed-architecture synthesizer*). Additionally, students can see each individual module and experiment with them individually, instead of having to use them in predetermined order.

There are, of course, disadvantages to modular synthesizers as well. First, to create a sound, one must use several patch cords. Secondly, all of the knobs and sliders must be reset for each different sound, as most modular synthesizers can't recall a programmer's sounds. Most modular synthesizers also allow the performer to play only one note at a time. Because of this, they are said to be *monophonic*. Many modular synthesizers are also becoming vintage instruments (older than 25 years) at this time and are becoming more and more unreliable. Despite all of these limitations, there is a large potential for making interesting sounds, and wonderful music.

Are Synthesizers Normal?

When sounds are created on the ARP 2600, certain modules must be connected in a certain way, and the appropriate knobs and sliders must be set just right to produce the desired sound. This collection of settings of patch cables, sliders, and knobs is called a *patch*. The term ‘patch’ comes from the patch cables used make these sounds. Modern synthesizers don’t use patch cables, but individual sounds are still referred to as ‘patches’.

All of this patching can be a lot of work, and many times, it is desirable to use the modules in a standard configuration (see Section 8 for more information). It would be very time consuming and monotonous constantly creating the same patches again and again, so the designers of the ARP 2600 came up with a good solution: normals.

What is a normal? A *normal* is simply a connection which is made to one of the ARP’s input jacks from one of the ARP’s outputs even before a patch cord is plugged into it. Another way to say this: Some outputs are internally wired to some inputs. All but eight of the ARP 2600’s inputs have something normalled to them. One can tell if an input has something normalled to it because there is some writing in a small white box that points to the input. The writing indicates what is normalled to that input. Another way to think of a normal is as a connection that is premade with an invisible patch cord. It is not possible for a user to change what is normalled to each input.

But What is Normal?

The normal represents the patch which is most commonly used. The ARP’s designers made the everyday connections into normals. They didn’t normal modules together that one would rarely connect. Thus, it is important to take note of which modules are normalled together, as this will give a student some clues as to how the synthesizer will ‘normally’ be patched. However, there are times when it is undesirable to make that particular patch or connection which is made by a normal. This is the time when the input jack will be used, and the normal will be *broken*. Breaking a normal means disconnecting that premade electrical connection in the synthesizer. To break a normal, all one must do is plug a patch cord into an input jack. When a patch cord is connected to an input jack, two things actually happen: First, the normal is broken and what was formerly connected to that input is now disconnected. Second, whatever is traveling down that patch cord is now connected to the input.

Figure 1-7: Nobody’s fool. Two dummy plugs

A great example of a normal is the headphone jack. The headphone jack is actually an output, since it puts out a signal for headphones, but it still represents a normal. Sound is normalled from the synthesizer’s internal amplifier to the synthesizer’s speakers. When a pair of headphones is plugged into the headphone jack, that normal is broken, and no sound can emerge from the speakers. The ARP 2600 has thirty-nine inputs that have something normalled to them.

Dummy Plugs

While normals are very convenient, there are times when it is desirable to break a normal without connecting anything to that particular jack. A synthesist might want to connect a module other than the one which is preconnected by the normal. One possible solution to this problem is to just plug one end of a patch cord into the jack, but the problem with this is that the other end of the cord can touch objects in the studio and create electrical noise. The cable could also pick up electromagnetic interference and add even more unwanted noise. A dummy plug is a much better solution to this problem.

A *dummy plug* (see Figure 1-7 on page six) is just a plug from a patch cord without the cord. Using a dummy plug, a normal can be broken without all of the disadvantages of plugging in one end of a patch cord. Throughout the experiments with the ARP that follow, the reader will make use of the dummy plug.

Modular Vs. Semi-Modular

As mentioned before, on a truly modular synthesizer, none of the modules are actually connected. Of course, normals actually make some connections between modules without using patch cords. So it would seem that the ARP 2600 is not actually a modular synthesizer. This is true; the ARP 2600 is not technically a modular synthesizer. It is still possible to use it as a modular synthesizer, though, and it retains all of the advantages of a modular synthesizer without some of the inconveniences. Because of these subtle differences, the ARP 2600 will be referred to as a *semi-modular synthesizer*. Basically, 'semi-modular' simply means that many of the modules have normalled connections.

Cloning in the Synthesizer World

(Or: Multiples and how to use them)

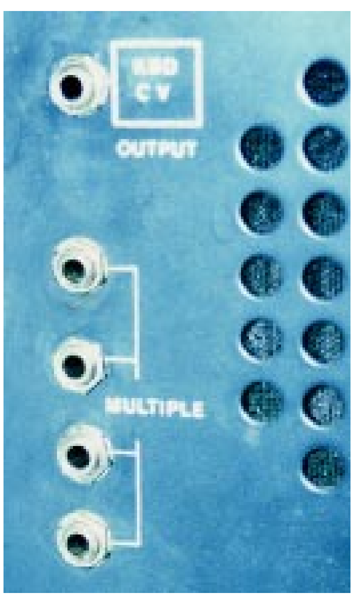


Figure 1-8: The ARP 2600's multiples

One of the first modules one will encounter on the ARP 2600 synthesizer is the multiple. It is fairly easy to understand and use, and it really adds to the flexibility of the synthesizer. The multiple, which is located in the lower left hand corner of the cabinet, simply makes extra copies of any signal. (See Figure 1-8) The multiple is made up of four jacks, which are all wired together internally. If one connects an output to any one of those jacks, three identical copies will come out the other jacks. This duplication occurs regardless of which jack one plugs into. Using the multiple, it is possible to make up to three copies of a signal. This will really come in handy later on.

Conversely, it is possible to plug three different signals into the multiple, they will be summed, and will all be output at the remaining multiple jack. While this is possible, it is not recommended. To properly mix signals together, they must be passed through a device called a mixer, which will be explored a bit more in Section 6.

Control Voltages and Voltage Control

To make a sound, different synthesizer modules are connected together using patch cords. However, the system that these modules use to control each other hasn't been explained yet. Several different systems have come and gone over time. The ARP 2600 uses one of the earliest, and most primitive. (It is one of the easiest to understand, though!) The 2600 uses a system called *voltage control* to send signals from one module to another.

In a voltage control system, modules send out a raw electrical voltage that represents a value. The greater the voltage, the higher the value it represents. This voltage is called a *control voltage*. The term *voltage control* is used to describe a system where these control voltages are used. Synthesizers do not use a lot of voltage to send these signals, so one is never in danger of getting an electrical shock from the synthesizer, as long as the cabinet is not opened, which is fairly difficult to do, anyhow.

Another way to remember these two, similar sounding terms is to remember that 'voltage control' is usually used as an adjective. It describes a synthesizer or a module of a synthesizer (e.g. the ARP 2600 is a voltage controlled synthesizer). Meanwhile, 'control voltage' is a noun. One might say that a control voltage is being produced by a certain module.

Voltage Control, Parameters & Values

Voltage control will be discussed in greater detail in the next section when it is possible to actually hear its effects. For now, students should just try to understand the basic concept. Earlier on in this section it was said that a parameter is something that can be changed, and the possible settings of that parameter are its possible values. On the ARP 2600, parameters are represented by control inputs. Values are represented by control voltages. By connecting a control voltage to an input jack, that value is assigned to whatever parameter the input jack represents. This will become clearer over time, especially when it appears again in the next section.

Keyboard Control Voltage

One device that creates control voltages is the keyboard. It was mentioned earlier in this section that the keyboard receives voltage from the cabinet through its connecting cable. However, the keyboard is also returning several signals of its own, one of which is the *keyboard control voltage*. The higher the key played, the greater the voltage the keyboard sends out. This voltage goes back to the cabinet and comes out the Keyboard CV output jack on the front panel of the cabinet. This jack is located just above the multiple and can be seen in Figure 1-8 on page7. This voltage is then used to control the pitch of the oscillators, as will be explained in the next section.

Experiments For Section One:

1. Demonstrate left to right signal flow on the ARP's cabinet. Why is the synthesizer designed this way?

2. Locate the keyboard and the cabinet of the ARP 2600.

3. Locate the cable which connects the keyboard and the cabinet.

4. Demonstrate the technique for 'zeroing' the synthesizer and demonstrate power-up procedure.

5. Locate main power switch, the light above it, and main power cord.

6. Locate an input, and notice the symbol below it indicating its normal.

7. Locate the speakers and their volume sliders.

8. Locate the headphone jack. Demonstrate what happens to the speakers when headphones are plugged into the headphone jack. What is this phenomenon called?

9. Demonstrate correct use and care of patch cords. Notice the colors and different lengths.

10. Demonstrate a dummy plug.

11. Locate the multiple on the front panel of the ARP.

12. Locate the keyboard control voltage output on the front panel.

Review Questions For Section One:

1. When was the ARP 2600 made? Is this a typical production time span for a synthesizer?

2. Why is the synthesizer designed to let signals flow from left to right? What was the primary goal of designing the ARP 2600 synthesizer?

3. Name the two main parts of the ARP 2600. Is it possible to use one part without the other? What is one purpose of the cable that connects the two parts?

4. What must be done before the synthesizer is turned on to avoid damage to other studio devices?

5. What happens to the speakers if you plug headphones into the synthesizer? How is this a little unusual?

6. What does 'zeroing the synthezier' mean? Why is it important to zero the synthesizer before using it?

7. How should patch cords be treated to protect them? Which cable generally represents which signal?

8. List the advantages and disadvantages of modular synthesizers.

9. Describe how modules are patched together.

10. What is the difference between Voltage Control and Control Voltage?

11. How does voltage control relate to parameters and values?

12. Where does the main power cable connect to the cabinet?

13. Should inputs be connected to inputs or outputs?

Terms To Know:

Audio Signal
Control Signal
Control Voltage
Dummy Plug
Fixed-architecture Synthesizer
Input
Jack
Keyboard Control Voltage
Modular
Module
Monophonic
Multiple
Normal
Output
Parameter
Patch Cable
Patch
Semi-Modular
Value
Voltage Control
Zero

SECTION 2 VCO-1

All About Oscillators

Oscillators are the fundamental part of any synthesizer. They are the module that creates the raw sound that will be shaped and molded by all the other parts of the synthesizer. Oscillators function by putting out voltage in a pattern. The faster they put out the pattern, the higher the frequency they produce. When this output voltage is amplified and connected to a speaker, a sound can sometimes be heard. Some people think that oscillators only put out voltage when a key is being played on the keyboard. This really isn't true, though. Oscillators constantly oscillate at a specified rate, even if a key isn't being played. Another word for rate is *frequency* and it is measured in *Hertz* (Hz).

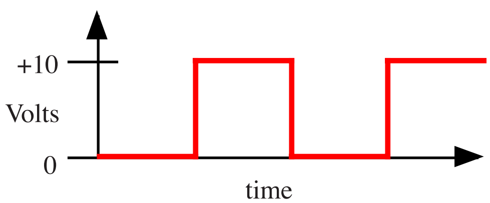


Figure 2-1: A square wave

VCO stands for *voltage controlled oscillator*. This means that this module is an oscillator and can produce audio signals and control signals. It also means that at least one of its parameters can be controlled via voltage control. This is another perfect example of the term 'voltage controlled' being used as an adjective as mentioned in Section 1.

Most oscillators are capable of producing different tone colors. This is accomplished by putting out voltage in a pattern called a *waveform*. For instance, to create a *square wave* (see Figure 2-1), the oscillator will put out no voltage for a moment, then put out ten volts for a moment. To produce a *saw wave* (see Figure 2-2), the oscillator must increase its voltage gradually to ten volts, then drop sharply back to zero volts.

Repeating a waveform very quickly (often thousands of times per second) produces an electronic signal which human ears will perceive as a tone after it is amplified and is connected to a speaker. Notice when the raw output of an oscillator is connected to a speaker that the sound is not particularly interesting to listen to. Because the sound is static and unchanging, it is rather monotonous or boring.

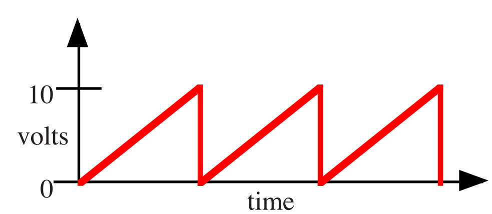


Figure 2-2: A saw wave

The Oscillator's Timbre

Oscillators have two different parameters, the first of which is timbre. *Timbre* comes from French, and is pronounced *tam-ber.* Timbre means tone color or raw sound. When timbre changes, the shape of the waveform changes. One can easily see by comparing Figures 2-1 and 2-2 above that a square wave does not look anything like a saw wave. The two will sound different as well, just the way a piano sounds different from a trumpet, even if each sounds the same note.

One selects a timbre by connecting a patch cord to one of the oscillator's two outputs. In Figure 2-3, VCO-1's two outputs jacks can be seen. The top jack constantly puts out a saw wave and the bottom jack puts out a square wave. It is very important to note that connecting a patch cord to one of the two outputs is the *only* way the timbre can be changed. For instance, if one wishes to hear a square wave, one must connect the square output of VCO-1 to the speakers. The only way to change the timbre that VCO-1 is creating is to manually connect the patch cord to the saw output. It is important to also realize that both outputs of an oscillator can be used at the same time so that both timbres can be heard simultaneously.

Figure 2-1:
VCO-1's saw and square outputs.

Although everyone will perceive timbres slightly differently, it is possible to make some generalizations about them which will guide the student in his or her studies. The saw wave has lots of harmonics, and as such has a sound that sounds buzzy. The square wave, on the other hand, has only the odd harmonics, and as such, it has a rather hollow sound. Take a moment now to listen to **CD track 01**. Several tones are played by VCO-1. First, the notes are played with a square wave produced by VCO-1. Then, the notes are played with a saw wave produced by VCO-1. Remember to listen for the raw sound or timbre of the sound, and not how quickly or slowly the sound begins or ends.

The Oscillator's Frequency

The second parameter of oscillators is frequency. Frequency is often referred to as *pitch* by musicians. While selecting a timbre is fairly simple, controlling frequency is a bit more involved. Frequency is controlled in several different ways. First, VCO-1 has a coarse frequency setting. This fader can change the oscillator's frequency over a very large range. It is possible to make the oscillator oscillate so quickly that it can't be heard at all (a *supersonic tone* - 20 kHz or higher) or so slowly that a tone can't be perceived (a *subsonic tone* - 20 Hz or lower).

When it is necessary to tune an oscillator to another source such as another oscillator, one needs better overall control than the coarse tuning slider can provide. This is the job of the fine tuning slider. (The earliest version of the ARP 2600 lacked the fine tune control on VCO-1.) See page three for more information.) The fine tuning slider increases or decreases the pitch a small amount from wherever it has been set by the coarse tuning slider. When attempting to tune an oscillator to match another source such as another oscillator, one should get the frequency close to that of the other source using the coarse slider, then tune it in perfectly using the fine tuning slider.

As the oscillator's tuning gets close to the pitch of the other source, a series of 'beats' can be heard. These beats are a sort of pulsing in the sound which occur when the waveforms of the two sources alternately cancel and reinforce each other. This results in a small change in volume which is perceived as beats. As the frequency of the two sources gets closer and closer together, the beats will gradually slow until finally, they stop, indicating that the two sources are perfectly in tune. Take a moment now to listen to **CD track 02**. Two oscillators are being tuned together. Listen for the slowing of the beats as they get closer to being in tune.

Modulation: The Key to the World of Synthesizers

The third way the oscillator's frequency is controlled is by the amount of voltage that it receives. This is why this oscillator is called a Voltage-Controlled Oscillator; its frequency can be controlled by an external voltage. The more voltage the oscillator is fed, the higher the frequency or pitch it will produce.

Things to this point have been pretty straightforward, but now comes the tricky part. On synthesizers, a technique called *modulation* is frequently used. Modulation allows one module to change the value of a parameter of another module. The easiest way to understand modulation is by looking at an example.

If Wendy is riding in a car and she is attempting to draw a straight line across a piece of paper, she could represent a module on a synthesizer. The line Wendy is trying to draw on the paper is the parameter which can be changed. When the driver drives over some big bumps in the road, Wendy's straight line is going to be changed with each bump she rides over. So, the road is changing the value of Wendy's line. Instead of a nice straight line, she might end up with one that goes all over the page. What is really happening here is that the texture of the road is modulating Wendy's drawing.

Whenever modulation occurs, there is a *carrier* and a *modulator*. The carrier is the module whose parameter is being changed (Wendy's drawing in the example above). The modulator is the module that is doing the changing (the road in the example above).

Understanding modulation is the key to understanding modular synthesis. Although modular synthesis is called "modular synthesis" because it involves different modules, it might just as well have been called "modular synthesis" because the individual modules change, or modulate each other. Understanding modulation is the key to understanding the ARP 2600. Once the concept of modulation is understood, everything else becomes much more clear, and more complex patches can be attempted.

Frequency Modulation

While it is not possible to modulate the timbre of VCO-1 from another source (remember: the *only* way to switch timbres on VCO-1 is to manually plug the patch cord into a different output), it is possible to modulate the frequency using a control voltage. (This process will be dealt with in depth in the next section.) When the frequency of an oscillator is being modulated, this technique is called *frequency modulation*. Frequency modulation is often abbreviated 'FM'.

To modulate the frequency of VCO-1, a control voltage must be connected to one of the four jacks below VCO-1. (See Figure 2-4 on page 14) These jacks are called *frequency modulation inputs* and they are labeled "FM CONTROL" on the cabinet's panel. When a control signal (like the control voltage output of the keyboard) is connected to one of these inputs, the stage is set to modulate the frequency of the oscillator. However, the ARP gives the user some options here. The observant student will notice that there is something normalled to each FM input jack. These devices will all be discussed in time. One of the most common examples of frequency modulation is a control voltage from the keyboard modulating the frequency of an oscillator.

When plugging into the FM three jacks on the right of VCO-1, the user can control the amount of control voltage that will actually get to the oscillator. When the slider or fader (the two terms are used interchangeably in this book) above a jack is all the way down, no signal will be passed to the oscillator from that jack. When the slider is all the way up, all of the incoming control signal will be allowed to modulate the oscillator. When a fader is all the way down (or all the way to the left) it is said to be *closed*. Conversely, when the slider is set all the way up (or all the way to the right) it is said to be *open*.

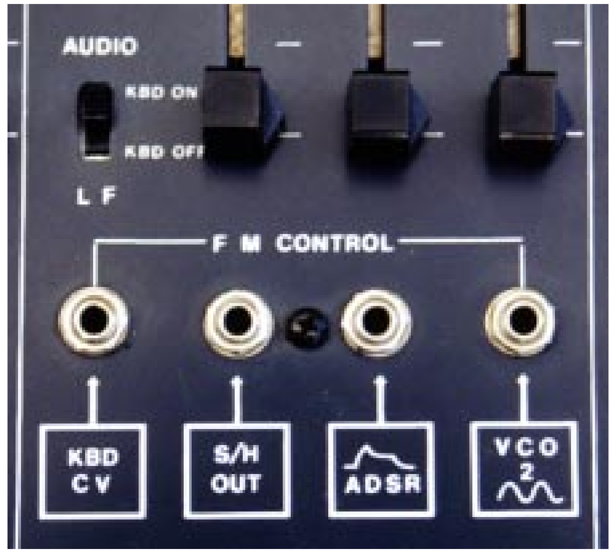


Figure 2-4: VCO-1's FM jacks

The left most FM CONTROL jack is normalled to the keyboard's control voltage. One can tell that the keyboard control voltage is normalled to this input since the words "KBD CV" appear in the white box under the input. Since it is keyboard CV that is normalled here, it is usually desirable to have all of the keyboard control voltage modulating the frequency of the oscillator. Thus, there is no fader above this input, and all of the incoming control signal will always modulate the oscillator. If a fader was present above this jack, then all of the voltage would not get to the VCO, and the keyboard would not produce chromatic half steps from one note to the next. There is some use for this technique, and it will be explored later in this section. Frequency modulation is the final way that Figure 2-5 below sums up all of the ways VCO-1's frequency can be changed.

The Keyboard and Redundant Patches

VCO-1

VCO-1's frequency is determined by:

- Coarse Tune
- Fine Tune
- Control Voltage connected to FM inputs. (This in cludes the keyboard CV normal)

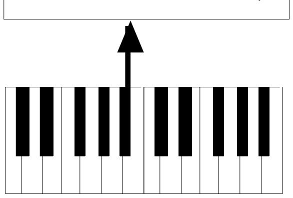


Figure 2-5

The control voltage produced by the keyboard is normalled to each oscillator. When the keyboard is played, it sends out a control voltage for each key, and the oscillator will change its frequency depending upon how much voltage it receives from the keyboard. This is a great example of voltage control discussed in Section 1. This is how the synthesizer is able to play different pitches when different notes are played on the keyboard. The keyboard's normal to the oscillator can be broken by inserting a dummy plug into the Keyboard CV jack.

Sometimes people have trouble remembering that the keyboard's control voltage is normalled to the oscillators. It is possible to patch the keyboard's control voltage output on the front panel to the keyboard control voltage FM input on VCO-1, but this is not necessary, since the keyboard control voltage is already normalled to each oscillator. Creating this patch would just be redoing what the normal has already accomplished. If a patch is created which duplicates the effect of a normal, it is called a *redundant patch*. Patching the keyboard CV output to the keyboard CV jack on VCO-1 is a perfect example of a redundant patch. This redundant patch is illustrated in Figure 2-6 on Page 15. (The heavy red line represents a patch cord.)

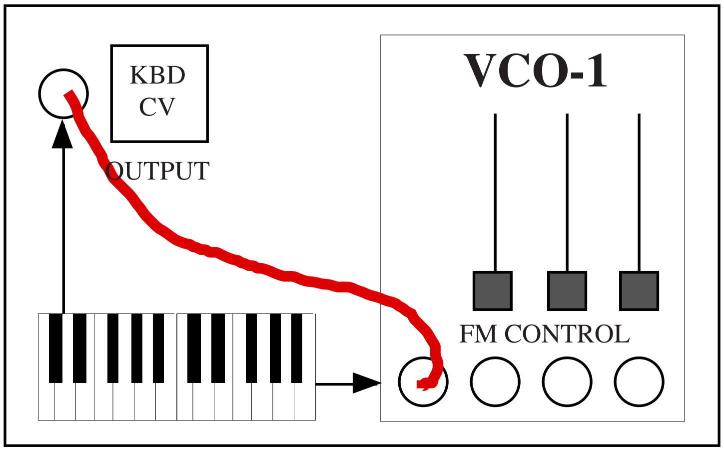


Figure 2-6: A common redundant patch

Redundant patches should be avoided for several reasons. First, they use a patch cord which could otherwise be used for some other purpose. At first, it might seem as though one would never actually use all of the available patch cords in a studio at once, but as additional synthesis techniques are discovered, experimenters will want to create ever more complex patches which will require many patch cords. Secondly, whenever more cables are used in an electronic music setup, there is a greater chance for things to go wrong. Jacks sometimes go bad, cables sometimes go bad, and cables can pick up hum from other electrical devices and even radio waves. Redundant patches make troubleshooting a patch much harder since they introduce so many variables. Normals have a fairly low failure rate, and it is much better to make use of them rather than using patch cords whenever possible.

LFO's and VCO's

VCO-1 also leads a double life as a low frequency oscillator. By moving the Audio/LF switch (upper left hand corner of Figure 2-4 on page 14) to the lowest position, VCO-1 will oscillate in the sub-audio range. *Sub-audio* means that the pitch is so low (the frequency is so slow) that humans aren't able to hear a tone. Instead, a repeating series of clicks is audible. When a VCO is in low frequency mode (LF mode for short) it is a *low frequency oscillator*, which is abbreviated *LFO*. Knowing what is now known about frequency modulation, think about how one oscillator in LFO mode could be used to FM another oscillator. (This is actually discussed in detail in the next section.)

As the frequency of the oscillator in low frequency mode is increased, the oscillator will eventually reach a point where a tone can be heard. This happens around 20 Hz, or 20 cycles per second, which is about the lowest pitch human beings can hear. Listen to **CD track 03**. One can hear the sound of an oscillator in low frequency mode and its frequency is gradually being increased so that it eventually reaches the audible range.

When a VCO is in low frequency mode, the keyboard control voltage normal is broken. This means that the keyboard CV will no longer reach the oscillator. This is desirable because an LFO is expected to oscillate steadily at one frequency and the keyboard CV would change the frequency at which the LFO was oscillating. Usually, LFO's are used to create vibrato, which will be discussed in the next section. For the rare occasions when a synthesist wants an LFO's frequency to follow the keyboard CV, the synthesist can use a patch cord to connect the keyboard CV output jack to one of the FM input jacks on the oscillator. The label to the right of the LF switch reminds the user that the keyboard will be disconnected when the switch is set to the low position. It says "KBD ON" near the audio position, and "KBD OFF" near the LF position.

Experiments for Section Two:

1. Listen to the raw sawtooth output by patching oscillator directly to an input on the mixer (your teacher can help you with this). Describe the timbre.

2. Listen to raw square output by patching oscillator directly to an input on the mixer. Describe the timbre. **CD track 01**

3. How else can the timbre the VCO is producing be changed without moving the patch cord?

4. Patch both wave outputs into the mixer. Is it possible to hear both timbres at once?

5. Control the frequency of VCO-1 by moving the INITIAL FREQUENCY slider. Try to play a simple song. Is this and effective way to control the frequency of an oscillator? Now control the pitch using the FINE TUNE slider.

6. While listening to either output of VCO-1, play the keyboard and notice that VCO-1's pitch changes. What is occurring here?

7. Create a redundant patch by connecting the keyboard's CV output to the FM input labeled "KYBD CV". Notice that this has the same effect as the normal in experiment #6.

8. Insert a dummy plug into the FM control input labeled "KYBD CV" on VCO-1 and play the keyboard. What is occurring now?

9. Put VCO-1 in LF mode using the Audio/LF switch. Lower the initial frequency slider so that it is open only about 1/4. What sounds do you hear when listening to the saw wave? When listening to the square wave?

10. After conducting experiment #9, try playing the keyboard. What happens and why?

11. Starting in the sub-audio range (LF mode), slowly bring VCO-1's frequency into the audio range. Notice the point at which one can first hear a tone. **CD track 03**

12. Zero VCO-1, then practice frequency modulation on VCO-1 by using a dummy plug to break the keyboard CV normal on the left most FM input. Then patch the Keyboard CV output to one of the other FM inputs and fully open the slider above that jack. How is this not as desirable as using the normal from the keyboard CV? What happens if you set the slider in a position other than fully open or fully closed? What is noticeable about the pitch even when the slider is fully open?

13. How many notes can VCO-1 produce at one time?

Review Questions for Section Two:

1. Describe an oscillator's function and role in the synthesizer, and tell how it works.

2. Explain when oscillators oscillate, and tell why the raw sound they produce is rather boring.

3. Name the oscillator's two parameters, and the way both can be changed from the oscillator itself.

4. Tell how many different waveform outputs can be used at once from VCO-1.

5. List the three things that determine a VCO's pitch.

6. Describe the timbre of saw and square waves.

7. Describe the best way to tune VCO-1 to another source.

8. Describe how the keyboard is connected to VCO-1.

9. List three reasons why it is highly desirable to avoid redundant patches.

10. Give an example of modulation. Be sure to include the words *carrier* and *modulator* in your explanation. Tell why it is so important to understand modulation.

11. Give an example of frequency modulation. Also, describe what connections must be made to frequency modulate an oscillator.

12. Speculate how an LFO will be useful later. Tell how to put a VCO into LF mode. Describe what happens to the keyboard CV normal to VCO-1 when VCO-1 is put into LF mode.

13. How many FM control inputs does VCO-1 have?

Terms To Know:

Carrier
Closed
Fader
Frequency
Frequency Modulation
Frequency Modulation Inputs
Hertz
LF Mode
LFO
Modulation
Modulator
Open
Oscillator
Pitch
Redundant Patch
Sawtooth Wave
Slider
Square Wave
Subsonic
Supersonic
Timbre
VCO
Waveform

SECTION 3

VCO-2

A First Look

In Section 2, a basic voltage controlled oscillator (VCO-1) was discussed in detail. In this section, VCO-2 will be examined. Although VCO-2 is more complex than VCO-1, all of the things that were said about VCO-1 apply to VCO-2. VCO-2 can be thought of as the 'super oscillator' on the ARP 2600 since it is the most flexible, and thus the most powerful oscillator of the three found on the 2600. The first thing one notices when looking at VCO-2 for the first time is the additional waveforms that VCO-2 can produce. In addition to the saw wave, VCO-2 can produce sine, triangle, and pulse waves.

Sine Waves

Sine waves (see Figure 3-1) have a very pure sound, because they have only one harmonic, and that is the fundamental. **CD track 07** It is also important to note that while the square and saw waves that VCO-1 produced ranged between 0 Volts and +10 Volts, the sine wave VCO-2 produces ranges between -5 Volts and +5 Volts. Overall, it still has the same amplitude span as the waveforms VCO-1 produced, but because it dips into negative voltage, it is more flexible. When the sine wave output is used to modulate another source, it can actually send a negative value to that carrier, rather than just a positive value. This technique will be explored in depth in a later section.

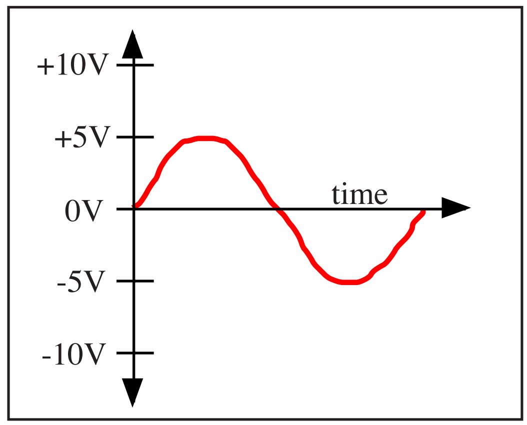


Figure 3-1: A sine wave

Triangle Waves

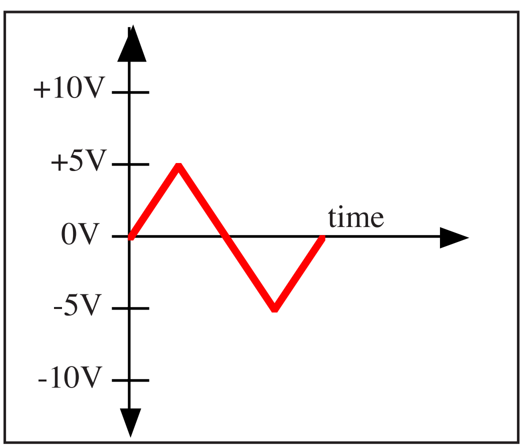


Figure 3-2: A triangle wave

Triangle waves (see Figure 3-2) sound a little buzzy, but smoother than a saw wave. Although they look very similar to a saw wave, they are actually most like square waves harmonically. **CD track 07** Just like the square wave, they have only the odd numbered harmonics. At first, this doesn't seem to make any sense. If the harmonic content of a waveform determines its shape, and the triangle and square waves both have the same harmonics, shouldn't they look alike and sound alike? Actually, it is possible for two different waveforms to have the same harmonics present, but present in different amounts. In the triangle wave, some of the upper harmonics are present in higher amounts than in the square wave. This gives it a buzzy sound which is different from the square wave. It is also important to note that like the sine wave, the triangle wave ranges from -5 Volts to +5 Volts, meaning that it can send a negative value to a carrier when it is being used as a modulator.

Not Just Another Pretty Phase

When two waveforms begin and end together, they are said to be *in phase* with each other. In Figure 3-3, one can see that the two sine waves shown are in phase with each other. (The two overlapping sine waves are represented by the thick red line.) When two identical waveforms are in phase with each other, an interesting thing happens. They are summed, and the result that humans hear is louder than the amplitude of either of the waveforms alone. This phenomenon is called *reinforcement*. In Figure 3-3, the taller black line shows the result of the two red sine waves being summed.

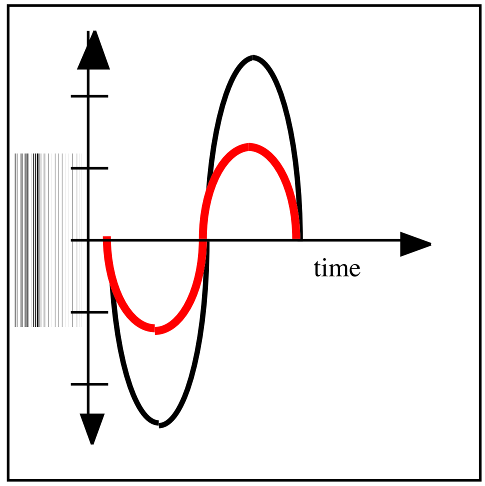


Figure 3-3: Reinforcement

A waveform's cycle can be measured in 360 degrees like a circle. In Figure 3-4, some of the important phase angle marks are indicated. When one waveform starts halfway though the cycle of another waveform, the two are said to be 180 degrees out of phase, since half of 360 is 180. Similarly, if one waveform starts a quarter way through another waveform's cycle, the two would be 90 degrees out of phase. This measurement is called the *phase angle*. When two waveforms are 180 degrees out of phase with each other, and the waveforms are identical, a strange phenomenon occurs. Because the crest of one wave occurs at the same time as the trough of another, the two cancel each other out, and the listener perceives a reduction in volume. This phenomenon is called *cancellation*. In the real world, it is very difficult to get two waveforms to cancel each other out completely, even under carefully controlled circumstances.

Phase Relationships

The triangle and sine outputs of VCO-2 have an important quality: They are 180 degrees out of phase with each other. In Figure 3-4, one can see that the triangle wave (green) reaches its highest amplitude at 90 degrees, just when the sine wave is at its lowest amplitude. However, if the triangle and sine waves are added together, they will not cancel each other out. Remember that the two have different waveforms, and a different harmonic content. The sine wave has only the fundamental, and as such, it will only cause cancellation of the triangle wave's fundamental.

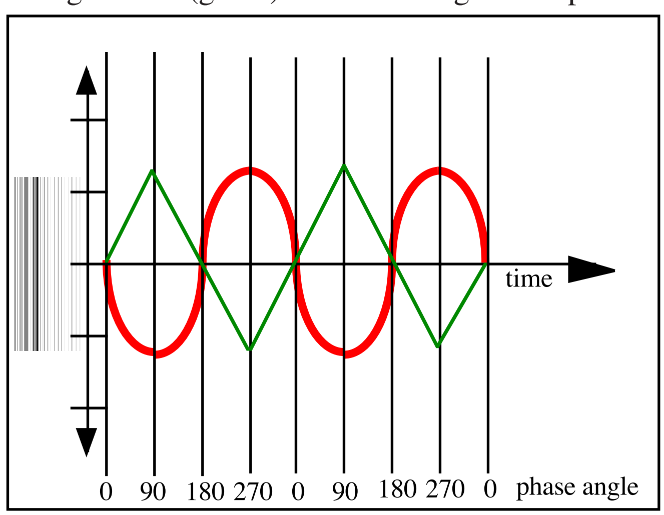


Figure 3-4: The sine and triangle waves are 180 degrees out of phase

The phase relationship between the triangle and the sine outputs is the most obvious and easiest to observe. However, when dealing with simple waveforms like the ones the ARP 2600 can produce, it is possible to say that two waveforms are 180 degrees out of phase when the highest point of one waveform occurs at the lowest point of another. The phase relationships of the other outputs can be easily summed up: outputs which are perpendicular to each other (i.e. above below, or to one side) are 180 degrees out of phase. Those diagonal to each other are in phase. Figure 3-5 illustrates this relationship. Red arrows represent out of phase relationships while blue and green arrows represent in phase relationships.

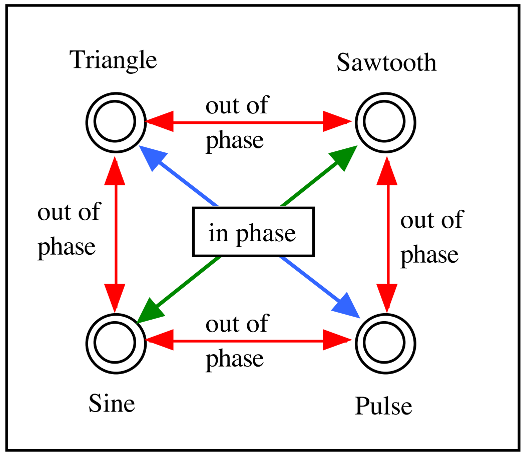


Figure 3-5: The phase relationships of VCO-2's outputs

Pulse Waves

VCO-2 doesn't really have a square wave output, but has a pulse wave output instead. This is a very important feature since a pulse wave is like a square wave, but is much more flexible. Pulse waves, like a square wave, are partially "on" and partially "off." In terms of voltage, a pulse wave consists of ten volts for a moment, then zero volts for a moment. For a square wave, these two 'moments' must be equal. In a pulse wave, they *can* be equal, but do not have to be equal. Thus, a square wave is really just a kind of pulse wave.

Pulse Width

One important aspect of a pulse wave is that it is another parameter to change. That parameter is the *pulse width*. Of course, in Section 2, the reader learned that there are only two parameters of an oscillator one can change: frequency and timbre. One might think of pulse width as a sort of subheading under the 'timbre' parameter heading.

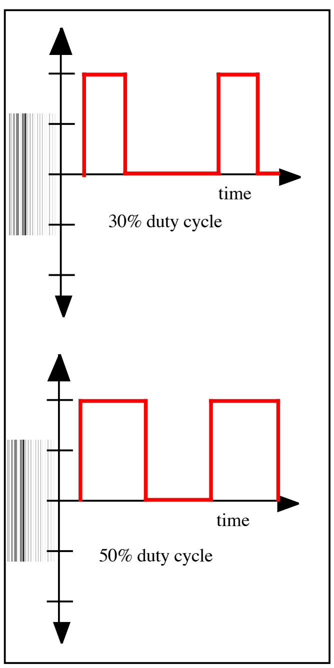


Figure 3-6: Pulse waves with different duty cucles

Pulse width is the name of the parameter that controls how much of the wave is 'on' and how much of the wave is 'off.' Another name for pulse width is *duty cycle*. The duty cycle of a pulse wave is expressed as a percentage. The percentage tells how much of the duty cycle is 'on.' For instance, a pulse wave that has a pulse width of 20% would be a pulse wave which is 'on' 20% on the time, and 'off' 80% of the time. Figure 3-6 illustrates a pulse width of 30%, and a pulse width of 50% (a square wave). Making changes in the pulse width changes the waveshape and thus the timbre. This is why it is difficult to describe the pulse wave's timbre: the timbre changes significantly when the pulse width is changed. The timbre can range from hollow when the duty cycle is near 50% to nasal when it is nearer 0% or 100%.

On the 2600, changing VCO-2's pulse width is simple. The pulse width is changed by moving the PULSE WIDTH slider, right under the FINE TUNE slider. This control can be seen, along with VCO-2's outputs and initial frequency controls in Figure 3-7 on page 21. When the pulse width is set to 50%, the VCO will produce a square wave (if it is properly calibrated).

A NOTE ABOUT CALIBRATION: Throughout this text, the reader will see phrases such as "if it is properly calibrated." One may begin to wonder how calibration is achieved. Once again, different versions of the 2600 have slightly different methods of calibration. In most cases, there are small holes in the front of the cabinet, each of which has a trim pot inside. One such hole can be seen in Figure 3-7. By inserting a standard screwdriver, one can turn the trim pot and calibrate each module. On the first models produced (the "Blue Meanies,") each calibration hole was actually labeled with the function it calibrated. While this was highly user-friendly, it seems that it was also highly inviting, and many people managed to get their 2600's out of whack before they were really familiar with them. By the time ARP made the grey faced 2600's, the labels had disappeared, and some of the less frequently used calibrations (such as the high frequency tracking on the VCO's) were moved inside the cabinet.

In any case, the calibration procedure was not one that ARP kept secret. In fact, the procedure can be found near the back of the 2600 manual. It is fairly easy to perform, and does not require any equipment other than a screwdriver and a good ear. It helps to have something to reference when calibrating the range of the oscillators, such as a tuning fork, but it is really not necessary. However, the 2600 rarely requires calibration, and when it does, it is truly as the manual states, "calibration without tears."

Notice that the label on the pulse width control only extends from 10% to 90%. This is because when the pulse width becomes too narrow, no sound can be heard. Another important fact to note is that the human ear cannot perceive the difference between a 10% duty cycle and a 90% duty cycle. Similarly, one can't tell a 40% duty cycle from a 60%, a 32% from a 68% and so on. This is because a 10% duty cycle is just the same as an inverted 90% duty cycle, and without a reference, the ear can't hear when a waveform has been inverted. Because of this phenomenon, some synthesizer manufacturers produce synthesizers whose pulse width is only variable from 10% to 50%, since sweeping the pulse width from 10% to 90% sounds the same as sweeping the pulse width from 10% to 50% and back to 10% again. Because of this phenomenon, some manufacturers decide to leave the pulse width control without any labels other than "pulse width."

Figure 3-7: VCO-2's controls

MORE ABOUT FREQUENCY MODULATION

In Section 2, the concept of frequency modulation was introduced. Frequency modulation occurs when a control voltage is used to change the frequency of an oscillator. The only frequency modulation that could be observed when the concept was introduced was the keyboard modulating the frequency of VCO-1. Now, however it is time to apply some of the knowledge collected in the first two sections and attempt some more complicated frequency modulation.

Creating the First FM Patch

VCO-2, like VCO-1, can be made into an LFO by switching the LF switch. In the example which follows, VCO-2 is used in LF mode to modulate the frequency of VCO-1. First, it is important to determine the relationship of the two VCO's. Recall that when modulation occurs, there must be a carrier and a modulator. VCO-1 is the carrier, so it will produce sound so that the frequency modulation can be heard. VCO-1 must be patched to the mixer so that it can be heard. Its frequency must be set in roughly the middle of its range so that it is easy to hear the effect of the modulator on it. If the frequency is set too high or too low, FM'ing it may push it into the supersonic range or subsonic range respectively, and then the results of the experiment cannot be heard. VCO-1 will be modulated by VCO-2, so VCO-2 is the modulator, and it will produce the control voltage which will be modulating VCO-1.

One of VCO-2's outputs must be connected to VCO-1 so that VCO-2 can modulate VCO-1. After switching VCO-2 to LF mode, the saw output can be connected or patched to one of the FM inputs on VCO-1. Recall from Section 2 that these FM inputs accept incoming control voltages and the more voltage they receive, the higher the pitch the oscillator produces. Finally, the fader above the FM input on VCO-1 must be raised to allow some of the incoming control voltage to modulate VCO-1's frequency. Figure 3-8 depicts this patch.

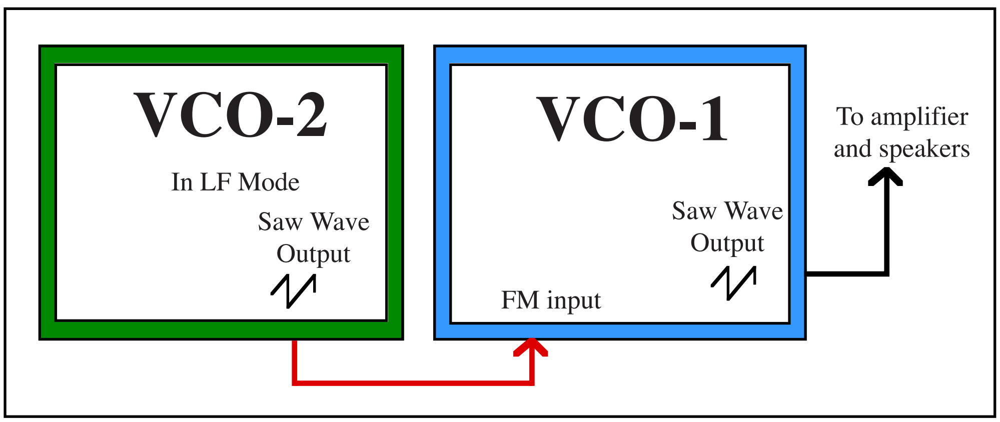


Figure 3-8: A basic FM patch

Being able to look at a block diagram of a sound or reading about a patch and being able to predict how it will sound is a very important skill; important enough that it is the subject of Section 14. It is important to begin to build a mental list of different simple patches and how each one works. It will soon become clear that even the most complex patches are really just a collection of very simple patches all used at the same time. For instance, four or five frequency modulations similar to the one shown above might occur simultaneously. That would produce some complex results indeed!

Dissecting the Basic FM Patch

In Figure 3-8, VCO-2 is putting out a saw wave slowly because it is in LF mode. This saw wave is connected to an FM input on VCO-1 using a patch cord. VCO-1's frequency will change with the incoming saw wave from VCO-2. As the saw wave from VCO-2 slowly goes up, the frequency of VCO-1 will go up as well. When VCO-2's saw wave suddenly drops low to begin again, the frequency of VCO-1 will also drop suddenly to begin rising again. This patch can be heard by listening to **CD track 04**. Listen as VCO-2's waveform is changed. Is it possible to determine which waveform is being used just by listening to the results?

The Black Art of Patch Diagramming

Patch diagramming and analysis will be explored in depth in Section 14, but there are some important aspects of patch diagrams which must be explored now. The following conventions are used in this book: First, control signals are drawn with red lines while audio signals are drawn with black lines, just as red patch cords are used for control signals and black cords for audio signals. Second, carriers are indicated with a blue background while modulators are indicated with a green background. Finally, audio signals exit each module to one side while control signals exit and enter modules from the bottom or top. The only notable exception to this last convention is the audio connection of signals to the amplifier and speakers. To conserve space, the connection is always drawn below the words indicating the speakers and amplifier. Each of these conventions can be seen in Figure 3-8 on page 22.

This is also the first really clear example of a patch which uses control and audio signals. VCO-2's output going to VCO-1 is a control signal. It is a control signal because it is never heard, nor is it intended to be heard. One can tell that it is a control signal because it is a sub-audio signal (VCO-2 is in LF mode) and the signal emerges from the bottom of the module in Figure 3-8, and it is drawn with a red line. (See the sidebar "The Black Art of Patch Diagramming" for more information.)

Don't think that VCO-2's output passes through VCO-1. This is strictly incorrect. The signal from VCO-2 just changes or modulates a parameter of VCO-1, but doesn't pass through it. However, VCO-1's output is an audio signal. The intent is that VCO-1's output will be heard, which is part of the definition of an audio signal.

The Parameters of a Basic FM Patch

There are several different parameters available to the reader in this patch. The first parameter is the starting pitch of VCO-1. Of course, the pitch or frequency is going to be changed through frequency modulation, but one must set a starting frequency using the INITIAL FREQUENCY slider to establish the range in which VCO-1 will move.

The second parameter is the waveform that VCO-1 is producing. This determines the timbre of the final sound. Either the saw wave or the square wave could be chosen. While it is important to understand that this parameter can be changed, it is not particularly important to the discussion at hand.

The third parameter one can change is the amount of control voltage VCO-1 is receiving from VCO-2. This parameter is referred to as *depth* or *modulation depth*. There are sliders above each of the FM inputs on VCO-1, and these determine how much control signal is allowed through to modulate VCO-1. The higher the slider is set, the greater the depth of modulation, and the more control voltage will modulate VCO-1. Essentially, the slider 'turns down' the amount of control signal getting to VCO-1 when the slider is moved down. The term used to describe this 'turning down' is *attenuation*.

The fourth parameter in this patch is the waveform that VCO-2 is producing. VCO-2 can produce four different waveforms, but the pulse width can be changed on the pulse wave, so that allows hundreds, if not thousands more possibilities. Selecting a different waveform for VCO-2 means that the frequency of VCO-1 will be modulated in a different pattern. While choosing a sine wave would produce a smooth rising and falling of tone, a triangle wave would produce a slightly more pronounced change in direction of the rising and falling of frequency. A saw wave would cause the frequency to ramp slowly up, then drop off. A pulse wave would cause the frequency to alternate between two specific pitches.

A possible fifth parameter for the patch given above is pulse width. Of course, this only applies when the pulse wave is being used for frequency modulation. The pulse width would determine how much of the control signal would be high and how much would be low, and thus, how much of VCO-1's frequency would be high and how much of VCO-1's frequency would be low.

The sixth parameter in this patch is VCO-2's frequency. This parameter also has a special name; it is referred to as *rate* or *modulation rate*. The faster VCO-2 oscillates, the greater the rate, and the faster VCO-1's frequency will change. As the frequency is gradually increased, the frequency begins to change so fast that human ears can no longer perceive the change. This happens when VCO-2 is oscillating so fast that it is actually in the audio range (i.e. 20 Hz or higher), and the depth is great enough (see the third parameter). When both the depth and the rate are great enough, a strange phenomenon occurs.

Sidebands

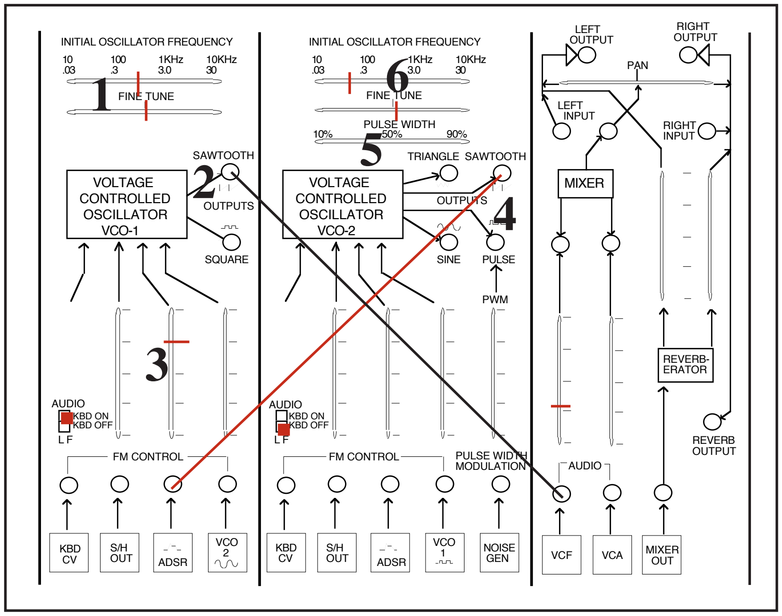
When both the depth and the rate are great enough, harmonics are added to the timbre VCO-1 is producing. These harmonics are called *sidebands*. An example of sidebands can be heard by listening **CD track 05**. Sidebands suddenly decrease if the carrier and modulator's frequencies are multiples of each other. For instance, if VCO-1 is tuned to 440 Hz and VCO-2 is tuned to 220 Hz, there will be fewer sidebands since 220 x 2 = 440. To summarize: Sidebands are produced when frequency modulation is occurring, the modulator is in the audio range, the modulation is deep enough, and the carrier and modulator's frequencies are not multiples of each other.

Some people spend hours mathematically calculating exactly what frequencies will be added as the sidebands form. (This can be calculated through a complex series of formulas which take into account the carrier's waveform, the modulator's waveform, exactly how loud each of the waveform's hundreds of harmonics are and the overall frequency of the carrier and modulator.) They can create charts and graphs showing the frequency content of the new sound, but this is really rather impractical for the purposes of this book. This analytical branch of electronic music will be ignored for the time being as it falls too far outside the practical application of music technology. It is merely important to understand that it is possible to mathematically predict which sidebands will occur.

Summing Up the Patch

To summarize, in the basic frequency modulation patch, there are five (sometimes six) parameters. Each of these is labeled in Figure 3-9, which depicts the front panel of the instrument rather than a patch diagram.

1. VCO-1's starting frequency
2. VCO-1's timbre
3. Modulation depth
4. VCO-2's waveform
5. Pulse width of VCO-2 (if using the pulse wave for modulating VCO-1)
6. Modulation rate



Frequency modulation is used all the time in music technology. The most common application of FM is vibrato. *Vibrato* is a very slight rising and falling of the pitch of a sound. It is this slight rising and falling that makes a sound more human and eliminates the sterile qualities of the raw waveforms.

Pulse Width Modulation

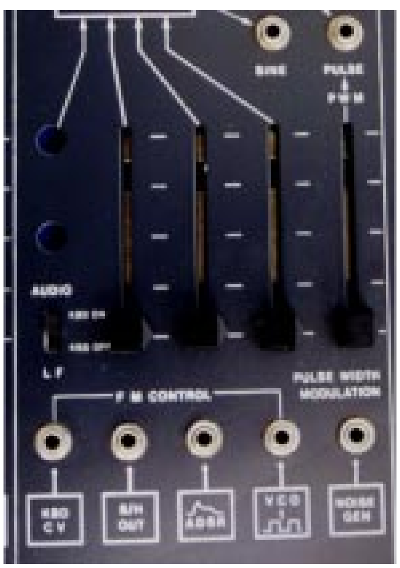


Figure 3-10: VCO-2's FM and PWM inputs

In addition to modulating frequency with a control voltage, it is also possible to use a control voltage to modulate the duty cycle or pulse width of the pulse wave. Because the pulse width is being modulated, this kind of modulation is called *Pulse Width Modulation* or PWM. Pulse width modulation has a distinctive sound which must be heard to be understood. Listen to **CD track 06** which demonstrates the patch shown in Figure 3-11 on page 26. PWM provides a wonderful way to create timbres which change continuously in time, holding a listener's attention.

Sometimes, if a synthesizer is out of calibration, the oscillator will stop sounding momentarily because the pulse width becomes so narrow or so wide that the oscillator no longer produces an audible sound. To prevent this from happening, the PULSE WIDTH slider should be set far enough left so that the pulse width never gets narrow enough to stop sounding.

Notice in Figure 3-10 that VCO-2 has an extra input located just to the right of the FM inputs. This is the PWM input. It has its own slider to attenuate the incoming control voltage. A control voltage input here will cause the pulse width to change continuously over time. When a parameter changes continuously over its entire range of values, it is said that the parameter is *swept*. A pulse width sweep can be created by setting VCO-1 to LF mode, and connecting its saw wave output to VCO-2's PWM input. Be sure that VCO-2 is in audio mode, and that VCO-1 is not oscillating too quickly. This patch is diagrammed in Figure 3-11 on page 26.

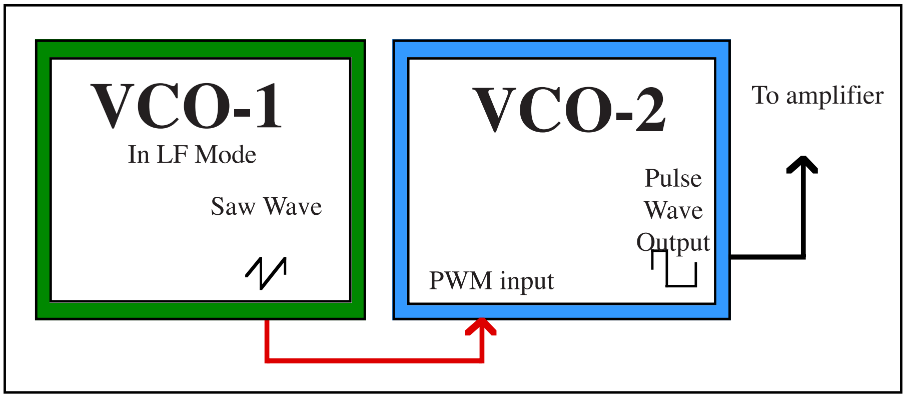


Figure 3-11: A basic PWM patch

In this patch, VCO-1 is the modulator, and VCO-2 is the carrier. The saw signal going from VCO-1 to VCO-2 is a control signal, while the pulse wave output of VCO-2 is the carrier signal. Pulse width modulation has a very distinctive sound and while it is not widely used, it is certainly an important and useful synthesis technique.

Dissecting the Basic PWM Patch

The basic PWM patch has only four parameters. First, the frequency of VCO-1 can be set. This parameter is once again referred to as *rate* in this patch. Secondly, VCO-1's waveform can be changed. However, the saw wave is really a more common choice than the square wave, since the saw wave will produce a continuous sweep. Third, the incoming control voltage can be attenuated by the fader above the PWM input. This parameter will once again be referred to as *depth*. When too much depth is used in PWM, it becomes hard to perceive a tone. When the depth and rate become fast enough, once again, sidebands are produced. These sidebands are very similar in sound to those created by FM. Finally, VCO-2's frequency can be set. VCO-2's timbre is not a parameter in this patch, since it is not something that can be changed. If one wants to hear PWM, one must use a pulse wave.

Other Possibilities with Two Oscillators

Another useful technique available to the experimenter at this time is using both VCO-1 and VCO-2 in audio mode, and listening to different waveforms from each of them at the same time. Patching the square output of VCO-1 and the saw output of VCO-2 to the mixer might sound pretty unpleasant at first, but this is merely because the oscillators haven't been tuned together.

In Section 2, the technique for tuning an oscillator to another source was discussed. Now is the time to apply that knowledge. It is important to first decide which oscillator will be the one that will be tuned and which oscillator will be the one that the other is tuned to. For demonstration purposes, VCO-1 will be the standard by which to tune VCO-2 and that VCO-2 will be the oscillator that will actually be tuned.

To begin, VCO-1 must be set at a comfortable frequency. Next, VCO-2's frequency must be roughly matched to VCO-1's. This is accomplished with the INITIAL FREQUENCY slider. It takes a good ear and a steady hand to accomplish this. Once VCO-2 is close to VCO-1, the FINE TUNE slider is used to bring them exactly in tune. As the frequencies get close to matching, 'beats' can be heard in the sound. These beats can be a big clue to how close the frequencies are to matching each other. The faster the beats, the farther the frequencies are from matching. If the FINE TUNE slider is being moved, and the beats are speeding up, then the frequencies are getting farther apart, and the slider is being moved the wrong direction. Eventually, if the slider is moved in the correct direction and the coarse tuning slider was set close to begin with, the beats will slow to an almost imperceptible speed. The oscillators are then in tune.

Phat Tuning

It is not always desirable to put the oscillators perfectly in tune, however. Many pop songs today use oscillators which are intentionally out of tune as this yields a wider, warmer sound that analog synthesizers like the ARP 2600 are known for. The term for this detuned sound is taken from urban hip-hop culture and is mutated from an English word. The word is *phat* and is pronounced just like 'fat.' This sound is the reason that old analog synthesizers like the ARP 2600 are highly sought after and are prized by synthesists everywhere. **CD track 11**

Herein is the big advantage of using two oscillators to create a square and saw wave as opposed to using one oscillator to create the same waveforms. Indeed, the more oscillators used to make a sound, the warmer, thicker, and more interesting the sound will be.

Another great possibility available to the student at this time is tuning the oscillators in different intervals. Great sounds can be achieved by tuning the oscillators in octaves, perfect fifths, and perfect fourths. Of course, there are lots of other interesting possibilities, and the general rule is that if a patch sounds good, then it is acceptable. Tuning in different intervals, combined with the phat tuning discussed earlier will yield some great new sounds.

Experiments for Section Three:

1. Listen to raw saw, square, sine and pulse outputs by patching each directly to the mixer. Describe the differences in timbre. Notice which waves seem softer. **CD track 07**

2. Experiment with the INITIAL FREQUENCY slider, and FINE TUNE sliders.

3. While listening to the pulse wave, trying varying the PULSE WIDTH using the pulse width slider. Try to duplicate the sound of a square wave perfectly. **CD track 08**

4. Create the FM patch given in this section, and experiment with each of the five or six parameters. Do not stop until all of the possibilities have been exhausted. Have patience and try all of the possibilities. This is not a quick procedure. It is important to constantly ask the following questions:
    1. How does this set of parameter values sound?
    2. Why does it sound like this?
    3. What exactly is happening between these two oscillators?

   It is also important to take notes on the results so that other patches that will be illustrated later can be better understood. **CD track 09**

5. Using a square wave from VCO-2 in LFO mode, produce leaps of different intervals in VCO-1 using FM. How does it sound to use octaves or tritones? Can the sound of a French Ambulance siren be generated? **CD track 10**

6. Using a ramp wave or the triangle wave, create a ‘siren’ type of sound. **CD track 10**

7. Slowly increase the frequency of the LFO until sidebands appear. Why do the sidebands seem to disappear at certain frequencies? **CD track 05**

8. Create the pulse width modulation patch given in this section, and once again, experiment with all of the available parameters. Note the different sounds that can be created. **CD track 06**

9. Patch the square output of VCO-1 to the mixer and the saw output of VCO-2 to the mixer. Practice tuning them together using the tuning technique given in this section. Also try tuning in different intervals.

10. Detune VCO-1 and VCO-2 to create the phat tuning. **CD track 11**

11. Combine the triangle and sine outputs of VCO-2 in the mixer. While listening to the triangle output, slowly raise the volume of the sine output. Is it possible to hear the fundamental being cancelled?

Review Questions for Section Three:

1. Compare and contrast VCO-1 and VCO-2.

2. Describe the timbre of the sine, saw and square waves, and tell why it is difficult to describe the timbre of the pulse wave.

3. Explain how duty cycles are expressed numerically, and why human ears can't tell the difference between 10% and 90%. Describe the relationship of square and pulse waves.

4. Name the six parameters of a basic frequency modulation patch and describe what effect changing the value of each has on the resulting sound. Draw a simple diagram of the patch.

5. Why is vibrato musically useful?

6. Tell which outputs are in phase with each other, and which outputs are out of phase with each other.

7. How did calibration change as the ARP 2600 evolved?

8. Draw a simple diagram of a PWM patch.

9. Describe the technique used for tuning two oscillators together and define and describe the phat tuning technique.

10. State why it is desirable to use two different oscillators to produce two waveforms.

11. Tell what is meant by 'tuning in intervals' and state why this is a highly useful technique.

12. Discuss the voltage ranges of the triangle, sine, and pulse waves produced by VCO-2.

Terms To Know:

Attenuation
Cancellation
Duty Cycle
Modulation Depth
Modulation Rate
Phase
Phase Angle
Phat Tuning
Pulse Wave
Pulse Width
Pulse Width Modulation
PWM
Reinforcement
Sidebands
Sine Wave
Sweep
Triangle Wave
Vibrato

SECTION 4

VCO-3

Hello Again!

VCO-3 is very similar to VCO-1. In Figure 4-1, it is easy to see how similar the two really are. It can only produce two different waveforms: saw and pulse waves. However, it lacks the PWM input that VCO-2 is blessed with, so PWM is not possible. It is somewhat more flexible than VCO-1, though, because VCO-1 can only produce a square wave. VCO-3 can create a pulse wave, which becomes a square wave when the pulse width is set to 50% and can also produce many other timbres as well. Other than the ability to vary the duty cycle on the pulse wave, VCO-3 and VCO-1 are identical.

Figure 4-1: VCO-3

It may seem as though a third oscillator is just excess baggage. Indeed some manufacturers who were making synthesizers around the time the ARP 2600 was being produced thought so also. They started making synthesizers with only one or two VCO's. VCO's are fairly expensive modules, so this seemed like a good way to lower the cost of products. There are several reasons that this is not good. First, when lots of oscillators are tuned together and layered, they make an incredibly wonderful sound. Second, there are more possibilities for modulation when there are more potential carriers and modulators. Third, since instruments made around this time period are subject to breakdowns, it never hurts to have one more oscillator in case one goes bad.

Double Modulation

There are many interesting ways to make unusual sounds on the ARP 2600, but double modulation is one of the most unique and predictable. When VCO-1 was discussed, it was said that it had four FM inputs. In Section 2, when an FM patch was created, only one of those inputs was actually used. In *double modula-*

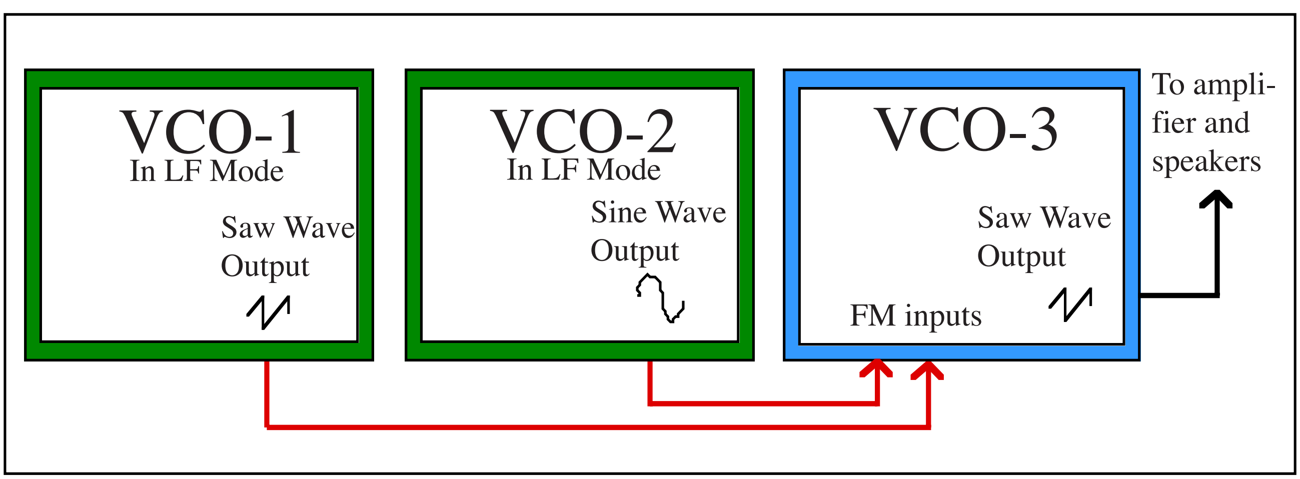


Figure 4-2: A basic double modulation patch

tion, two of the FM inputs on an oscillator are used simultaneously. In Figure 4-2, VCO-3 is the carrier while both VCO-1 and VCO-2 modulate its frequency. This more complex patch can be thought of as two basic FM patches which happen to share a carrier. Remember that just about any complex patch can be broken down into two or more simple patches.

In this patch, the outputs of VCO-1 and VCO-2 are control signals, and VCO-3's output is an audio signal. VCO-1 and VCO-2 are modulators and VCO-3 is the sole carrier. It would be foolish to stop to see all of the possible parameters in this patch because they would be very similar to the parameters in the first frequency modulation patch from Section 3. (Figure 3-8) Several different examples of double modulation using three oscillators can be heard on **CD track 12**.

Generally, two different waveforms are used on the modulators in double modulation, since this produces more variety than using two of the same waveforms. There are times when the use of two of the same waveforms might be interesting and musically useful, though. One may discover how interesting it is to create sidebands with one modulator while using the other in the sub-audio range. Keep in mind as more modules are discovered that double and cross modulation does not necessarily require that all of the carriers and modulators are oscillators. Other modules could fall into these places. Experimentation is the key to mastering and understanding all of the concepts in this book. The more experiments the reader performs, the better each synthesis technique will be understood.

What kinds of sounds can be created with double modulation? Double modulation usually yields sounds that are not intended for playing melodies. (I.e. it is sometimes hard to perceive a pitch.) There are, however, some very musical possibilities. One interesting possibility is to use one of the modulators as a source for vibrato while the other modulator performs more drastic frequency modulation. One can also create sidebands while causing the fundamental frequency to move up and down.

Cross Modulation

It seems that there is some disagreement in the electronic music community as to exactly what cross modulation is. While some define it as any sort of FM where sidebands are being produced, other

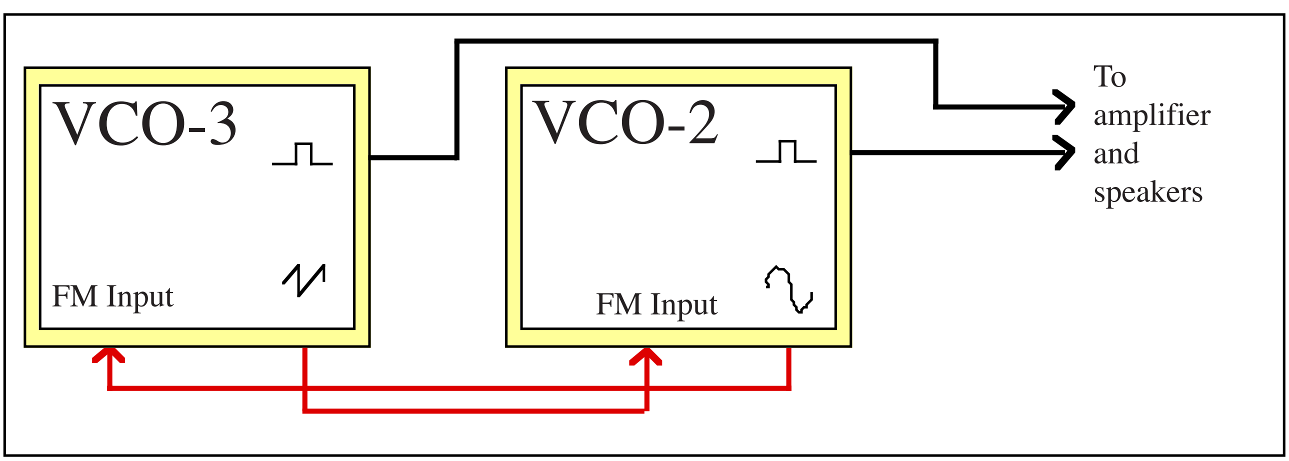


Figure 4-3: Cross modulation in action

sources specify certain connections of oscillators. For the purposes of this book, *cross modulation* will be defined as a patch in which two oscillators are both in the audio range, both are frequency modulating each other, and both can be heard. In Figure 4-3 on page 31, VCO-2 is modulating VCO-3. Meanwhile, VCO-3 is modulating VCO-2. Notice that both oscillators are in the audio range, and both are being heard.

Figure 4-3 shows both VCO's in light yellow because each functions as both a carrier and a modulator simultaneously. It would be incorrect to diagram them in either blue or green. While the results of double modulation are fairly predictable, the results of cross modulation are much less so. It is possible to produce clangorous, metallic sounds, which do not seem tuned. Take a moment to listen to **CD track 13**. Several different examples of cross modulation created with two VCO's can be heard.

Cross modulation creates an interesting dilemma. What happens if the pulse output of VCO-3 is used to modulate VCO-2, but one wishes to use the pulse output to send to the mixer as VCO-3's timbre? It is possible to first send VCO-3's pulse output to the multiple, where the signal will be duplicated and can then be routed to both VCO-2's FM input and the mixer. In this way, a given output on an oscillator can be used both for the purpose of modulating the other oscillator and generating an audio signal.

Series Modulation

Series modulation (also referred to as 'modulation in series') is like a combination of double and cross modulation. One oscillator modulates the next oscillator, which in turn modulates the final oscillator. Although there are three oscillators in this patch, there are two carriers and two modulators.

In Figure 4-4, VCO-1 is modulating VCO-2 which in turn modulates VCO-3. VCO-1 is the first modulator, modulating the frequency of VCO-2, which is the carrier. However, VCO-2 is also a modulator as it is modulating the frequency of VCO-3. It's important to understand that an oscillator can be a carrier and a modulator at the same time. However, it is not necessary to have three oscillators to have series modulation. This will become clear how this can be as more modules are discovered. **CD track 14**

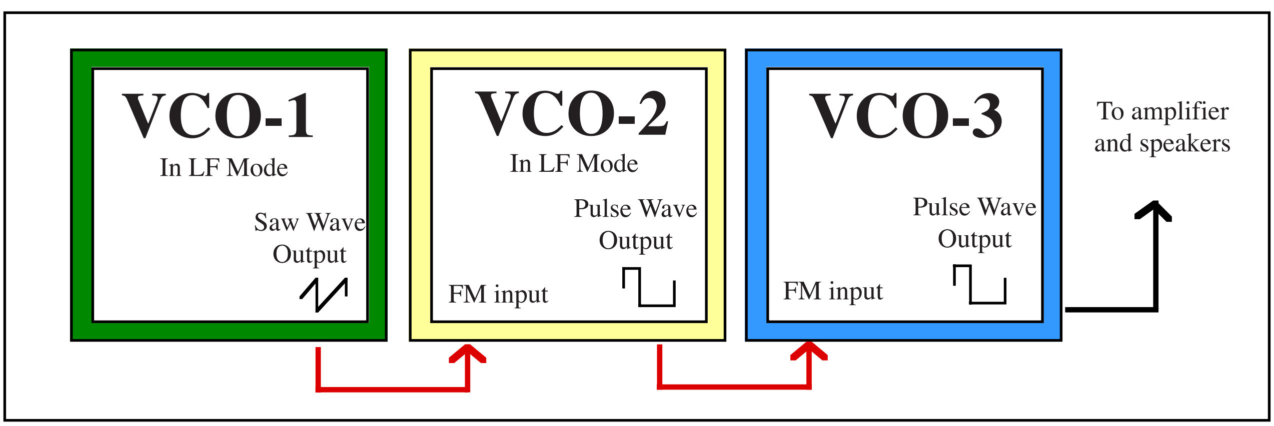


Figure 4-4: A basic series modulation patch

THE MASTER-SUBMASTER RELATIONSHIP

The second important concept that series modulation exposes the reader to is a *master-submaster relationship*. In this arrangement, the master selects from a broad range of values, while the submaster fine tunes this selection. A perfect example of the master-submaster relationship are the coarse and fine tuning controls on the VCOs, as described on page 12. The initial frequency slider or coarse tuning slider is the master, and the fine tune slider is the submaster. The master sets the coarse range of possible values, and the submaster chooses the specific value from the range provided by the master. There are several more master-submaster relationships between different controls on the ARP which will be pointed out as they come along. In Figure 4-4, VCO-1 is the master, and VCO-2 is the submaster.

OTHER THINGS TO DO WITH THREE OSCILLATORS

The phat concept can be taken to the next level now that three oscillators are available. It is possible to use one oscillator as the ‘in tune’ oscillator, then tune another oscillator slightly higher and one other oscillator slightly lower for the ultimate phat sound.

Another very popular application which requires three oscillators is to tune the oscillators in various triads and then play short melodic riffs. This is the ultimate extension of the ‘tuning in intervals’ concept that is possible on the ARP. Just imagine the kinds of sounds one could make with more oscillators! Some modern synthesizers have as many as 128 oscillators!

EXPERIMENTS FOR SECTION FOUR:

1. Create the double modulation patch shown in Figure 4-2 and experiment with each of the seven or eight parameters. Do not stop until all of the possibilities have been exhausted. Have patience and try all of the possibilities. This is not a quick procedure. It is important to constantly ask the following questions:
    1. How does this set of parameter values sound?
    2. Why does it sound like this?
    3. What exactly is happening between these two oscillators?

    It is also important to take notes on your findings so that you can understand other patches that will be illustrated later. **CD track 12**

2. Repeat #1, creating the cross modulation patch shown in Figure 4-3. **CD track 13**

3. Repeat #1, but this time, create the series modulation patch shown in Figure 4-4. Be sure to observe the master-submaster relationship which exists between VCO-1 and VCO-2. **CD track 14**

4. Practice tuning VCO-1 and VCO-3 together. **CD track 12**

5. Demonstrate phat sound by combining all three oscillators slightly detuned. **CD track 11**

6. Practice tuning in intervals. If more than two oscillators are needed at once, your teacher can help to mix the sounds together. Tune in a major chord, octaves and fifths, and three different octaves. What other cool combinations can be created? **CD track 15**

7. Use cross modulation to create a wavy sound which gradually increases and decreases in pitch.

8. Use double modulation to create a sound that has sidebands and jumps up and down in octaves.

9. Notice any modules normalled to VCO-3 which have been studied so far. To which modules is VCO-3 normalled?

Review Questions for Section Four:

1. Compare and contrast VCO-1, VCO-2, and VCO-3.

2. Do all synthesizers have 3 oscillators? Why would manufacturers make synthesizers with fewer oscillators? What are some of the advantages to having three oscillators?

3. Draw a simple diagram showing a basic double modulation patch. If you wish, you can use colored pencils to indicate the relationships of the different oscillators

4. Draw a simple block diagram showing a basic cross modulation patch.

5. Draw a simple block diagram showing a basic series modulation patch.

6. Discuss the main differences between cross modulation, double modulation, and series modulation and tell what kinds of sounds each one is good for producing.

7. Explain the master-submaster relationship and give two examples.

8. Explain when an oscillator can be a carrier and a modulator at the same time.

9. Give at least two examples of tuning tricks that can be done only with three oscillators.

Terms To Know:

Cross Modulation
Double Modulation
Master-Submaster Relationship
Series Modulation

SECTION 5 NOISE GENERATOR

A NOISY PARTNER

So far, three different sound producing modules have been explored. VCO-1, VCO-2, and VCO-3 all had several things in common. Each could produce a variety of timbres by producing different waveforms and each could be frequency modulated from an external source. The *noise generator* is very different from all of these modules, but is still a highly useful tool.

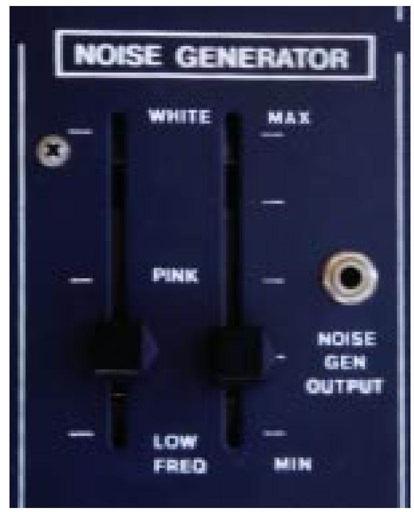


Figure 5-1: The Noise Generator

The most obvious thing one can see when first looking at the noise generator is what it lacks. It has only an output, and no inputs whatsoever. Because it has no inputs, it is impossible to modulate any parameters of the noise generator. It does have one output, which can be patched to the mixer so that it may be heard. Unfortunately, this is not of much use, as sustained noise is only slightly less interesting to listen to than the raw output of an oscillator. **CD track 18**

THE NOISE GENERATOR'S PARAMETERS

It is clear just by looking at Figure 5-1 that the noise generator has two different parameters: The first parameter is level or amplitude. The noise generator has a dedicated slider (the right one) to determine its output volume. (Level, volume, and amplitude all mean the same thing.)

The second parameter is the noise generator's *harmonic content*. Understanding this parameter requires some understanding of what the noise generator does. The noise generator produces noise. This much is obvious, but what is noise really? *Noise* can be defined as any unpitched sound. That is, a listener cannot hear any single pitch or tone when listening to the sound since it has so many frequencies present in such high amounts. The waves studied so far have had a very organized series of harmonics, and the harmonics were all present in relatively small amounts. In noise, there are millions of harmonics, all at a relatively high level. One example of noise is the static sound a television makes when it is unplugged from the antenna or cable.

WHAT COLOR IS THAT NOISE?

Not all noise is the same however, and electronic musicians describe the frequency content of noise using colors. *White noise* is a random amount of all frequencies simultaneously. Humans tend to hear some frequencies better than others, specifically those most used in human speech. White noise, which has all frequencies will tend to sound like it is predominantly 1-2 kHz, because that is the range humans hear best. It is important to think of the noise generator and white noise as a random source. There will be times in the future when a random source is needed to carry out other tasks, and this is an area where the noise generator is perfect.

Pink noise is much like white noise, but has fewer high frequencies, and thus has a duller sound. The frequencies are removed with a device called a *filter*. Filters will be discussed in great detail in Section 6. For now, it is only important to understand that the filter on the noise generator can attenuate the amount of high frequencies the noise generator actually produces. When the noise generator's frequency slider is set at LOW FREQ, only low frequencies are produced (low frequency noise), and the noise generator makes a rather low, rumbling sound not unlike a waterfall heard from a distance.

Musicians sometimes refer to other colors, such as blue noise, red noise, and even green noise, but these are all for rather specialized purposes, and it is unnecessary to understand them completely at this time. It is just important to realize that they exist.

The Noise Generator in FM

The noise generator has hundreds of potential uses, but only a few of them are available to the student at this time, since the only modules that have been studied are the oscillators. It is possible, however, to use noise to modulate the frequency of an oscillator as the patch in Figure 5-2 shows.

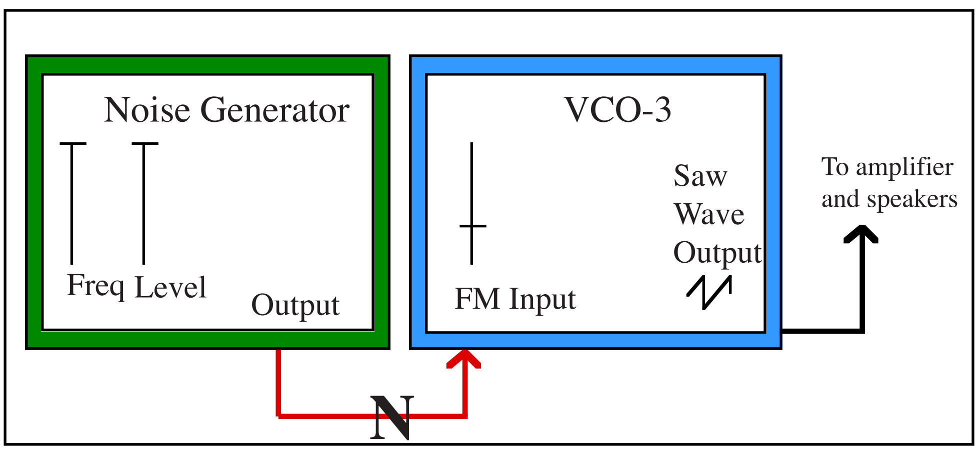


Figure 5-2: The noise generator in FM

The noise generator's output has been patched to an FM input on VCO-3. (VCO-3 is a good choice as the noise generator is already normalled to one of its FM inputs. Notice that a large 'N' has been drawn on the line representing the patch cord to indicate that this patch is a normal.) VCO-3 is the carrier in this patch, while the noise generator is the modulator. (Remember that the noise generator can't be modulated, so this is the only possibility.) The volume of the noise generator has been set all the way up, and it is set to produce white noise. Generally, the noise generator's level will be set full open, and then its level will be attenuated at the inputs of the various carriers where it arrives. This is the best strategy, since the noise generator's level can always be attenuated at any given input, but it can't be amplified. The volume control on the output of this module is something of a fluke on the 2600. The noise generator is the only module on the 2600 which allows the user to set an output volume.

The depth of modulation in Figure 5-2 is not set too high as too much frequency modulation by the noise generator will easily cause a complete loss of tone. This sort of frequency modulation produces rather interesting sounds which tend to be more organic and grungy and lack the sterile, boring quality of pure waveforms. Some FM patches created with the noise generator can be heard on **CD track 16.**

There are five (possibly six) parameters to this patch, just as in the FM patch explained in Section 3. First, the noise generator's level can be set (this is effectively the same as parameter 3 in which the modulation depth is set). Secondly, the noise generator's frequency attenuation slider can be set which determines how many upper harmonics will be added to VCO-3's output. Third, the level of modulation can be set using the attenuation slider above the FM input (master-submaster relationship with parameter 1.) Fourth, the frequency of VCO-3 can be set. Fifth, the timbre of VCO-3 can be set. Sixth, when VCO-3's pulse output is used, its pulse width can be set.

The Noise Generator in PWM

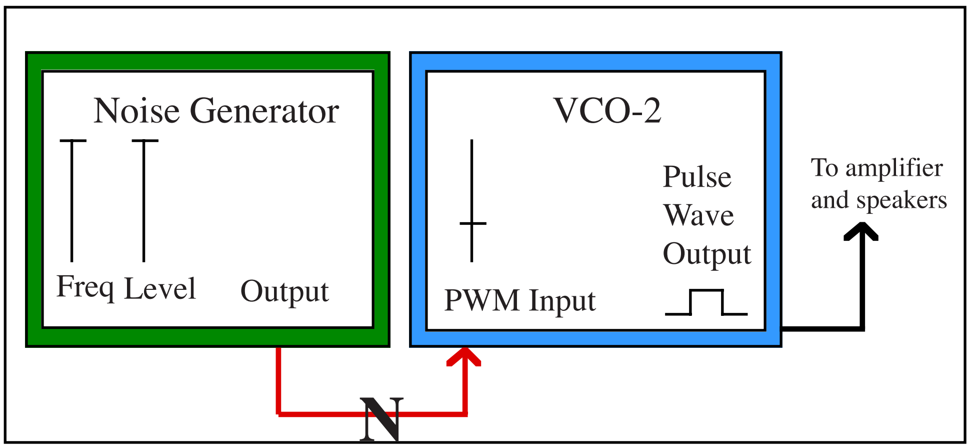


Figure 5-3: The noise generator in PWM

Another excellent possibility for use of the noise generator is using it to change the pulse width or duty cycle of the pulse wave output on VCO-2. The output of the noise generator could be manually patched to the PWM input on VCO-2, but closer inspection of this input will reveal something important: the noise generator is normalled to the PWM input. This is important to note, since normals represent the way things are most commonly patched together. Thus, patching the noise generator's output to the PWM input must be a fairly common application.

Once again, the noise generator is the modulator in this patch, and VCO-2 is the carrier. This patch has four possible parameters: First, the noise generator's level can be set. This will effectively do the same thing as the attenuation at the PWM input. Second, the noise generator's frequency slider can be set to tailor the noise. This will determine the level of the higher harmonics in the pulse waveform that VCO-2 will produce. Third, the depth of modulation can be attenuated by the slider above the PWM input. This effectively does the same thing as the first parameter. Similarly to the FM patch illustrated earlier, the noise generator's level has been set to full open, and its level has been attenuated at VCO-2's PWM input. Finally, VCO-2's frequency can be set. (One might think that VCO-2's timbre could be a fifth parameter in this patch, but this is not so; the pulse wave must be used if one is attempting pulse width modulation.) Notice that parameter 1 (the noise generator's level) and parameter 3 (modulation depth) have a master-submaster relationship. The parameters of the patch illustrated in 5-3 are summarized in Figure 5-4.

1. Noise generator's level
2. Noise generator's frequency
3. Depth of PWM modulation (effectively same as #1)
4. VCO-2's frequency

Figure 5-4: Parameters of a PWM patch using the noise generator

Note again that too much modulation depth yields a sound that is so noisy, it cannot really be of much use. This is a highly distinctive sound, and is especially useful for producing growling bass sounds, percussion sounds, and distorted types of sounds. Examples of this patch can be heard on **CD track 17**.

More Possibilities

Sounds which are pulse width modulated by the noise generator may not prove to be the most interesting or musically useful sounds, but it is important to understand this technique and add it to a mental inventory of simple patches which can be drawn on in the future. When combined with some of the techniques explained later in this book, this patch becomes much more spectacular.

It is important to understand that while the noise generator has no inputs, its output can function as either an audio signal (when patched to the mixer, for instance) or as a control signal (when patched to an FM input or the PWM input for instance). When other modules are discovered later which allow the user to shape the sounds produced by the noise generator, it will become more useful as a sound producing module, whereas this section focused mainly on its abilities as a control signal.

Experiments for Section Five:

1. Listen to the raw output of the noise generator by patching it into the mixer. **CD track 18**

2. Move the output level control and note the change in output.

3. Adjust the frequency slider and observe the difference in sound. **CD track 18**

4. Use the noise generator to control pulse width on VCO-2. (Redundant patch) As before, all possible parameters should be set in all possible combinations. This sort of experimentation can be tedious, but is highly rewarding in terms of sound produced. Take careful notes on which settings produce which sounds. **CD track 17**

5. While conducting #4, try adjusting the noise generator's frequency slider.

6. While conducting #4, try varying the amount of PWM. **CD track 17**

7. Try using the noise generator for FM purposes. When might this sound be useful? **CD track 16**

8. While conducting #7, try adjusting the noise generator's frequency slider.

9. In general, what sorts of sounds benefit most from the addition (or use) of noise?

10. Create a patch which sounds like a grungy bass guitar.

11. Create a sound that sounds like a distorted guitar.

12. Create a percussion-like sound.

13. What other sounds can be created using the noise generator?

Review Questions for Section Five:

1. Which parameters of the noise generator can be modulated?

2. Can the noise generator be a carrier? Can it be a modulator?

3. Name the parameters of the noise generator.

4. State why white noise sounds like it does. Include in your answer a discussion of harmonic content and level as well as human hearing.

5. State why the noise generator will be of great importance later on.

6. Draw a picture showing a patch in which the noise generator frequency modulates VCO-2. Number and list the parameters of this patch.

7. Draw a picture showing a patch in which the noise generator is modulating the pulse width of VCO-2. Number and list the parameters of this patch.

8. What sorts of sounds generally benefit most from the addition (or use) of noise?

9. State which parameters in a PWM and FM patch involving VCO-2 and the noise generator have a master-submaster relationship.

10. To which inputs on VCO-1, 2, and 3 is the noise generator normalled?

11. Give an example of the noise generator's output being used as a control signal and being used as an audio signal.

Terms To Know:

Filter
Low Frequency Noise
Noise
Noise Generator
Pink Noise
White Noise

SECTION 6

VCF

A Brief Introduction to Subtractive Synthesis

The filter is unlike the rest of the modules mentioned thus far in that its primary function is not to produce sound, although it can be used for this purpose. Rather, it is used to change and shape other sounds being made by the other modules of the synthesizer.

The *Voltage Controlled Filter* (VCF) is perhaps the single most important part of a synthesizer, because it determines the overall sound of the synthesizer and opens the doors to a new method of synthesis: subtractive synthesis. *Subtractive synthesis* is a method of synthesis which starts with one or more harmonically rich waveforms from which some harmonics are then removed. An oscillator usually produces this harmonically rich waveform, (usually a saw or pulse wave, but sometimes a triangle wave) but it is the filter that performs the task of eliminating some harmonics. The filter has several audio inputs, so the outputs of several oscillators can be connected to just one filter using patch cords.

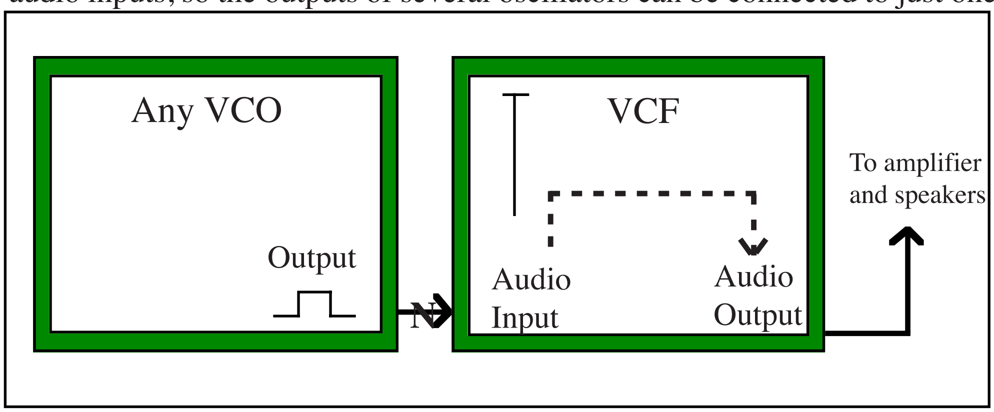


Figure 6-1: A basic subtractive synthesis patch

The oscillator's output is filtered, and then comes out of the filter's output jack. A block diagram of a subtractive synthesis patch can be seen in Figure 6-1. Notice that every VCO has an output normalled to the filter.

It is important to understand that filtering is not like frequency modulation. The two can easily be confused since the FM jacks on the VCO's are aligned perfectly with the audio inputs on the filter. Both also have attenuation sliders above them, which only adds to the confusion. However, in FM, an incoming control signal modulates the frequency of the oscillator, but does not pass through the oscillator. On the filter, signals coming into the audio inputs are actually modified by the filter and then passed through to the speakers or another module as shown by the dotted line in Figure 6-1. Notice also that both modules are outlined with green, indicating that there is no modulator in this relationship. Notice also that the signal flowing from the VCO to the VCF is an audio signal, not a control signal.

Basic Principles of Filtering

Filtering, by definition, means to remove certain elements from others. A filter on a synthesizer is a device which removes some harmonics. As an example, if one fills a glass with water and then places marbles into the glass, we have a perfect analogy of a harmonically rich waveform. Say the marbles are the harmonics that one wishes to remove, and the water represents the harmonics one wishes to preserve. When the contents of the glass are poured through a handkerchief, the marbles are not permitted through. They are filtered, while the water is allowed to pass through, mostly unchanged.

This analogy is a good one because as the water passes through the handkerchief, some of it is absorbed by the handkerchief, so not all of the water is allowed to pass through. This holds true when passing sounds through a filter. While the unwanted harmonics can be removed, some of the other harmonics end up being filtered out as well.

When learning about filters, it is important to understand that there are many types of filters. Early synthesizers usually only offered the user one type of filter, but sometimes had as many as four. Modern synthesizers, however, may offer as many as 36 different types of filters! The study of some of these more esoteric filters is, for the moment, beyond the scope of this first volume, but they will be taken up at a future time. It is also worth noting at this time that most older synthesizers had only one filter on them, but modern synthesizers sometimes have as many as 128 independent filters!

What Do Filters Do?

The exact electronic workings of a filter are unimportant at this time. It is very important, however, to understand the function of a filter. As stated above, a filter removes unwanted harmonics, along with a few of the wanted harmonics. However, one cannot pick and choose exactly which harmonics one wants to remove. To take the next step in understanding, one must first understand that there are four basic types of filters, each of which performs a specific job.

The four basic types of filters found on older synthesizers are lowpass, bandpass, band reject and highpass. Of the four, the *lowpass filter* is by far the most common. The ARP 2600 has one lowpass filter, but has no other filters. (The noise generator has a lowpass filter on it, but it is dedicated to the noise generator's output, and cannot be used for general purpose filtering.) Each type of filter is capable of filtering different ranges of harmonics. Some filters have the circuitry for some or all of these different types of filters. When a filter can operate in more than one mode (for example highpass and lowpass) it is said to be a *multimode filter*. Some synthesizers have filters that can actually perform all of these types of filtering simultaneously!

Lowpass Filters

Lowpass filters filter out high harmonics. At first, 'lowpass' may seem like a strange name for a filter that attenuates high harmonics, but it does make some sense. Filters are named by the information they allow to pass through rather than by the information they remove. So, a lowpass filter will allow all information below a certain frequency to pass through, while a highpass filter will allow all information above a certain frequency to pass through.

Although filters have several parameters, the most important is the *cutoff frequency*. Cutoff frequency is abbreviated *Fc*. Cutoff frequency is the frequency at which the filter will begin to attenuate the volume of harmonics. This attenuation is actually how a filter filters out harmonics. On the ARP 2600, Fc is set using two controls. There is an INITIAL FILTER FREQUENCY slider, and a FINE TUNE slider, which function much like the corresponding sliders on the VCO's. Instead of determining the frequency of an oscillator, however, here they determine the Fc. This is yet another example of a master-submaster relationship. These controls can be seen in Figure 6-2 on page 44.

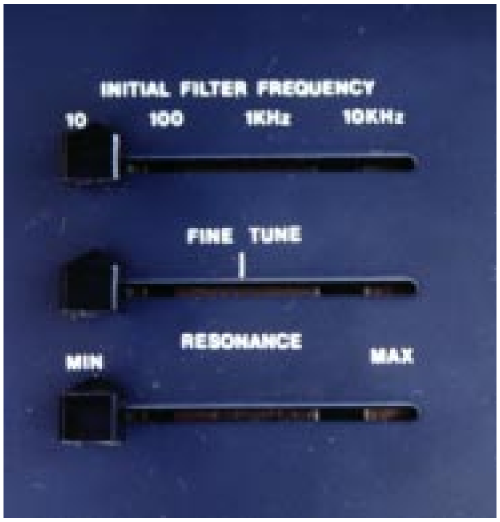


Figure 6-2: The filter's controls

As the Fc is raised, the filter is 'opened' and more harmonics are allowed to pass through. The more the filter is opened, the brighter the sound. As the cutoff frequency is lowered, the filter is said to 'close.' One might think that the filter would completely block all harmonics above or below the Fc, (depending upon the type of filter) but this isn't how filters really work. Remember that the Fc is the frequency at which the filter *begins* to attenuate harmonics.

In the Figure 6-3, the lines on the graph represent a harmonically rich waveform. The vertical black lines show the harmonics of a made-up waveform. Their height represents the volume of each harmonic. The area shaded gray represents all of the possible harmonics which could pass through unfiltered. If the sound is unfiltered, then all harmonics of the waveform are allowed to pass through the filter unchanged, no matter how high or low they are. Remember that a single sawtooth wave has high harmonics which may fall just about anywhere along this graph, depending upon the fundamental.

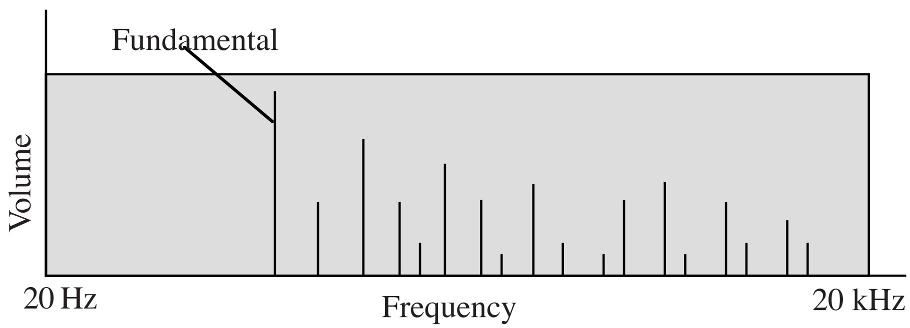


Figure 6-3: The harmonic content of a waveform

Now, in an ideal world, a low-pass filter would entirely eliminate all harmonics above the Fc. This is not how filters work, however. If the signal shown above is put through a lowpass filter, the volume will begin to *gradually* decrease once the frequency of the harmonics get higher than the Fc. It is important to note that if the fundamental frequency is too much higher than the Fc, no sound will be heard at all. This is the first thing a synthesist should check when troubleshooting a patch which uses the filter. If the Fc is completely closed, no sound will get through the filter at all.

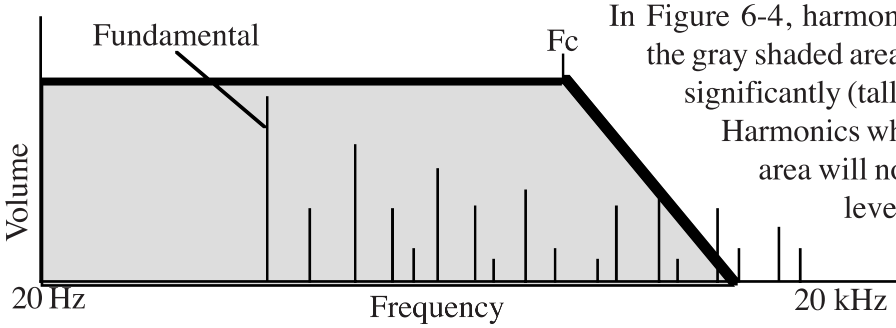


Figure 6-4: The effect of a lowpass filter

In Figure 6-4, harmonics which fall partially outside the gray shaded area will have their volume reduced significantly (taller lines represent more volume). Harmonics which fall entirely outside the gray area will not be heard at all as their volume level will be so greatly reduced. So, a saw wave would sound less buzzy than normal, since it is the high frequencies found in a saw wave that give it its buzzy sound. As the Fc is lowered, more and more harmonics will be removed and a saw wave will begin to sound more and more smooth until finally, it will sound almost exactly like a sine wave. This is because as the lowpass filter's Fc is lowered, more and more harmonics are removed. **CD track 19** The difference between a sine and a saw wave is that the sine wave has no harmonics other than the fundamental, while the saw wave has lots of harmonics. This brings up an interesting point: What happens if a sine wave is put through a lowpass filter?

The Ugly Truth Revealed

Assuming that the sine wave being fed into the filter is pure and truly has no overtones, there should not be any change in sound at all until the Fc is moved so low that the cutoff slope is over the fundamental tone the sine wave is producing. Then the sine wave will gradually decrease in volume as the Fc is moved lower and lower until it cannot be heard at all. However, connecting VCO-2's sine output to the filter and changing the Fc results in a surprising occurrence: One can hear upper harmonics being attenuated as the Fc is swept lower. This is because none of the waveforms that the 2600's VCOs produce are perfect, and when the shape of the waveforms change, their harmonic content changes slightly as well. It is very difficult to produce a true sine wave using technology that was available at the time the 2600 was built, so manufacturers came as close as they could while staying within budget.

How does a lowpass filter sound in general (when used with a signal other than a sine wave)? As the Fc is moved lower, the sound becomes duller as harmonics are attenuated and finally eliminated. In this respect, a filter is like the tone controls on a stereo or boom box. As the treble is decreased, the sound becomes duller. This sound can be heard on **CD track 19**

Grab Your Poles; Let's Hit the Slopes!

As different synthesizer companies started working on different filter designs, they changed something about the filter. The rate at which a filter attenuates frequencies is called the *cutoff slope*. Most synthesizers use either a -24 dB per octave slope or a -12 dB per octave slope (sometimes written -24 dB/8va and -12 dB/8va respectively). Decibels are a measure of volume, which means that for every octave higher the sound is, a filter with a -24 dB/8va slope would attenuate the sound 24 decibels. This is a steeper cutoff slope than a filter with a -12 dB/8va slope.

Sometimes, filters are referred to by their poles. A *pole* is a measure of attenuation, or how much the filter can reduce the volume over a given frequency range. A pole is -6 dB/8va of attenuation. Thus, filters which employ the -24 dB/8va slope are called 4-pole filters while filters which employ the -12 dB/8va slope are called 2-pole filters. While this is certainly something which differentiates different filters, cutoff slope will not be considered to be a parameter at this time, as it is not possible to change the cutoff slope on the filter on the ARP. On a few synthesizers, it is actually possible to change the cutoff slope. Some filters even offer 1-pole filters for extremely subtle and gentle cutoff slopes.

It is also possible to chain filters together, and their effect is cumulative. One can just add the cutoff slope amounts together to find the cutoff slope of the combined filters. For instance, if two -12 dB/8va or 2-pole filters are chained together (the output of the first is fed to the input of the second) they will have the same sound as a single -24 dB/8va or 4-pole filter.

Of Patents and Infringement

It was the Moog Music company that pioneered the highly desirable 2-pole filter, and the gentle cutoff slope gave their synthesizers a trademark sound. The ARP company also originally designed a 2-pole filter for their synthesizers. This filter sounds remarkably like the Moog filter, because the two designs are really very close to each other. Moog Music felt that ARP's design was too similar to theirs, and threatened to sue ARP. ARP realized that they had indeed infringed on Moog's patent, and hastily changed their filter design to a 4-pole filter. However, many of the early ARP 2600's had been made with what was essentially a Moog filter (part 4012). This makes older ARP 2600 cabinets highly sought after among synthesists. The easiest way to tell what kind of filter a given 2600 has in it is to open it up and look. 4012 filters are sealed in epoxy, and have the number 4012 stamped on their backside. The 4012 filter appeared on all of the blue and gray meanies, and some of the gray faced 2600's. All of the black and orange models and some of the gray-faced models have the redesigned 4-pole filter.

Resonance

The second parameter of filters is *resonance*. Resonance is often referred to as *Q*, but not all companies have the same name for resonance. The Moog Music company is a notable example. They used the term *emphasis* instead. Yet other companies have used the term *regeneration*. Resonance controls the amount of feedback in the filter. To say this another way, if part of the filter's output is fed back to the input of the filter, this would be a kind of feedback.

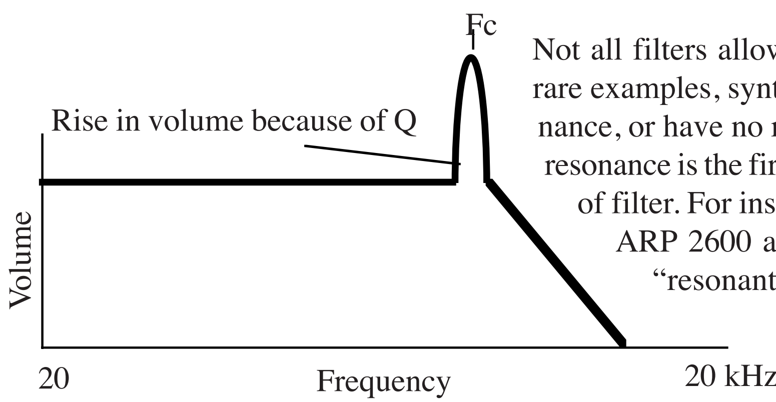


Figure 6-4: Resonance in the response curve of a VCF

Not all filters allow control of resonance. Except for a few rare examples, synthesizers either allow full control of resonance, or have no resonance at all. When describing a filter, resonance is the first quality mentioned, followed by the type of filter. For instance, one might describe the filter on the ARP 2600 as a "resonant lowpass 2-pole filter" or a "resonant lowpass 4-pole filter" depending upon when the cabinet was built.

Of course, a lowpass filter doesn't sharply cut off all frequencies above the Fc. Instead, it gradually reduces the volume as the frequency of the harmonics gets higher. To get filters closer to the ideal vertical cutoff slope, resonance can be added. Resonance is a small rise in volume of the frequencies at the Fc. Figure 6-4 shows resonance in a lowpass filter cutoff slope. It helps to accentuate the frequencies about to be cut off and results in a familiar whistling sound. **CD track 20**

Self Oscillation and Feedback

Resonance has a very distinct sound that is instantly recognizable. In large enough quantities, it has a whistling sound to it. Some filters which have a resonance control have a special property. They can be made to *self-oscillate*. To explain this strange phenomenon, it is first important to understand what was said earlier about resonance. Resonance is simply controlled feedback from the output of the filter back to the input. One might think of feedback as the horribly obnoxious sound that is made when a microphone is pointed at an amplified speaker it is connected to. A loud, piercing sound is emitted, usually followed by everyone in the room clapping their hands over their ears.

Resonance is this same sort of feedback, but in very carefully controlled amounts. If enough signal is fed back through the filter, however, it can begin to produce the obnoxious sort of feedback. The volume of the filter can be carefully controlled so that it isn't as unpleasant as when an amplified microphone is pointed at a speaker. Instead, one can hear the actual timbre that feedback produces: a sine wave. The frequency of this sine wave can be controlled by the Fc. The frequency which is boosted by the resonance (the highest point occurs at the Fc) is the frequency at which the filter will oscillate. In this way, the VCF can be a VCO since the Fc is controlled by the keyboard CV. **CD track 21**

Using the filter as an oscillator is clever, but not particularly useful for several reasons. First, the ARP gives the user three great, full-featured oscillators to use, and the filter has an important role in providing subtractive synthesis. When it is being used as an oscillator, it cannot be effectively used as a filter. Secondly, analog oscillators are notorious for drifting out of tune. The VCO's on the ARP are among some of the most stable ever built, partially because they have a temperature compensation circuit. If the temperature of the room changes even slightly, some analog oscillators will rapidly drift out of tune because they lack this crucial feature or it is poorly implemented. However, because changes in Fc are not as readily apparent to our ears as changes in tuning, the filter has no temperature compensation. Thus, when the filter is used as an oscillator, its tuning can be somewhat problematic and unreliable. Finally, the filter serves an important role on the ARP 2600 by mixing together the outputs of the various oscillators and other sound producing devices. (This will be discussed later on.) Once again, if the filter is used as an oscillator, this facility is lost. It is important to understand the principle of self-oscillation to understand why another tone is being added to a patch if resonance is set too high, rather than having the knowledge that the VCF could be used as an oscillator.

Modulation and the VCF

So far two parameters of the VCF have been discussed: Fc and resonance. However, nothing was mentioned about modulating those parameters. Although a few synthesizers allow the amount of Q to be controlled using control voltages, most (including the ARP) do not. Thus, it is the Fc which is the most important parameter, and the **only** parameter that can be modulated on the ARP's filter.

All of the modules that have been discussed so far are normalled to the filter. VCO-1's square wave output, VCO-2's pulse output, VCO-3's saw output, and the noise generator are all normalled to the filter's audio inputs.

In addition to its five audio inputs, the VCF on the ARP 2600 also has three control inputs, located just to the right of the five audio inputs. Modulation on the filter is straightforward. As more voltage comes in, the filter's Fc is raised. If negative voltage is connected to the control input, the Fc will be lowered. (Recall from Section 3 that both the sine and triangle waves range from +5 volts to -5 volts.) Notice that each of the control inputs on the filter have an attenuation slider; all the inputs but one, that is. The

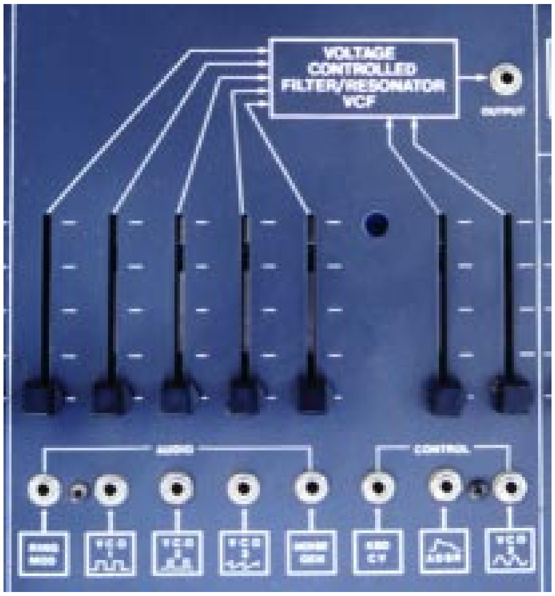


Figure 6-5: The VCF's audio and control inputs

reason for this will become clear in a moment. For now, just notice what is normalled to each of the control input jacks. The VCF's five audio inputs and three control inputs can be seen in Figure 6-5.

The left most jack is normalled to the keyboard CV (this will be explored in a moment). The middle control jack is normalled to a module that has not yet been discussed. The right most jack has been normalled to VCO-2's sine output. Again, it is important to note the normals, as they indicate what will be connected to the control jacks most frequently. Also notice that the filter's output is normalled to the mixer's input.

Key Tracking

As mentioned before, when the fundamental frequency of a sound gets higher than the set Fc, some higher harmonics will be attenuated changing the patch's timbre as it gets higher. If the fundamental frequency is too much higher than the Fc, no sound will be heard at all, since even the fundamental will be entirely filtered. So it would seem that one must be careful to set the Fc higher than the highest harmonic one intends to create. This can quickly become more trouble than it is worth. No one wants to stop to decide what note is the highest they might play, since this really does not invite spontaneous or creative playing. Fortunately, the ARP's designers came up with an inventive way to get around this problem.

Key tracking works by sending a copy of the keyboard control voltage to the left control input on the filter. As higher notes on the keyboard are played, more voltage flows into the control input. As was mentioned a moment ago, the more voltage flowing into the filter's control inputs, the higher the Fc. Thus, the problem of the fundamental tone of different sounds is solved. Note that key tracking is a rather fine adjustment, and not anything as drastic as moving the filter's initial frequency slider even as much as a centimeter. It is easy to observe key tracking in action by using a dummy plug to break the keyboard CV normal to the filter's control inputs. When this normal is broken, sounds become duller and duller as higher and higher notes are played on the keyboard. This can be heard on **CD track 22**.

Highpass Filters

Although the ARP 2600 has only a lowpass filter, it is important to understand the usefulness of the other three filter types, as they will come up time and time again in the study of music technology. The second most common filter type is the *highpass filter*. As its name

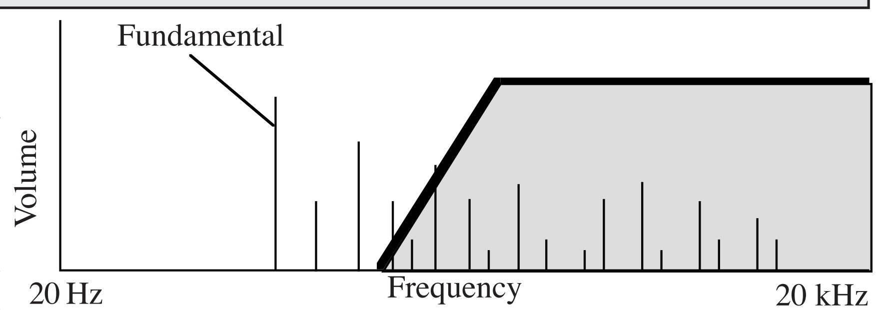


Figure 6-6: The highpass filter in action

states, the highpass filter passes harmonics above the Fc, while attenuating harmonics below the Fc. A highpass filter is the exact opposite of a lowpass filter. Here, the filter is said to be open when the Fc is low, allowing all frequencies to pass through unfiltered. Note that as the Fc is raised, the first harmonic to be attenuated is the fundamental, while the higher harmonics will be the last ones attenuated. The effect of the highpass filter can be seen in Figure 6-6 on page 48. The highpass filter, like the lowpass filter, can have resonance added to it. Although the ARP 2600 lacks a highpass filter, one can be heard on **CD track 23**. This track features a modern Roland synthesizer (a JP-8000) which sports a multi-mode filter.

Band Reject Filters

Once the concepts of a highpass and lowpass filter are understood, understanding a *band reject filter* (also called a *notch* filter) is really rather simple. A band reject filter can very easily be created by connecting the output of a highpass filter to the input of a lowpass filter. It actually makes no difference which filter comes first, the output of one just gets connected to the input of the other.

The band reject filter attenuates frequencies in a specific frequency band. Rather than specifying a Fc, on a band reject filter, one specifies a *center frequency*, around which other frequencies are removed. The distance between what would have been the two Fc's is called *bandwidth*. Thus, in addition to a center frequency parameter which is like the Fc on a highpass or a lowpass filter, the band reject filter also has another parameter: bandwidth.

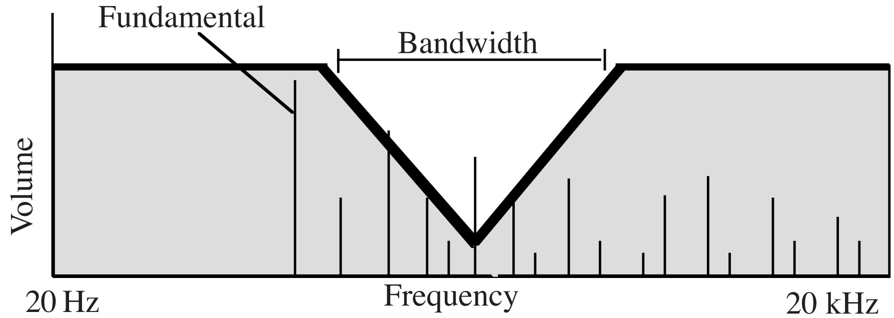


Figure 6-7: The response of a band reject filter

Bandpass Filters

The *bandpass filter* is simply the opposite of a band reject filter. Bandpass filters allow only a set range of frequencies to pass while attenuating all others. This range can be controlled using the bandwidth control. The range of frequencies passing is adjusted with the Fc control. Bandpass filters are commonly used on voices to recreate the sound of a telephone conversation or cheap clock radio. By the nature of the small cheap speakers used in these devices, they can only produce a limited set of midrange frequencies while attenuating low and high frequencies. A Bandpass filter sweep can be heard on **CD track 24**.

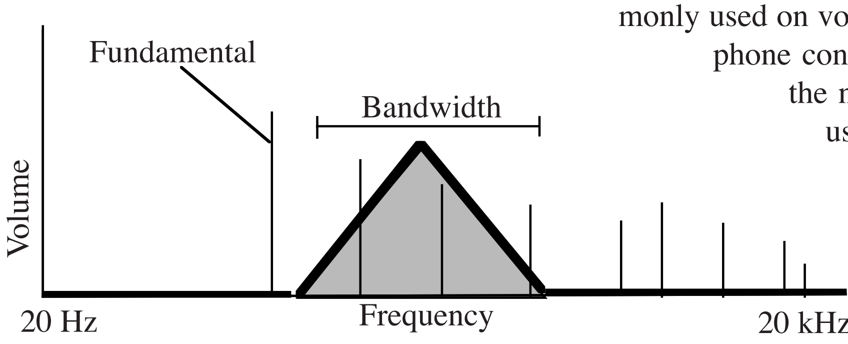


Figure 6-7: The response of a bandpass filter

The Role Of The Filter In The Synthesizer

It is important to realize that filters are not a 'set and forget' module of a synthesizer. Although there are times when their Fc is set and left at a particular frequency, most of the time, the Fc will be constantly changed via modulation. This constant change is what brings the sound to life and makes the sound interesting. There is nothing more deadly than a synthesizer sound that does not change. Some musicians think that they can make up for incredibly dull sounds by playing lots of interesting notes, but when it comes to synthesizers, the notes that are being played are only half of the music being produced. In the world of synthesizers, the sound itself is as important as the music being played.

Using the VCF as a Mixer

In addition to control sliders for cutoff frequency, fine tuning of Fc, and resonance amount, a physical survey of the VCF will reveal that it is blessed with eight inputs. The three inputs on the right side are control inputs and the left five inputs are audio inputs. These inputs allow signals to be input to the filter so that they can be filtered and passed out the filter's output (right hand side of the filter). It is interesting to note that signals move left to right even within different synthesizer modules.

There are many inputs on the filter so that many different signals can be fed to the filter for filtering at once. However, the synthesist can use this feature to his/her advantage by using the filter to mix several sounds together. All of the audio inputs have a slider above them which allows the user to control the volume level of each signal being input. The inputs are mixed together, filtered, and appear at the filter's output.

Notice also that each of the filter's audio inputs is normalled to the output of a different module. One is normalled to each of the VCO's, one to the noise generator's output, and the final one to a module yet to be discovered. Thus, another piece of the puzzle has been filled in for us. The most basic patch starts with oscillators and possibly the noise generator, all of which is then fed to the filter.

The VCF In Practice

Up to this point, this section has dealt with raw factual information about filters, their types, how they work, etc. However, nothing has been said about how they are commonly used in synthesis applications. Generally, the outputs of the oscillators (and sometimes the noise generator) will be routed to the audio inputs on the filter (note that they are already normalled there). Notice that there is an attenuation slider above each of the audio inputs. This allows a user to control the volume of each incoming signal. Thus, the filter is a very useful tool for mixing sounds together. When a little resonance is added, and the filter's Fc is swept up and/or down, an effect called a *filter sweep* is created, which is one of the most commonly used filtered sounds today.

Another way in which the filter is extremely helpful is that it can stop the constant monotonous output of the oscillators. Of course, the oscillators continue to oscillate no matter what they are connected to. However, when the filter's Fc is set low enough, it can stop all sound coming through the filter. When the device in the next section is explored (the envelope generator), the process of automating the change in Fc will be explained.

The filter is also the device used to shape the overall sound the instrument will produce. If one wants duller sounds, the Fc should be decreased. If one wants brighter sounds, the Fc should be increased. Innovative synthesists use modulation to change the Fc and create constantly evolving, vibrant sounds which capture the listener's imagination. Remember also that the ARP is a modular instrument, and that any electronic signal from the rest of a studio is fair game for processing through its wonderful filter!

The filter is by far one of the most important modules in modern synthesis. It is so important that subtractive synthesis remains one of the most important kinds of synthesis today. Although now there are many different forms of synthesis, most are designed to mimic subtractive synthesis in their operation to make programming simple. Thus, someone who is familiar with the process of subtractive synthesis can work almost any synthesizer in the world after only a few minutes to figure out how its user interface works.

Experiments for Section Six: VCF

1. Demonstrate a nonmusical filter such as the handkerchief experiment described on page 42.

2. Route VCO-1's square output to the filter's audio input. Demonstrate that this is a redundant connection. Route the filter's output to the mixer. Demonstrate that this is a redundant connection. Bring up the filter's output in the mixer. Raise VCO-1's input in the filter. Change the filter's cutoff frequency setting and listen to the sound change. **CD track 19**

3. Listen to the change in the cutoff slope shape when resonance is added to the sound and the Fc is swept again. **CD track 20**

4. Use the VCF as a mixer. Mix all three VCO's and the noise generator output in different amounts. (No resonance, filter cutoff frequency at maximum)

5. Modulate the filter's Fc with a control signal from VCO-2. Try using different waveforms from VCO-2 as modulators and also try inputting different waveforms into the VCF's audio inputs. See if it is possible to create sidebands using the filter. **CD track 25**

6. Demonstrate the keyboard CV connection to the filter control. Use the dummy plug to eliminate this.

7. Using a dummy plug to eliminate keyboard CV of a VCO, and listen just to the keyboard's CV controlling the filter's cutoff frequency. Raise the resonance level and listen to the various harmonics which are accented as notes are played up and down on the keyboard. **CD track 26**

8. Observe the relationship between the initial frequency setting of the filter and the input from the control jacks. Use VCO-2's sine wave in the sub audio range to modulate the Fc. Notice that the Fc moves first above and then below the level set by the initial frequency slider.

9. Use the filter as a wave shaper. Reduce the harmonics of a saw wave until it sounds like a sine wave. Repeat this experiment with a pulse wave. **CD track 19**

10. Use the filter as an auto-wah pedal. Use a sine wave to sweep the filter in a restricted bandwidth.

11. Use the filter with a lots of resonance (make sure the filter is not self-oscillating, though) to isolate individual harmonics from a saw wave by slowly sweeping the Fc. Why wouldn't this procedure work with a sine wave? Why do certain frequencies 'jump out' while doing this experiment? **CD track 27**

12. Use the filter as an oscillator by causing it to self-oscillate. Control the pitch of the oscillator from the keyboard and explain why it is that this can be done. Play up and down the keyboard and notice how out of tune the filter is even when moving as much as an octave. **CD track 21**

13. After experimenting with the lowpass filter on the ARP, make some generalizations about where the initial frequency slider will be set for most patches.

14. Use the filter as a manually-controlled gate, thus stopping the monotonous sound.

15. Note all of the modules which are normalled to the filter, both audio and control inputs. Also explore the front panel of the ARP and discover where the filter's output is normalled. What additional clues does this give you about how the filter is usually used?

16. Listen to several resonant filter sweeps with different timbres, and identify several filter sweeps in musical compositions. (Your teacher can help you with this.) Discuss the musical use of a filter sweep. **CD track 19-20, 23-24**.

17. Draw some conclusions about these experiments. In general, is the filter's Fc set and left at one point, or moved a lot? Is the resonance amount constantly adjusted, or is it set and left? Is it easier to get a phat sound using the filter than without? Is the filter the most important part of the synthesizer?

Review Questions for Section Six:

1. Compare and contrast FM with filtering. Why could the two be confused? How are they completely different from each other?

2. How does a filter change an incoming waveform? How is this function controlled?

3. Name the four main types of filters, tell how many kinds of filters some modern synthesizers have, and tell how many filters are on the ARP 2600.

4. Tell how filter types are named and what each is useful for.

5. State what happens when a sine wave is passed through a lowpass filter.

6. What happens to a sound which is put through a lowpass filter as the filter's Fc is lowered?

7. Name the two most common cutoff slopes found in filters.

8. Do all synthesizers allow control of resonance, and do all synthesizers have resonant filters?

9. Explain when and why self-oscillation occurs. State why using the filter as an oscillator is not particularly useful.

10. List all of the parameters of the VCF on the ARP 2600. Name all of the parameters that can be modulated.

11. State the relationship between the initial frequency slider and the incoming control voltages.

12. Explain where the initial frequency slider will be set for most sounds.

13. Explain how key tracking can be disabled.

14. State the filter's role in shaping sound on the synthesizer, and its importance to musicians.

Terms To Know:

Bandpass filter
Band Reject Filter
Bandwidth
Center Frequency
Close
Cutoff Frequency
Cutoff Slope
Emphasis
Fc
Filter
Filtersweep
Highpass Filter
Key Tracking
Lowpass Filter
Multimode Filter
Notch filter
Open
Pole
Q
Resonance
Self Oscillation
Subtractive Synthesis
Sweep
VCF
Voltage Controlled Filter

SECTION 7

ADSR & AR

I Hope You Have the Right Postage for That Envelope

While the ADSR and AR envelope generators are certainly not the most glamorous modules on the ARP 2600, they are possibly some of the most useful and helpful in controlling the instrument. Before the purpose or function of these modules is discussed, it is important to understand what an envelope is.

An *envelope* can be thought of as a graph of changing voltage over time. A typical envelope is illustrated in Figure 7-1. This changing voltage is produced by a device called an *envelope generator*, or *EG*. One might think that this definition of an envelope sounds very close to the definition of a waveform, which is also a graph of changing voltage over time. There are several differences between the two, however.

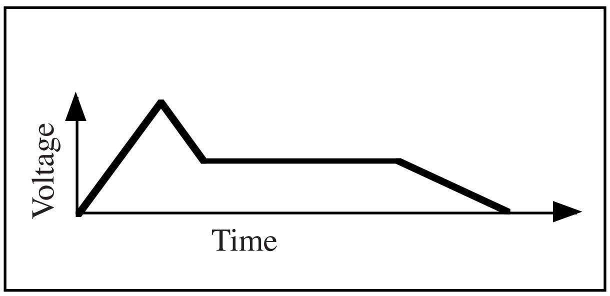


Figure 7-1: A typical envelope

First, an envelope usually moves rather slowly, and can sometimes take up to a minute to be produced just once (about 0.016 Hz), whereas even the slowest LFO will move about twice as fast. While oscillators will frequently produce waveforms whose frequency is in the audio range, an EG will almost never produce an envelope so quickly that it could be heard.

The second big difference between the two lies in repetition. While a waveform from an oscillator repeats over and over (usually thousands of times per second) an envelope is produced only once, and then the EG waits for a signal to begin the envelope again. So the question becomes, 'what is the signal that the EG waits for?'

The Keyboard's Three Signals

The signal that the EG waits for is a +15 volt spike of voltage that lasts only a moment. This spike is called a *trigger pulse*. The next question is: 'where does one get such a pulse?' Reviewing the modules studied so far, it is clear that the oscillators won't work. While an oscillator could produce a pulse wave with a very narrow width, the pulses would have to be fairly far apart. One will recall that oscillators can't really oscillate slowly enough to make this practical. The noise generator and filter are equally of no help in solving this problem.

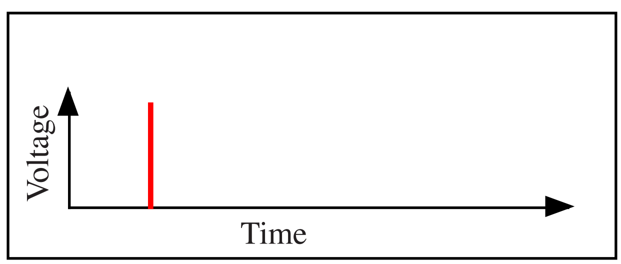


Figure 7-2: A trigger pulse

It turns out that the designers of the ARP put the circuit that generates the trigger pulse in the most logical place of all: the keyboard. In Section 1, the cord that connects the keyboard to the cabinet was described as having six prongs, one of which carried raw voltage to power the keyboard and one pin to return the CV. Now, another pin is explained. This pin returns trigger pulses created by the keyboard. Each time a key on the keyboard is pressed, a short burst of voltage is sent back to the cabinet.

The keyboard's CV has its own jack on the front of the cabinet on the left side, so one might think that the trigger output should have its own output as well. However, it is rare to need a copy of this signal, so there is no trigger output. There is an additional trigger input on the 2600 so that external devices and signals can be used to trigger the EG's. This input is located just below the AR generator on the ARP's cabinet and it is labeled TRIG. Some specific uses of this input are discussed in Section 15.

The ARP 2600 provides another way to trigger the EG's without using a trigger pulse from the keyboard or from an external source. There is a single red button just above the AR generator which is labeled MANUAL START and when it is pressed, it sends out a trigger pulse to both EG's simultaneously. The MANUAL START button can be seen in Figure 7-3.

Figure 7-3: The manual start button

So now that the basic concepts of an envelope, envelope generator, and trigger pulse have been explained, it is time to talk about how EG's are controlled. EG's allow users to change the envelope they produce by allowing users to control two elements: level and time. For convenience, the envelope is split up into several different parts often called *stages*, each of which has a specific name.

Attack!

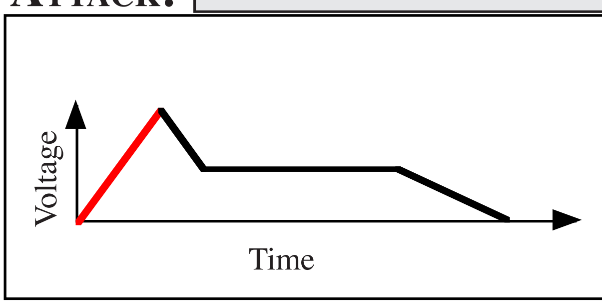


Figure 7-4: Attack

The first stage is called the *attack*. Attack allows the user to change the amount of time it takes the EG to reach its fullest height. The shorter the attack time, the faster the voltage rises to its greatest amount. The attack stage begins as soon as a key is depressed and lasts until the amount of time specified is up. A trigger pulse starts the attack stage. In Figure 7-4, the attack stage of the envelope has been drawn in red.

Decay and the Downward Slide

The second stage of an envelope is *decay*, which is the amount of time the EG takes to decrease from the greatest height of the attack to the next stage. Again, it is important to note that decay is a parameter which deals with time, not a level. The decay stage of an envelope has been drawn in red in Figure 7-5. The slope of the decay actually changes when the amount of decay time changes.

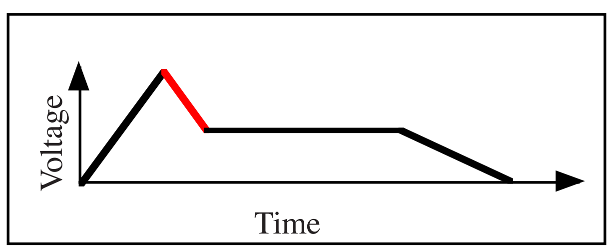


Figure 7-5: Decay

Hold it Right There!

The next stage of the envelope is unique in that it is the first, and only stage to deal with a level instead of time. The third stage of an envelope is called *sustain* and it determines the amount of control voltage the EG will put out while a key is being held down. Of course, this creates another little conundrum. How does the EG know when a key is being held down?

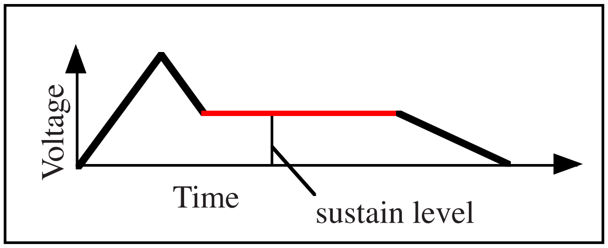


Figure 7-6: Sustain

Yet another pin on the cable which connects the keyboard to the cabinet is used to send voltage to the cabinet as long as a key is being held down. This voltage, which is 'on' while a key is held down, is called a *gate signal*. There is an input for an external gate signal which is located just to the left of the trigger input. Once again, specific uses will be described later. As with the trigger pulse, the manual start button will also produce a gate signal as long as it is held down, so that all the stages of the envelope can be heard.

Because sustain refers to a level and not a time period, the sustain stage sometimes causes some confusion. The time between the end of the decay and the beginning of the final stage is called the *sustain*, but the level of this stage is also referred to as *sustain*. The amount of time this stage lasts is determined by the length of time a key on the keyboard is held down or the length of time the manual start button is held down.

Please Release Me; Let Me Go

Finally, when a key on the keyboard is released, the gate signal being fed to the EG is abruptly cut off, and the EG goes into its final stage, called *release*. Release determines the time it takes the envelope generator to go from the sustain level to no voltage. Release can be seen in Figure 7-7 where the release portion of the envelope is drawn in red.

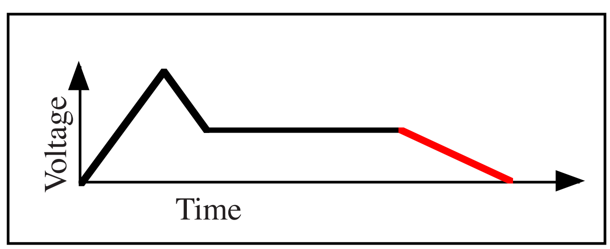


Figure 7-7: Relase

Taking the first letter of each stage, one gets the abbreviation "ADSR" which is pronounced "Add-Sir." One will frequently refer to the ADSR generator, but the other generator on the ARP is merely called the A-R generator. Although *Bob Moog* invented the VCO and VCF, the idea for the ADSR generator was not his. Although the Moog music company built the first ADSR generators, the module was the idea of Russian electronic music composer *Vladimir Ussachevsky* (1911-1990), who was *Wendy Carlos's* teacher at the Columbia-Princeton Electronic Music Center. How are each of these times and levels set? The ARP 2600 provides the user with a separate slider for each stage of the envelope. With the sliders, one can almost see a sort of graphic representation of the envelope about to be produced.

When A Key Is Played

To summarize what happens when a key is played on the keyboard: The keyboard generates a trigger pulse which is sent to the EG's. (As soon as the key is pressed, the keyboard begins generating a gate signal as well.) The trigger pulse causes the EG's to begin the attack stage. Following the attack stage, they go into the decay stage. Then, if the key is held down, the gate voltage will keep the EG in the sustain stage as long as the gate voltage is present. When the key is released, the gate voltage is instantly gone, and the EG begins the release stage, during which it decreases gradually to zero volts. This brings up a rather interesting question: What happens if a key is played, but released before the EG reaches the end of the attack stage? It is possible to set the attack time so long that a key can be released before the attack is complete, but when the key is released and the gate voltage disappears, the EG will immediately jump to the release stage. This is true of releasing the key at any time during the first three stages.

It is also interesting to note that it is possible to program envelopes that have no sustain level at all, and if a key is held down, the EG will stop producing voltage after the decay. However, it is also possible to program a sound which has no sustain, but has a release time. If the key is released before the EG gets to the sustain stage, a release will occur. If the key is released after the sustain stage is reached, no release will occur. This is because of the actual functioning of the release stage.

The release stage is activated exactly when the key is released, and it will cause the voltage to ramp down from wherever it was when the key was released. Remember: release is a setting of time, not a level setting. Thus, the release stage will always cause the voltage to decrease from where it was last being produced rather than decreasing voltage from a set level every time.

Theme and Variation

Thus far, the AR generator has received little attention. This is because it is very similar to the ADSR generator. One might ask, "but what about the decay and sustain stages?" In the AR generator, the decay stage is not present. This is acceptable, as it is the least noticeable of all of the stages. The sustain stage is still present; it is just not programmable by the user. It is permanently set to full open.

It is interesting to note that when the sustain stage is set to full open on the ADSR generator that the decay parameter has no effect on the envelope that the EG produces. This is because the decay stage sets the amount of time the EG will take to decrease from the highest point of the attack to the sustain level. When the sustain is set full open, the decay becomes an early extension of the sustain stage, as illustrated in Figure 7-8. Since the sustain stage in the AR generator is permanently set full open, there is no need to even consider including a decay stage in this module. Although the AR generator has fewer features than the ADSR EG, it is still very important to synthesists.

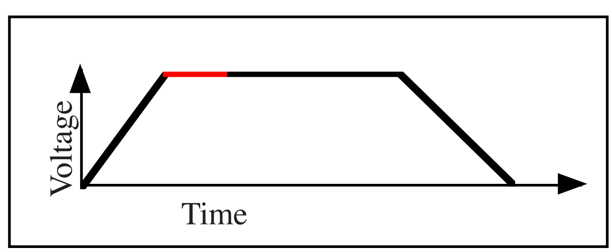


Figure 7-8: The decay stage is negated by the sustain stage

EG's In Practice

The envelope generators on the ARP 2600 are used exclusively as a source of control voltages. They can be used to control the frequency or pulse width of an oscillator or the cutoff frequency of the filter. The ADSR EG is normalled to FM inputs on each oscillator, as well as one of the control inputs on the VCF. When used to FM a VCO, an envelope generator can produce a wild disturbance in pitch depending upon how deep the modulation is set. More importantly, the EG's can be used to raise the filter's Fc every time a key is pressed, and thus stop or *gate* the sound when a note isn't being played. Essentially, the envelope generators are a useful tool whenever one wants to have a voltage contour created whenever a key is played. Of course, there are other ways to cause the EG's to fire, but their use is generally tied to a key press.

Figure 7-9: The ADSR and AR generators

The level or time of each stage of the EG's is set using sliders which can be seen in Figure 7-9. One will note that the EG's only have outputs, but no inputs, which means that they cannot be modulated. It is possible to design EG's which allow voltage control of each stage, but such features are a rare item in a commercially-produced synthesizer.

Observant persons may note that there is a jack and a switch just below the AR generator which has gone unexplained. However, this jack is intimately connected to the module which will be discussed in Section 10, so it is best left unexplained until that time.

The Evolution of the EG

Modern developers have changed the EG in many different ways, but the most simple change has been the addition of more stages. Some companies offer DADSR generators, which have a programmable delay time before they begin the attack stage. This is particularly useful if many EG's are available, as they can all fire at slightly different times after a key is pressed. Other synthesizer companies such as E-mu Systems have developed a DADHSR EG which not only had the delay stage, but an extra 'hold' stage as well. Many modern synthesizers have abandoned the ADSR concept entirely and have begun to just allow users to set four different times with four different levels. Some EGs even have up to eight stages! These super flexible EG's are explored in depth in the second volume of this series.

Experiments for Section Seven:

1. Use the ADSR EG to frequency modulate VCO-1, 2, and 3. Notice that this is a redundant patch. Trigger the EG with the keyboard. Experiment with different frequencies, and different waveforms. Repeat this experiment using the manual start button instead of the keyboard.

2. While conducting experiment #1, create a patch that just has attack. Try adding different amounts of attack, at different modulation depths. Is it an effective technique to change the attack time while notes are being played? **CD track 28**

3. While conducting experiment #1, create a patch that just has decay. Try adding different amounts of decay, at different modulation depths. Is it an effective technique to change the decay time while notes are being played? **CD track 29**

4. Compare and contrast experiments #2 and #3.

5. While conducting experiment #1, create a patch that just has sustain. Try different amounts of sustain, at different modulation depths. Change the sustain while holding a note. **CD track 30**

6. While conducting experiment #1, create a patch that just has release. Try adding different amounts of release, at different modulation depths. What happens here that is unusual? **CD track 31**

7. While conducting experiment #1, try releasing the key being used to trigger the envelope before the EG has moved through the entire attack stage. Repeat and try to cut off the envelope during the decay stage. What stage does the EG jump to?

8. How long is the longest attack? The longest decay? The longest release? When would each of these be useful? Notice the effect of having a very short attack or decay time on the sound.

9. Use the ADSR EG to modulate the filter's cutoff frequency. (Notice that the ADSR is already normalled here.) Begin by creating a sound which uses white noise from the noise generator which is then fed to the filter. Close the filter completely. Set the attack and decay times very very low, with no sustain or release. What happens when a key is played? Try changing the ADSR settings slightly to create percussion sounds. Try adding resonance. **CD track 32**

10. While conducting experiment #9, use the VCO's in a phat tuning instead of the noise generator. Create a patch that has only attack. Is it an effective technique to change the attack time while notes are being played? **CD track 33**

11. While conducting experiment #9, create a patch that has only decay. Is it an effective technique to change the decay time while notes are being played? If low enough notes are played, is this an effective bass sound? **CD track 34**

12. While conducting experiment #9, create a patch that has only sustain. If low enough notes are played, is this an effective bass sound? **CD track 35**

13. While conducting experiment #9, create a patch that has only release. Is it an effective technique to change the release time while notes are being played? When a long enough attack time is selected, and slurred notes are played, what begins to happen to the notes? **CD track 36**

14. Are the EG's a useful way to control the filter's cutoff frequency? Where does one usually have the Fc set for this operation? What happens to this patch with the addition of resonance? Is it possible to create an automated filter sweep? **CD track 27**

15. Try creating the patch in experiment #9 simultaneously with the patch in experiment #1. How is this not as useful as each patch alone? Use the AR generator to control the pitch of an oscillator while using the ADSR generator to control the filter's Fc. Is this more useful than the patch in #14? **CD track 37**

16. Now create a patch in which the AR EG controls Fc and the ADSR controls the pitch of an oscillator. In general, which is better suited to controlling the pitch of an oscillator, the AR or ADSR generator? **CD track 38**

17. Why would it be useful to have an extra EG on the 2600?

18. Create a bass patch with a short decay, little sustain, medium release and the attack set to about 80%. Can this patch be played in such a way that true legato is possible?

19. Using the ADSR generator to control the frequency of an oscillator, create a patch which automatically bends up to the proper pitch each time a key is pressed. **CD track 39**

20. Patch the ADSR generator to all of the oscillators (review Section 1 if you forgot how) and set different modulation depths for each oscillator. Route the VCO's to the VCF and use the AR generator to control the Fc. **CD track 40**

21. Draw the envelopes of several common instruments. Be sure to include snare drum, piano, flute, and voice. Draw several.

22. Use the ADSR generator to PWM VCO-2. How is this timbral shift made even more interesting when the same EG is controlling the VCF's Fc? **CD track 41**

Review Questions for Section Seven:

1. Compare and contrast envelope generators and oscillators.

2. Who designed the ADSR generator, and who built the first one?

3. Name the three control signals the keyboard produces and tell when it produces each one. Tell what effect each one has on an EG.

4. Name the four stages of an ADSR generator, and tell if each is a time or level setting.

5. Be able to pick the correct settings of an ADSR generator by looking at a drawing of an envelope.

6. State what is different physically and in operation between the ADSR and AR generators.

7. Tell what happens when the gate signal disappears at any given point in the envelope.

8. Tell what happens to the decay stage when the sustain level is set to full open.

9. Locate the manual start button and tell what it does both in general and in terms of electronic signals.

10. Name all of the places to which the ADSR and AR generators are normalled.

11. Review the settings for percussion, bass, and common other patches on the 2600.

12. Tell how the EG's could be caused to fire without a trigger pulse or gate signal from the keyboard and without using the manual start button.

Terms and Names To Know:

ADSR
AR
Attack
Bob Moog
Decay
EG
Envelope
Envelope Generator
Gate Signal
Manual Start Button
Release
Stage
Sustain
Trigger Pulse
Wendy Carlos
Vladamir Ussachevsky

SECTION 8

VCA

Introduction to the VCA

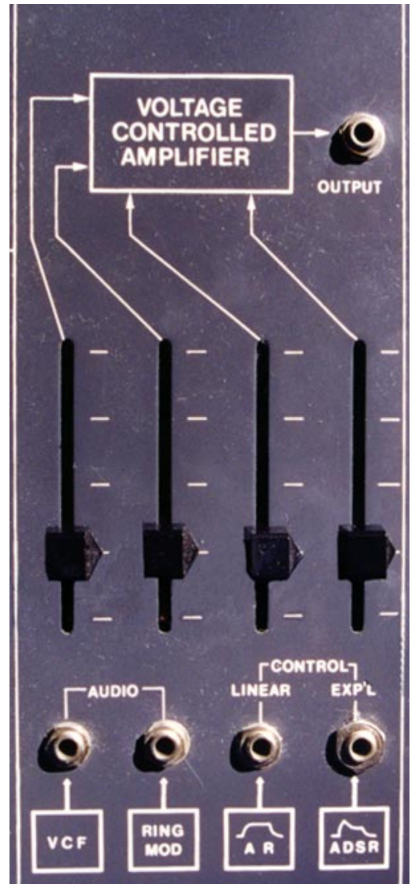


Figure 8-1: The VCA

The final module in the signal path of a typical synthesizer patch (not counting the speakers and amplifier) is the *voltage controlled amplifier*, or *VCA*. The VCA performs a very simple task on the ARP 2600: It is responsible for controlling the *amplitude* or volume of the signals which pass through it.

A physical survey of the VCA module will reveal that there is really not much to it. In Figure 8-1, one can see that it is blessed with four inputs, two of which are audio, two of which are control inputs. It has sliders above each of the four inputs for attenuating incoming signals. It has a single output jack and a slider labeled INITIAL GAIN. *Gain* is another word for volume.

Of course, the inputs are aligned with most of the other inputs on the cabinet, including those of the VCF, and the VCO's. One of the two audio inputs is normalled to the filter's output, which makes sense. The VCA is the last module in many synthesizer patches. So, a block diagram of the signal flow of a typical synthesizer patch can now be drawn (See Figure 8-2.)

Of course, a patch may contain additional oscillators, possibly the noise generator, and one EG might be used to control both the VCA and VCF. One might even use an EG to control the VCO. However, Figure 8-2 is the fundamental synthesizer patch. It should be thought of as a common element, as many of the world's synthesizers will use this layout as a blueprint for sound production. Of course, there are exceptions, and modern synthesizer designers have usually replaced voltage controlled modules with digitally controlled counterparts. Their parameters look, and feel the same, even if their exact operation and electronic functioning do not. This is why it is so important to learn about subtractive synthesis. Roughly 80% of the synthesizers in the world are either truly subtractive, or set up their controls to emulate a subtractive synthesizer. The most notable exception to this are the synthesizers which use FM synthesis or additive synthesis as their main synthesis method.

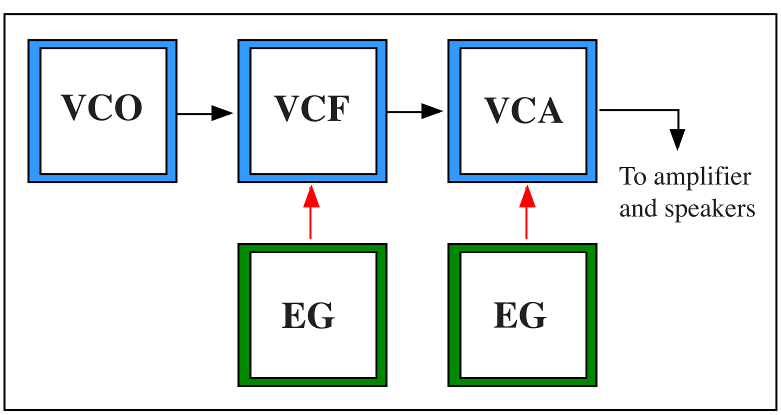


Figure 8-2: A typical synthesizer patch

The VCA Vs. The VCF

One may recall from Section 6 that by setting the VCF's Fc all the way closed, one can stop the signal from passing through the filter, thus changing the signal's volume. Isn't this what the VCA does? At first glance, it might seem so. Actually, as the filter's Fc is lowered, the actual timbre of the sound coming through it will change as well, since harmonics will be removed from the sound and the sound will become duller as the Fc gets lower. When using the VCA to change amplitude, the signal coming through will have a consistent timbre regardless of the volume at which the VCA is outputting the signal. One can hear the difference between a filter sweep and a VCA sweep by listening to **CD track 42**.

How Does One Use It?

Like the VCF, sounds input at the audio inputs will emerge from the audio output. Sounds coming into the VCA actually pass through it and come out the output. The VCA does not actually make any sound. In order to use the VCA, audio signals must be fed into it via one of the two audio inputs. The attenuation slider above this input must be raised, or else no sound will be permitted to enter into the VCA. One must take care not to raise the slider too high, however, as distortion can occur. *Distortion* is a state in which more electrical signal than a circuit can handle is put through a circuit and it will be discussed in Section 12.

The INITIAL GAIN must also be set. Under normal circumstances, the INITIAL GAIN slider will be set to closed (full left) as it will then *gate* the sound and create silence in between the times when notes are being played. When an audio signal is being fed to one of the VCA's audio inputs, moving the INITIAL GAIN slider is like turning the volume knob on a radio. It changes the amplitude of the signal coming out.

Most of the time, it will be desirable to have another module modulating the amplitude of the incoming waveforms on the VCA. One might think of this as 'amplitude modulation,' and indeed this is a form of AM. However, this term will usually be used for another technique which will be explained at another time. Generally, either an EG or an LFO is used to control the VCA's amplitude. Careful observation will show that one of the two EGs is normalled to each of the control inputs on the VCA. The VCA's gain or amplification is the only parameter which can be modulated on the VCA, hence the name 'voltage controlled amplifier.'

Using the VCA in Patches

If one creates a patch in which there is an audible signal present at the VCA's audio input, the corresponding attenuation slider is raised, and the initial gain is set fully closed, the VCA will be silent since the gain is set fully closed. As more voltage comes into the control input, the VCA will allow more and more incoming signal to pass through to its output.

It is interesting to note that the VCA's gain is reduced from the initial gain setting if the incoming control voltage is negative. For instance, if a sine wave is fed to the control input, the gain will be increased, then decreased from the initial gain setting. In this way, it is similar to the effect of negative control voltage on the VCF's cutoff frequency. However, there are very few practical applications of this design, since it means that sound is constantly flowing through the VCA, regardless of whether a key is being played. Over time, this design disappeared. Almost without exception, modern VCA's will only increase gain when responding to incoming control signals.

Linear Vs. Exponential

In Figure 8-1 on page 62, one can see that the two control inputs on the VCA have a small label over them that says control. The left jack, which is normalled to the AR generator, is also labeled LINEAR, while the right jack, normalled to the ADSR generator, is labeled EXPONENTIAL. Although many early synthesizers offered one or the other, the ARP 2600 was among the select few that offered users both response curves.

Volume is typically measured in a unit called *decibels* or *dB*. One dB is the smallest change in volume that a person can perceive. Although people can hear volumes much louder, 140 dB is considered to be the *threshold of pain*, or the volume at which a sound is so loud, it is painful to hear. 1 dB is the softest sound that human ears can hear. One might think that in order to create a sound which is twice as loud as a 50 dB sound, one would just double the voltage being fed to the VCA. However, human ears do not respond in the way that one might think. The decibel is not a linear measurement. Although the decibel scale increases smoothly upwards in a straight line (the blue line in Figure 8-3), the amount of sound energy it takes to produce those sounds does not (the red line in Figure 8-3). In fact, a 20 dB sound is actually ten times as intense as a 10 dB sound. A 30 dB sound is actually one hundred times as intense as a 10 dB sound, and a 40 dB sound is actually one thousand times as intense as a 10 dB sound.

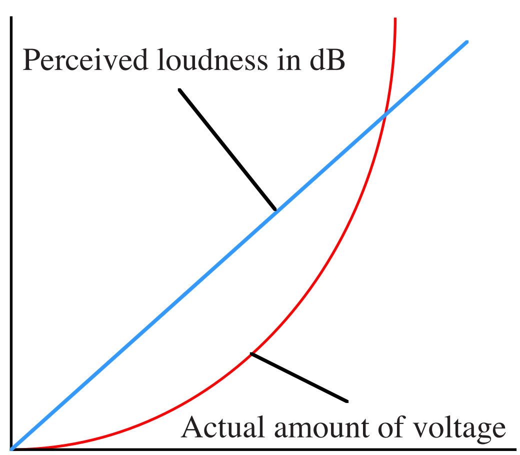


Figure 8-3: A linear and a logarithmic curve

When it takes increasingly large amounts of volume to produce the same amount of perceived change, this is called an *exponential* response. Human ears function in this way. Sometimes, it is advantageous to have an amplifier which functions in a linear mode, and sometimes it is advantageous to have an amplifier that functions in exponential mode. The ARP 2600 gives the user both options.

When the linear control input of the VCA is used, volumes will seem to change smoothly over the course of the envelope or whatever is controlling the VCA's amplitude. While using the linear input, it does not take as much voltage to create a change in gain, especially at lower levels. When the exponential input is used to control the VCA, the changes in amplitude will be more dynamic and certain elements of an envelope will seem exaggerated (most notably the decay). If a waveform which is naturally smaller in amplitude is used (e.g. a sine or triangle wave) little change may be heard using the exponential input. The difference between the two control inputs can be heard on **CD track 43**

Of Lawsuits and Patent Infringement- Part II

Many of the modules studied thus far were developed by Bob Moog, and the VCA is no exception. It is interesting to note that at the same time Dr. Moog was building his first VCA, Vladimir Ussachevsky was also designing a VCA. He shared his plans with Dr. Moog, only to discover that Dr. Moog had already designed and built his own VCA. However, the Moog VCAs did not have an exponential option at first. This requires the use of an electronic circuit called a *linear to exponential converter*. This circuit just turns linear signals into exponential ones, as the name implies. This circuit was designed by the ARP company, and it appeared on the gigantic ARP 2500 modular synthesizer.

Before too long, the circuit also began appearing on Moog synthesizers without ARP's permission. One may recall from Section 6 that ARP had infringed on Moog's filter design patent, and that Moog music threatened to sue ARP if the design was not changed. ARP did change its filter design, but it needed time to do so. They told Moog music in no uncertain words that if they sued ARP for infringement of the filter patent, that they would return fire with a lawsuit over the linear to exponential converter circuit, which Moog Music actually continued to use until its demise.

Other Fun Things To Do With A VCA

Since there are two inputs on the VCA, it could really be used as a mixer, albeit a small one. Whatever two signals are fed into the VCA will be combined together and appear at its output. One must again be careful of distortion, though. Typically, when two signals are used in the VCA, the levels must be set even lower than when one signal is used.

By controlling the VCA with a sine wave in LF mode, one can create the popular effect called *tremolo*. Tremolo is a constant change in volume and creates a wavy sound not dissimilar to vibrato. It is possible to purchase a device which is a dedicated tremolo effect unit, and this is common in the world of electric guitars. These units simply contain a variable rate LFO which always produces a sine wave which in turn modulates a VCA. An example of tremolo can be heard on **CD track 44**.

Users may be interested to note what occurs when the VCA's gain is modulated quickly and deeply enough by a VCO in LF mode: just like FM in the audio range, sidebands are produced. They sound a bit different from their FM counterparts, but they are definitely sidebands, nonetheless. An example of these sidebands can be heard on **CD track 45**.

Because of the flexible design which is employed in the ARP 2600, the VCA can be used to process control signals as well as audio signals. In this respect, the VCA can be thought of as a voltage controlled attenuator for any incoming signal. However, the VCA cannot simultaneously process both audio and control signals since they would be mixed together. When it is used to process control signals, the VCA cannot be used to process audio signals, which leaves the task of gating the sound up to the filter. Of course, using the filter is not the best choice for this task, since the filter will change the timbre of the incoming sound as it changes its volume. However, it is important to understand that the VCA can be used to adjust the amplitude of incoming control signals.

The VCA In Practice

Different patches call for different settings, but there are some general guidelines which can help to create highly effective patches. In general, the initial gain setting will be set at its lowest setting so that the VCA acts as a gate in between notes. It is also highly effective to control the VCA using the AR generator, with no attack and maximum release time. This will ensure that the gate will open fully as soon as a key is pressed, and then close slowly so as not to cut off any release on the filter. Of course, this reduces the VCA to a simple gate and it is not allowed to perform any particularly interesting function, but it is always a good place to start so that one can get an idea of the raw, unshaped sound before it is altered by the VCA. It is also possible to use two different control signals to control the VCA simultaneously. Remember that only the highest incoming voltage will have any effect, however.

The VCA's settings will determine the overall volume of the sound, and more importantly, how that volume will change over time. It is important to be able to recognize the envelopes of different instruments and sounds so that they may be easily recreated using the VCA and an EG. For instance, a drum might have a very short attack, no sustaining volume, and no release. Attention to detail in setting these parameters are extremely important to the overall sound the 2600 will produce.

Summing It All Up

The VCA has a total of five parameters, only one of which can be modulated. The first two parameters are the attenuation levels of the audio inputs. The second two parameters are the attenuation levels of the two control inputs (linear and exponential). The final parameter is the initial gain, set using the INITIAL GAIN slider. Initial gain can be modulated using control voltages input in either the linear or exponential inputs. The gain can be increased by positive voltages and decreased by negative voltages. The VCA can also be used to attenuate the level of control signals.

The VCA is the last module in a typical synthesizer patch, and as such, it performs the task of shaping the overall volume of the sound. It also performs the task of gating the sound so that no sound comes out when no keys are being played.

Experiments for Section Eight

1. Connect the VCF to the VCA. Demonstrate that this is a redundant patch. Use the lowest initial gain setting to 'gate' the incoming sound and use an EG to open and close the VCA.

2. Connect VCO-3's saw output to one of the audio inputs on the VCA. Raise its attenuation slider. Try moving the initial gain slider to determine its effect on the sound.

3. While conducting experiment #2, connect a sine wave from VCO-2 in LF mode to the linear control input. Raise its attenuation slider. Try changing the attenuation of both the incoming audio signal and the incoming control voltage. After observing the effects of these settings, try increasing the initial gain setting, and notice how the effect of the incoming control voltage is decreased. What effect is being created here? **CD track 44** Try increasing the rate and depth to produce sidebands. **CD track 45**

4. Try experiment #3 again, but this time, use the exponential control input. What sounds different about the output now? Why is there almost no change in amplitude? Try using VCO-2's pulse output instead of the sine output. Why is this more effective?

5. Patch VCO-3's saw wave to one of the audio inputs of the VCA, and this time, use the ADSR generator patched to the linear control input (redundant patch) to control the VCA. Repeat this experiment, and this time patch the AR generator to the exponential response circuit. What is different about this sound? **CD track 43**

6. Create a patch in which VCO-2 and VCO-3's saw waves are fed into the VCF. The VCF's Fc should be controlled by the ADSR generator. The VCF's output should be fed to the VCA. Control the VCA using the AR generator. Notice how effective this patch is. Note which settings are the most beneficial on the AR generator regardless of the settings of the ADSR generator. Now repeat this experiment, and control the VCA using the ADSR generator and control the VCF using the AR generator. Which of the two configurations do you prefer? **CD track 46**

7. Connect two different VCOs tuned to different pitches to each of the VCA's audio inputs, and demonstrate its ability as a mixer.

8. Compare the VCF's gating to the VCA's gating by patching the saw wave from VCO-2 to the audio inputs of each, and bring up the outputs of the VCF and the VCA in turn at the mixer. Start with the VCF, open the filter completely, with no resonance. Sweep the Fc down until the filter is closed. Notice that as the sound got softer, the timbre of the sound changed as well. Next, listen to the VCA perform the same gating task by starting with the INITIAL GAIN set full open and then sweep it down until it is fully closed. Notice that the timbre of the sound remains constant throughout the sweep, and only the amplitude or volume of the sound changes. **CD track 42**

9. Note what is normalled to each input of the VCA. Notice where the VCA's output is normalled.

Review Questions for Section Eight:

1. Draw a diagram representing the signal path of the most typical synthesizer patch. Why is this patch so important to understand?

2. Tell what the VCA does. In what way is a waveform reshaped by the VCA?

3. Compare and contrast the VCA and the VCF.

4. What does the initial gain setting control, and how does this relate to incoming control voltage signals? What could a sine or triangle wave do to take advantage of this function?

5. Describe the differences between the response of the linear and exponential control inputs.

6. What legal troubles did Moog and ARP have, and how were they resolved?

7. Tell how to create tremolo.

8. Tell how the VCA will usually be used, and how one might go about setting it. Be sure to include the settings and connections for the EG's.

9. List all of the parameters of the VCA, and tell which ones can be modulated.

10. Tell what is normalled to each of the VCA's inputs and where the VCA's output is normalled.

11. Describe the correct envelope settings for the following instruments: Drum, Organ, Piano, Voice (slow crescendo).

Terms To Know:

Amplitude
dB
Decibel
Distortion
Exponential
Gain
Gate
Initial Gain
Linear
Linear to Exponential Converter
Tremolo
VCA
Volume
Voltage Controlled Amplifier

SECTION

9

Mixer Section

The Mixer

Figure 9-1: The mixer section

The mixer section is a module of the synthesizer that has been used almost since the beginning of this book, but has yet to be explained fully. The mixer section is one of the most simple modules, and easiest modules to understand. The mixer performs a simple task: it combines two different signals and prepares them for output to the ARP 2600's internal amplifier. One can see in Figure 9-1 that the mixer features two audio inputs. Each one features an attenuation slider so that the signals can be combined at the desired levels. Note that the outputs of the VCF and VCA are normalled to the mixer's inputs. This makes sense, since these are the two modules most likely to be at the end of the signal chain of a patch (not counting the mixer section, which has to be at the end of every patch if one wants to hear any sound at all).

The mixer's signal flow is easy to see by following the white lines screened on the ARP's front panel. One can see in Figure 9-1 that first, each input leads to a separate jack. Unfortunately, there is no mention of the function of these jacks anywhere in the ARP 2600's user manual. (A copy of the author's user manual is available on-line at http://www.EmusicDIY.com/arp/pages.) These jacks aren't labeled, which makes it rather difficult to discover their purpose. Careful inspection of the 2600's schematics reveal that these jacks are individual outputs. These jacks pass a copy of whatever comes in each input back out. However, when a plug is inserted into them, they cut off the signal flowing to the mixer and then to the amplifier and speakers. One might think that this would render them completely useless, but this is not really so. They allow either of the sliders on the mixer to be used to attenuate audio or control signals independently of any other module.

It is interesting to note that while the mixer has been used exclusively for audio signals to this point, it is possible to use it to mix control signals as well. This technique is of rather dubious value, however, since the mixer's ability to mix and attenuate incoming audio signals is lost when it is used with control voltages. There are modules better suited to this task which will be explored in the next section.

Panning

In the next part of the mixer, the white lines converge and meet at one output. This is where the actual mixing occurs. Whatever is input to the mixer's two channels appears here at this single output. However, this jack is more than an output. It is also an input. Incoming signals are fed to the next part of the mixer: the PAN control. *Pan* is an abbreviation of the term *panoramic potentiometer*. The PAN control simply determines the amount of signal which is going to each output, and to each speaker. When the

PAN slider is moved to the left, a greater amount of signal will come out the left speaker than the right. Some synthesizers offer voltage controlled panning, but alas, the ARP 2600 does not. A pan control can be found on most mixers, from the most humble two channel mixers like the one on the ARP to dedicated mixing consoles which are six to twelve feet long. The PAN slider can be seen more clearly in Figure 9-2.

MAIN INPUTS AND OUTPUTS

The mixer has four more jacks. Two of these jacks (the LEFT INPUT and RIGHT INPUT) are considered the *main inputs* on the ARP 2600. These jacks bypass the PAN slider and are fed directly to the internal mixer. It is important to understand that any signal being fed into these inputs will not be attenuated at all, and will reach the amplifier at full strength. This can result in some rather loud sounds, depending upon where the speaker controls are set. Generally, it is a much wiser practice to patch sounds into the mixer first so that their level can be attenuated to a more desirable level. The main inputs and outputs can be seen in Figure 9-2. The main inputs could be used to connect a CD player or a tape deck.

Figure 9-2: The pan slider and the top of the mixer section

The other two jacks are the *main outputs*. These jacks output a copy of the signal being fed to the ARP's internal amplifier. The output jacks are used when one wants to connect the ARP 2600 to a large amplification system, a tape recorder, or as in the case of the CD which accompanies this book, a computer. The input/output jack which is just under the pan slider is normalled to another device which exists within the mixer section: The reverberator.

REFLECTIONS AND ECHOES

One of the most basic elements of music is the sound of sound waves bouncing off of walls or other surfaces and returning to the listener at a slightly different time than the sound waves coming directly from the instrument to the listener. When sound bounces off of a hard surface and returns to a listener, it is called an *echo* or *reflection*. Human ears are not very sensitive to echoes, and the echo must return to the listener some time after the original sound is heard, or else the listener won't hear the echo and the original sound separately. One can experience this effect by backing twenty to thirty paces away from a house, and clapping one's hands loudly. Just after the sound of the hands clapping is heard, another distinct clap can be heard. This is because the sound waves traveling directly from one's hands have a very small distance to go to the ears. The sound waves which went away from the body had to go all the way to the house, bounce off the wall and then return to be heard. Of course, the surface that the sound waves bounce off has a lot to do with how much sound will return. While sound bounces fairly well off of wood and brick, sound does not bounce well off of carpet and fabric. Hence, adding things like curtains, furniture and carpeting to a room will greatly reduce the amount of echo in the room.

Imagine a situation in which one is in an empty room which has little or no resistance to sound waves bouncing around (e.g. no carpeting, draperies, furniture, etc.). In such a room, sound waves will bounce off just about every surface. A gymnasium is a perfect example of such a room.

Reverberation

When a sound is produced, it travels outwards in every direction at once. If Bob claps his hands, the sound waves will hit the floor first. Bob won't hear the echo from the floor as it is too close to him and as such, he can't perceive that echo as a separate signal. However, the walls and ceiling may be far enough from Bob so that he can hear the echo from each of them. However, since the walls and the ceiling are not the same distance from Bob, the echoes will come back at different times. Recalling that human ears aren't very sensitive to picking up different sounds which occur very close to each other in time, instead of hearing lots of individual echoes, Bob will hear *reverberation*. Reverberation or *reverb* is a phenomenon which occurs when many separate echoes come back to a listener so quickly that he or she can no longer hear them as individual sounds. The resulting wash of sound is reverb. A common example of reverb is the sound a basketball makes when it is bounced in an empty gymnasium.

Reverb occurs naturally in large concert halls, but it is desirable to add it electronically to electronic signals as it gives them a more natural release and makes them sound more real, as though they occurred in a natural space. To help create this natural sound, the ARP company installed a *reverberator* in the mixer section. Modern synthesizers come with many fancy sorts of effects, but they almost always include reverb. An inventor and recording technology pioneer from Waukesha, Wisconsin named *Les Paul* discovered that when audio signals are passed through enclosed springs, they create an effect which is very similar to natural reverb. The part of the 2600 that actually has these springs is called the *reverb tank*. The reverb tank is bolted to the inside of the 2600's cabinet on the bottom left hand side. It is a brass-like sealed metal box with an input and two outputs. If the springs in the reverb tank get jostled (it is possible to do this by gently thumping the top of the ARP's cabinet or the table on which it is sitting) the springs will clang together and make some rather cacophonous noises. As long as the thumping is done gently, no harm will be done to the springs. **CD track 49**

The output of the mixer is normalled to the input on the reverberator. The reverberator has an output of its own, but it is also normalled to the ARP 2600's main outputs, and thus to the speakers. The reverberator has rather simple controls, which control the amount of signal returning from the reverberator. These can be seen in Figure 9-1 on page 70. In this way, the amount of reverberation can be controlled. There are two controls, one for the reverb returning to the right speaker, and one for the reverb returning to the left speaker. Generally, these levels will be set equally. If the reverb level is set too high, sounds coming in will have a rather watery effect to them, **CD track 50** which is generally considered to be undesirable. The wonderful thing about the reverberator is that since each of the springs in the reverb tank will react slightly differently to incoming signals, the two signals coming out of the reverb tank are a bit different from each other. When they are connected separately to each speaker, the formerly mono signal is now close to stereo. (True stereo is a bit different, but this is about as close as the ARP 2600 comes.) The bad thing about the reverb tank is that the ARP's designers didn't set the input levels as well as they could have. As a result, the reverberator tends to add a lot of unwanted noise to signals when it is used.

Experiments for Section Nine:

1. Could a mixer be helpful on a larger scale (i.e. more inputs) in the studio in general?

2. Try moving a signal from side to side using the panning controller. How could this be useful on a larger scale?

3. Locate the mixer post input/output jack. How is this helpful at times?

4. Locate the 2600's main audio outputs. When might these be used?

5. Locate the 2600's main inputs. When might these be used?

6. Locate the mixer's direct outputs. How are these useful?

7. Patch VCO-1 to the reverberator's input. Practice using different amounts of reverb. **CD track 47**

8. Add reverb to a signal coming into the mixer such as the noise generator gated by the VCF. How and why is this more effective than the results of #7? Try using diferent amounts of reverb. **CD track 48** Try adding too much reverb and listen for the watery effect. (Also listen for added noise) **CD track 50**

9. Determine how the reverberator is connected to the mixer by tracing the connecting lines on the front panel.

10. What is the electronic device which creates reverberation in the ARP 2600?

11. Try jostling the springs in the reverb tank by thumping gently on the top of the cabinet or on the bottom of the table on which the cabinet sits. **CD track 49**

Review Questions for Section Nine:

1. What does the mixer do, and how many inputs does it have?

2. What kinds of signals can the mixer mix?

3. When would the mixer's main outputs be used? How could this task be accomplished without using the main output jacks?

4. How does reverberation occur in natural spaces?

5. How does the ARP 2600 produce reverberation? Who invented this device?

6. How does one use the direct outputs just above the mixer's two sliders?

7. How is the reverberator connected to the mixer?

8. What does the panning control do? Why is this useful?

9. Why isn't it particularly useful to patch into the ARP's main inputs?

Terms and Names To Know:

Echo
Les Paul
Main Inputs
Main Outputs
Mixer
Pan
Panoramic Potentiometer
Reflection
Reverb
Reverb Tank
Reverberation
Reverberator

SECTION 10 S/H MODULE

INTRODUCTION

The sample-and-hold module is actually composed of three discreet modules, each of which can be used on their own. It does make sense that they have been grouped together, however, since both the sample-and-hold module and the electronic switch are dependent upon the internal clock for operation.

THE INTERNAL CLOCK

The *internal clock* is different from just about every other module on the 2600. It is almost never heard, nor was it intended to be heard. The internal clock performs a fairly simple task: It produces a constant stream of trigger pulses, one after the next as shown in Figure 10-1. The drawing on the front panel of the cabinet also depicts these constant trigger pulses, and can be seen in Figure 10-2.

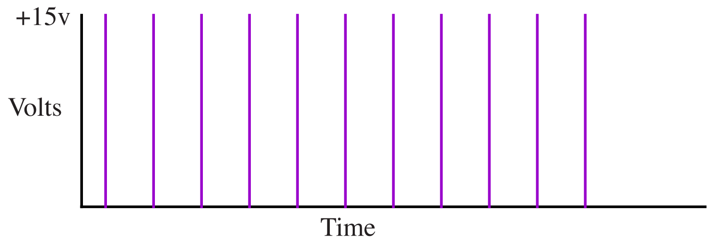


Figure 10-1: The trigger pulses output by the internal clock

The internal clock has only one parameter, and that is *rate*. Rate determines how quickly the internal clock puts out pulses. The rate at which the internal clock puts out trigger pulses can be changed by moving the RATE slider shown in Figure 10-2 up or down. Again, this is the internal clock's one and only parameter.

THE INTERNAL CLOCK'S NORMALS

The internal clock is normalled to three other modules, but unlike the normals that have been examined to this point in this text, one of these normals cannot be broken. First, the internal clock is normalled to the sample-and-hold circuit. This connection will be discussed in detail in a moment. Secondly, the internal clock is normalled to the electronic switch. This normal cannot be broken. There are a few ways to work around this problem which will be discussed later in this section. Finally, the internal clock is normalled to the envelope generators. This normal can be broken, but there are two ways to do this, both of which will be discussed in a moment.

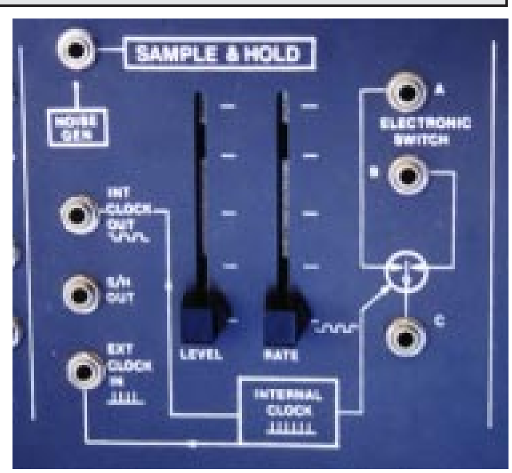


Figure 10-2: The S/H Module

Although it is not labeled to this effect, the EXT CLOCK IN jack is where the clock is normalled to the sample-and-hold unit, and it is here that the internal clock's normal to the sample-and-hold unit can be broken. A pulse wave connected here will trigger the sample-and-hold unit and cause it to sample. However, the incoming pulse wave will not affect the electronic switch, because the internal clock cannot be controlled by an external device. This will be discussed in depth in a moment.

Clock In, Clock Out

There are two jacks associated with the internal clock, one of which is an input, and one of which is an output. A signal from another clock (or an oscillator) in the form of a pulse wave or trigger pulse can be connected to the EXT CLOCK IN jack, and the normal between the internal clock and the sample-and-hold unit is then broken. When an external trigger signal is connected to this jack, the incoming signal will replace the timing pulses to the sample-and-hold unit which was formerly controlled by the internal clock. However, the timing pulses just affect the sample-and-hold unit, since it is not possible to synchronize the internal clock to an external source. The lines drawn on the front panel (see Figure 10-2 on page 75) seem to indicate that a signal connected to the EXT CLOCK IN jack would go to the clock and cause it to follow the incoming source. Sadly, this is not the case. Thus, if one wants to synchronize the internal clock and an external device, the internal clock must be allowed to control the timing of the external device, since the external device can't control the timing of the internal clock.

The second jack associated with the clock is the clock output jack. One might think that since the internal clock generates a series of trigger pulses that a series of trigger pulses would be output here. This is not the case, however. The internal clock actually drives a small oscillator which puts out a square wave synchronized with the trigger pulses the clock produces. It is this square wave which is output at the INT CLOCK OUT jack. It is possible to connect this output to the mixer and hear a square wave if the internal clock's rate is set in the audio range, but since the frequency of the internal clock is not voltage controllable, this is really not particularly useful for anything other than testing the module.

S/H Gate Switch

One may recall that at the end of Section 7, a small switch and a corresponding jack were noted just below the EGs. To date, all of the experiments have made use of this switch in the upper position where the keyboard's trigger and gate signals are normalled to the EGs and cause them to fire when a note is played. However, it is possible to use the internal clock to cause the EGs to fire. All one needs to do is to move the switch to the lower position. In the lower position, the keyboard's trigger and gate normals to the EGs are broken, and the internal clock is normalled to the EGs. It will send trigger pulses to them and cause them to fire at its specified rate. The internal clock's normal to the EGs can thus be broken by returning this switch to the upper position. This opens the door to many patches which seem to play themselves, and are rather automatic in nature.

The jack which is below this switch is indicated as being the normal from the "S/H Gate" which is rather misleading. It seems that the 2600's designers considered the electronic switch, the internal clock and the sample-and-hold units all part of the "Sample-and-Hold module." Hence the label on the jack below the EG's. However, it is really the internal clock which is responsible for causing the EGs to fire, not the sample-and-hold unit.

As one would expect, the normal from the internal clock to the EGs can be broken by inserting a plug into the jack just below the switch. This will allow some marvelous possibilities later on when other devices are used to control the 2600. When this jack receives a gate type voltage from something such as a pulse wave or an external clock, it will generate the appropriate gate and trigger signals to cause the EGs to fire.

It is also possible to trigger the EGs using the gate and trigger in jacks. The use of these jacks will be discussed briefly in Section 15 when the ARP sequencer is introduced. However, unless the 2600 is going to be interfaced with other equipment, these jacks are not used nearly as much as an FM input on a VCO or even the output on the filter.

The Electronic Switch

The *electronic switch* is one of the most unique and oft forgotten parts of the 2600. It consists entirely of three jacks labeled A, B, and C. The electronic switch alternately connects jack C to jack A and jack B. When A is connected to C, B is disconnected from C, and vise versa.

The internal clock is permanently normalled to the electronic switch, and determines the electronic switch's rate of switching. One may recall that the internal clock was described as putting out a series of trigger pulses which drive a square wave oscillator, and it is this square wave which drives the electronic switch. The 2600's designers thoughtfully indicated this with a rather long square wave next to the line indicating the internal clock's normal to the electronic switch. (See Figure 10-2 on page 75.)

It is interesting to note that the electronic switch's jacks are neither specifically inputs nor outputs. For instance, one could connect the output of an oscillator to jack A, and it would come out jack C when the electronic switch connected the two. One could also connect the output of an oscillator to jack C, and the signal would alternate between coming out jack A and jack B. Thus, the electronic switch's jacks are either inputs or outputs, depending upon what is connected to them.

One may recall that by connecting a pulse wave to the EXT CLOCK IN jack, the rate at which the sample-and-hold unit sampled could be determined by an external source. Again, the internal clock's normal to the electronic switch cannot be broken, and so the internal clock will always determine the rate of switching. This is both useful and unfortunate. While it is wonderful to be able to make the sample-and-hold unit sample at a rate which is independent of the electronic switch, it is unfortunate that they cannot both be synchronized to an external source using the EXT CLOCK IN jack. Of course, if the external source can be synchronized to the internal clock, this problem can be solved.

The Electronic Switch in Practice

There are hundreds of potential uses for the electronic switch, only a few of which are presented here. They basically fall into one of two categories: patches which use *distribution* and patches which use *source switching*.

In the basic distribution patch, jack C is an input, and the incoming signal is alternately distributed to jack A and jack B. One unique possibility with this configuration is a panning patch. If a sound source such as an oscillator (or the VCF's output, for that matter) is connected to jack C, and jacks A and B are connected to the LEFT INPUT and RIGHT INPUT jacks in the mixer section, the sound coming into jack C will be switched between the left and right speakers. This can be heard on **CD track 51**.

Yet another wonderful possibility is to connect a control source, such as an LFO to jack C, and then connect jacks A and B to the FM inputs on two different oscillators. When the outputs of these oscillators are brought up in the filter and sent to the mixer, the two will be alternately modulated. **CD track 52**

Another favorite technique is to connect the output of the last module in a patch to jack C, and connect one of the two remaining jacks to the mixer. In this configuration, the switch will switch between the patch and silence which creates a wonderful pulsing sound. When combined with a resonant filter sweep, this creates a sound which is very popular in today's dance music. This sound can be heard on **CD track 53**.

Switching patches are also very interesting and useful. It is possible to connect two different oscillators, perhaps tuned together but producing different timbres to jack A and B, and the electronic switch will switch between the two. It is even possible to switch between two different waveform outputs of one oscillator in this way.

Two sound sources tuned to different pitches can also be connected to jacks A and B, and the electronic switch will switch between the two. The switch could also be used to switch between tuned oscillators and the noise generator's output. Examples of these techniques can be heard on **CD track 54**. Alternately, two different versions of the same sound (filtered and unfiltered for example) could be connected to jacks A and B and the electronic switch will switch between the two.

Again, the electronic switch is not just for audio signals. It can also be used to switch between different control signals as well. On the most basic level, the electronic switch could be used to flip between two different waveforms from an LFO in an FM patch. Or, the switch could be used to flip between two different LFO's modulating one VCO. Perhaps one LFO could be in the audio range while the other is in the sub audio range. Examples of this technique can be heard on **CD track 55**.

The Sample-and-Hold Unit

The *sample-and-hold* unit (often abbreviated *S/H*) employs a series of fairly simple concepts to generate a useful control voltage. There are several steps to its operation, the first of which is sampling.

Sampling basically means taking a measurement. The S/H module samples voltage coming into the S/H input which is the top most jack. In Figure 10-2 on page 75, one can see that the noise generator's output is normalled to it, and the label to the right side says SAMPLE & HOLD. Like most of the ARP's inputs, there is a slider which allows the user to attenuate the level of signal coming into this jack. The slider labeled LEVEL attenuates the incoming signal.

If a slow-moving saw wave is sampled, the S/H module might take samples every half second or so. The point at which the S/H module sampled is indicated in purple in Figure 10-3 on page 79. Notice that the module takes samples at evenly spaced intervals. In this example, the S/H unit might take the following readings of incoming voltages for the saw wave in Figure 10-3: 0.6 volts, 1.4 volts, 2.5 volts, 3.6 volts, 4.4 volts, 5.6 volts, 6.4 volts, 7.5 volts, 8.6 volts, 9.4 volts, and finally 9.8 volts at the top of the saw wave.

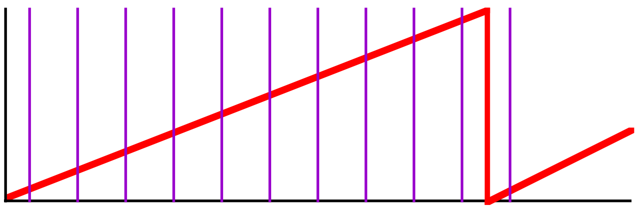


Figure 10-3: The S/H unit samples a saw wave

It is important to note that when the S/H unit isn't sampling, it simply ignores incoming voltages. Of course, one begins to wonder just how the S/H module knows when to sample. It would take a device which constantly puts out a stream of pulses in even intervals, and this is the reason that the internal clock is built into the S/H module. The internal clock's rate determines the rate at which the S/H module samples incoming voltages. By adjusting the internal clock's rate, one can take more samples or fewer samples of an incoming voltage.

Now that sampling has been established, it is time to look at what the S/H module does with the voltage it samples. The sample-and-hold module is so named because after it samples an incoming voltage, it holds onto that value, and continually puts that voltage out its output. This voltage is almost always used as a control signal, since it does not usually fluctuate rapidly enough to be heard. (One can use an external signal to make the S/H unit sample extremely rapidly, which will cause it to output an audible signal.) For instance, in Figure 10-4, one can see how the S/H module puts out a stepped voltage (represented in blue) for the incoming control voltage (represented in red). Each time the S/H unit samples (represented in purple) a new voltage value is read and held until the next time the unit samples.

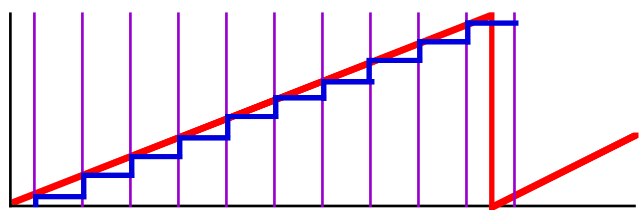


Figure 10-4: The samples taken by the S/H unit

The S/H Unit In Practice

The most common application of the S/H unit is to sample a random source (the noise generator) and use its output to FM the VCOs. This generates random frequencies which jump all over the spectrum. This effect has been used for countless "computer" sound effects. **CD track 56** Another great possibility is to use the S/H's output to control the Fc on the Filter. This gives a rhythmic, stepped effect to the sound which can be highly pleasing. **CD track 57** Yet another wonderful possibility is to use the S/H unit to FM the VCOs, but to sample a slow moving LFO and time the sampling with the rate of the LFO so that the VCO's pitch will rise or fall in half steps or whole steps. **CD track 58**

For unpredictable results, the output of an entire patch can be sampled and then used to FM all of the oscillators. In this way, a sort of feedback loop is being created within the 2600. For instance, the VCOs feed the VCF which feeds the VCA,and the VCA is routed to the S/H unit and sampled. Then, the output of the S/H unit is routed back to each VCO's FM input. This can yield surprising and often whacky sound effects. **CD track 59** As with any module on the 2600, the key to mastering it is experimenting with it every way possible.

Experiments for Section Ten:

1. Connect the INT CLOCK OUT jack to an input on the mixer. Try to increase the clock's rate high enough that its square wave output can be heard.

2. Flip the S/H gate switch below the EG's to the lowest position so the internal clock will trigger the EG's. Create a patch in which the VCO-1 and 2 are tuned in unison and fed to the VCF. Use the ADSR generator to modulate the VCF's Fc. Try changing the clock's rate, and try changing each of the stages of the EG.

3. While conducting experiment #2, use a pulse wave from VCO-3 in LF mode to trigger the EG's by connecting it to the S/H GATE jack. What happens? Why does this happen?

4. Create a patch using all three oscillators tuned in unison and routed to the filter. Add 50% resonance, and close the filter. Connect the VCF's output to jack A on the electronic switch. Connect jack C on the electronic switch to the mixer and raise that mixer input's level. Now sweep the filter's Fc up and down to create a pulsing sound with a filter sweep. **CD track 53**

5. Create a patch using all three oscillators tuned in unison and routed to the filter. Add 50% resonance, and close the filter. Connect the VCF's output to jack C on the electronic switch. (Decrease the volume of the speakers before moving on to the next step.) Connect jacks A and B to the LEFT INPUT and RIGHT INPUT jacks. What is happening and why? **CD track 51**

6. Tune two oscillators to different pitches and connect an output from each to jacks A and B on the electronic switch. Connect jack C either to an input on the filter or directly to the mixer. **CD track 54**

7. Connect two different control signals to jacks A and B of the electronic switch. Now connect jack C to the FM input on a VCO. **CD track 55**

8. Connect the pulse output of VCO-3 to the EXT CLOCK IN jack while conducting experiment #5, and notice that it has no effect on this patch. Why is this?

9. Use the S/H to sample the noise generator (be sure to raise the level on the noise generator). Use the S/H output to FM VCO-1 and VCO-2. Raise the clock rate slider about halfway, route the VCO's to the filter, then the mixer and add a little reverberation. **CD track 56**

10. Now use the S/H output to control the Fc of the VCF. **CD track 57**

11. Create a few different 'feedback' patches. Because there are so many variables in this patch, your results may sound nothing like those on the CD. **CD track 59**

Review Questions for Section Ten:

1. What is the internal clock's only parameter?

2. When and where does the internal clock put out trigger pulses and/or square waves?

3. What three things does the internal clock control?

4. Which of the internal clock's normals can and can't be broken?

5. Describe how the internal clock can control the EG's.

6. Describe how to synchronize the internal clock with an external source.

7. Name the two main kinds of patches the electronic switch can be used for, and give examples of how each could be useful.

8. Step by step, describe the process by which the S/H unit samples incoming voltage.

9. Name three ways the S/H unit can be used.

10. How is the level of voltage being input to the S/H unit attenuated?

11. Why are the electronic switch, the internal clock, and the S/H unit grouped together on the ARP's cabinet?

Terms To Know:

(Clock) Rate
Distribution Patch
Electronic Switch
Feedback Patch
Internal Clock
Sample
Sample-and-Hold
S/H
Source Switching Patch

SECTION 11

Preamp Et Al

The Preamp

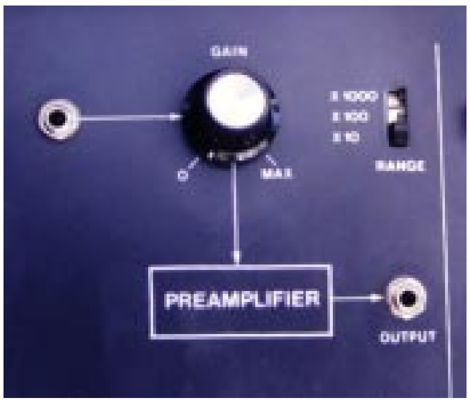


Figure 11-1: The ARP 2600's preamp

To this point, all of the experiments and examples have used the modules contained in the ARP exclusively. While this is a wonderful way to learn about the ARP, one must understand that the 2600 is most powerful and useful when used with other devices in a studio. Unfortunately, connecting devices directly to the ARP's modules usually doesn't work particularly well, since the signal coming from these devices is much weaker than the signals the ARP uses. Before signals from devices such as CD players, tape decks, or other synthesizers can be used, they must be *amplified*. Amplified means 'made louder,' which means increasing the height of the waveforms. This job is left to an *amplifier*.

The amplifier, found in the upper left hand corner of the ARP's cabinet (see Figure 11-1) is referred to as a *preamplifier* because it amplifies signals before they go to other modules. The job of a preamplifier is to raise the level of a signal to match a specific level.

The preamplifier's input is not labeled, but it is the left most jack on the module. Unfortunately, the ARP 2600's designers chose to use an 1/8" jack here. While this conforms to the jacks on the rest of the instrument, this would have been a very good place to put a 1/4" jack, since most external equipment that one might want to connect to the ARP has 1/4" jacks.

Following the white line which indicates the patch of the signal through the module, one can see that the gain parameter is the next item encountered. *Gain* is another word for volume. Although all of the other controls on the 2600 have sliders, the preamp features a rotary knob. The farther clockwise this knob is turned, the greater the amplification of an incoming signal. In Figure 11-2, one can see a square wave both before and after amplification.

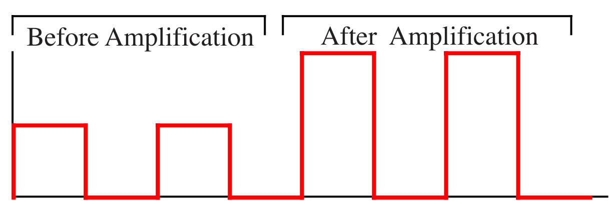


Figure 11-2: A square wave before and after amplification

In its default mode, the preamp can increase the height of a waveform up to ten times its original height (when the gain knob is set to MAX). While this might seem like a lot of amplification, it really isn't. There are times when more is needed. Thus, the preamp allows the user to set the range of values over which the gain knob will function. This is set using the switch labeled RANGE. This switch has three possible settings: 10x, 100x, and 1000x. When set to 10x, the preamp will increase the waveform's height tenfold when the gain knob is in the MAX position. When the switch is set to 100x, the preamp will increase the waveform's height one hundredfold when the gain knob is in the MAX position and so on with 1000x. Of course, 1000x is a great deal of amplification, and there are limits to what the preamp circuit can handle.

Distortion

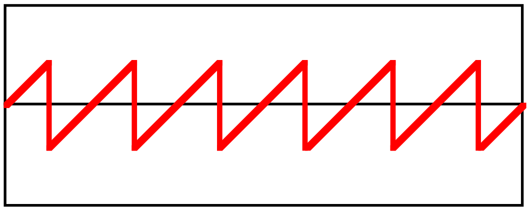


Figure 11-3: A saw wave within the preamp's dynamic range

The preamp has a range of amplitudes of signals that it can handle. As long as signals stay within this range, the preamp will faithfully amplify signals, and put out what comes in, only louder. (See Figure 11-3)

The distance from the top of the black rectangle in Figure 11-3 to the bottom represents the preamplifier's *dynamic range* or the range of amplitudes which the preamp can accurately reproduce. However, it is entirely possible to amplify a signal to the point where the peaks and troughs of the waveform reach outside the dynamic range. (See Figure 11-4)

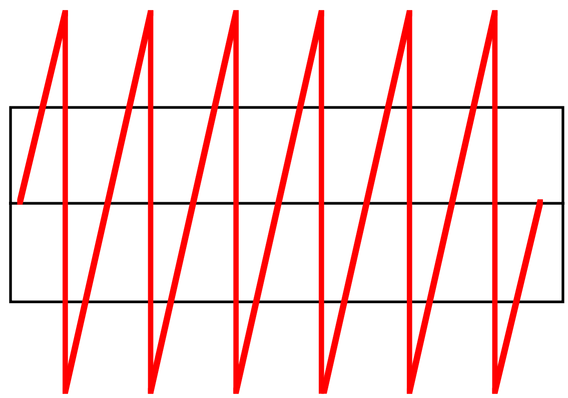


Figure 11-4: A saw wave which has exceeded the preamp circuit's dynamic range

The preamp cannot handle the highest points of these waveforms, and they become clipped off when they reach the end of the dynamic range. This phenomenon is known as *clipping* or *distortion*. The word distortion has many uses, but one must think of it as the actual shape of the waveform being changed or distorted, similarly to the way a funhouse mirror distorts the image of one's face. When the waveform in Figure 11-4 emerges from the preamp, it will look like Figure 11-5.

Because the actual shape of the wave has changed, so has the harmonic content, and thus the waveform's timbre. This waveform which was formerly a saw wave will sound more like a square wave. In fact, as the amplitude is increased, the waveform will become more like a square wave. In this sense, the preamplifier can actually be used to reshape the incoming waveform and turn it into a different wave.

Distortion: Friend or Foe?

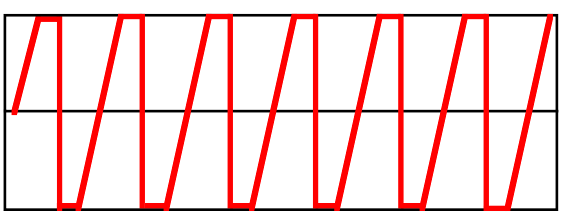


Figure 11-5: A clipped saw wave

To this point, nothing has been said about whether distortion is a good thing or a bad thing. For many years, distortion was considered to be a very bad thing. Distortion in a recording was to be avoided at all costs. The faintest crackle was considered to be the sign of a poor recording. During the late 1950's and early 1960's, guitar players discovered that if they increased the volume of their amplifiers enough, distortion would occur and change the timbre of their guitars. Distortion thus became a popular effect for guitars. In the 1990's, artists such as Skinny Puppy and Trent Reznor of Nine Inch Nails have taken distortion to a new level, distorting everything from their voices to all of the musical instruments in a song.

Perhaps the most important thing to learn about distortion is not so much if it is a good thing or a bad thing, but rather when it is appropriate and when it is not. For instance, if one is trying to make a recording of Beethoven's fifth symphony, distortion is probably a bad thing. If one is trying to create hard-core or industrial music, distortion is probably a good thing. The presence or absence of distortion is often times determined by the genre of music being produced.

The Preamplifier in Practice

The preamplifier is generally used when interfacing the ARP 2600 with other equipment. Specifically, the preamp is used to bring the output levels of other devices up to the level required by the 2600. This opens the door to hundreds of new possibilities, far too numerous to list here. A few possibilities include: connecting a microphone for adding distortion, filtering with the VCF, or shaping with the VCA. One could also connect other synthesizers to make use of the ARP's filter and/or VCA. One could feed the ARP's own signals (either control or audio) into the preamp for amplification and/or distortion. Audio from a CD player could be input for distortion or filtering (this works particularly well with drum loops!) While these are just a few ideas, the important thing to understand is that most external equipment can be connected to the ARP using the preamplifier.

The Envelope Follower

Nothing is normalled to the preamplifier's input, but the preamp's output is normalled to something: the input of the envelope follower. The envelope follower is located on the left most position on the cabinet. Like the preamp, the envelope follower performs a fairly straightforward job. Looking at the module (Figure 11-6) one can see that it has a single input, an output, and a single fader.

Figure 11-6: The envelope follower

The *envelope follower* turns incoming audio-range waveforms into a steady control voltage up to +10 volts which can then be sent to other modules. For instance, if one connected a sawtooth waveform from an oscillator, the envelope follower analyzes the amplitude of the incoming waveform. In Figure 11-6, the saw wave coming into the envelope follower's input is shown in red, while the output is shown in purple. The amplitude of the incoming waveform can be attenuated using the single fader on the envelope follower.

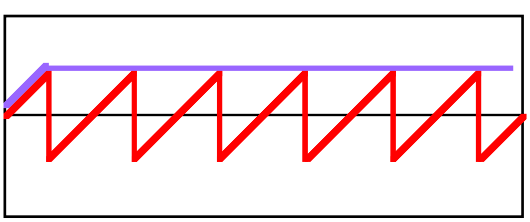


Figure 11-7: The output of the envelope follower

The envelope follower is used to create an envelope which matches the volume envelope of the incoming waveform. Thus, one must already have changes in amplitude if one wants to make effective use of the envelope follower. Most of the time, it is used with external devices, hence the fact that the preamp's output is normalled to its input. One way the envelope follower is commonly used is to track the amplitude of a signal coming into a microphone. As a person speaks more loudly into the microphone, the more control voltage exits the envelope follower. It is important to remember that the envelope follower is not able to sense changes in frequency, and thus, changes in pitch will have no effect on the signal it puts out. It will only detect and react to the volume of the incoming signal. The envelope follower can be used in place of an EG whenever an external signal is available for use. The envelope follower can thus be used to modulate the VCO's, VCF, or VCA in a fashion very similar to the way the EG's modulate them. **CD track 60**

The Ring Modulator

The ring modulator is the most complex of the three modules presented in this section. Many modern synthesizers claim to have a ring modulator when they actually have a *balanced modulator*. The difference between the two is actually in the design and construction of the circuit. This point is beyond the scope of this book, but it is important to understand that they are really two different modules. Many synthesizers today have modules which produce ring modulator-like effects and claim to have ring modulators, when the truth is that these modules are not ring modulators at all. Synthesizer companies realize that musicians will be more comfortable working with something which is familiar, and thus continue to use the term "ring modulator."

Figure 11-8: The ring modulator

The ARP 2600's manual gives a succinct, but mostly useless definition of the way the ring modulator works: "The ring modulator is essentially a voltage multiplier; from two inputs A and B it produces the output function A x B/5." The ring modulator is much easier to understand when the kind of modulation it allows is understood. The ring modulator allows a new kind of modulation which has not been experienced up to this point. *Amplitude Modulation* or *AM* is a process in which the amplitude of one waveform is used to modulate the amplitude of a second waveform.

How Does the Ring Modulator Work?

Essentially, many new harmonics are added to a sound, and the original sound is then removed from the signal using cancellation. To accomplish this, two different signals are input to the ring modulator's inputs which can be seen in Figure 11-8. Each incoming waveform has its own *frequency content*. (Frequency content is defined as all of the harmonics of a particular waveform. This is also sometimes referred to as *harmonic content*.) The ring modulator adds the entire harmonic content of the two waveforms, and subtracts the harmonic content of the two waves. The ring modulator comes up with a sum and a difference of these two incoming waves. For instance, a 210 Hz fundamental would be added to a 441 Hz fundamental to give a 651 Hz fundamental. Similarly, an 255 Hz fundamental could be subtracted from an 880 Hz harmonic to give a 625 Hz harmonic. The ring modulator causes *every* harmonic of one incoming waveform (including the fundamental) to be added to *every* harmonic (including the fundamental) of the second incoming waveform and *every* harmonic of one incoming waveform to be subtracted from *every* harmonic of the second incoming waveform. The results are difficult to predict.

Although the ring modulator is incredibly complex, some generalizations can be made about the kinds of sounds which can be produced with it. Metallic instruments naturally have a very dense harmonic content, and as such, the ring modulator does an amiable job of imitating them. **CD track 61** It is also good at accentuating the high frequencies of different timbres, and can be a highly effective addition to a patch. **CD track 62** Another interesting effect can be achieved by using one VCO in LF mode while the other is in audio mode. If a square wave from the LFO is used, a pleasant pulsing effect can be created. **CD track 63**

The ring modulator's output is normalled to both the VCF and the VCA. While its presence in the VCF is no surprise, the fact that it is normalled to the VCA is a little unusual. Certainly the VCF should be normalled to the VCA, since that is the normal signal flow in the most common synthesizer patch. However, out of all of the modules present on the 2600, one is left to wonder why the ARP's designer's chose to normal the ring modulator to this input. Perhaps it was chosen by default; the VCF was already normalled to the VCA, and choosing a VCO would mean selecting one above the remaining two. Other than the ring modulator, there are few other sound-producing modules. One can almost imagine the 2600's designers pondering a choice between the noise generator and the ring modulator...

The DC/Audio Switch

Near the bottom of the ring modulator, between its inputs lies a switch labeled AUDIO and DC. This switch determines whether the ring modulator's inputs are AC or DC coupled. It is not as important to understand the issues of AC and DC coupling as to understand the changes each setting brings about in the ring modulator's operation.

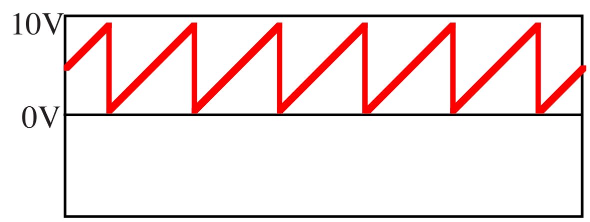


Figure 11-9: a standard incoming saw wave

When the ring modulator is AC coupled (the switch is in the AUDIO position) any direct current (DC) in the signals is cancelled out before it reaches the modulator circuit. Another way to say this is that the incoming waveform's voltage offset will be raised or lowered so that the waveform's average height is zero volts. In Figure 11-9, one can see a saw wave as it would be produced by VCO-1. When the saw wave hits the AC coupling circuit, it is changed to look like Figure 11-10. Its average is now 0 volts instead of +5 volts.

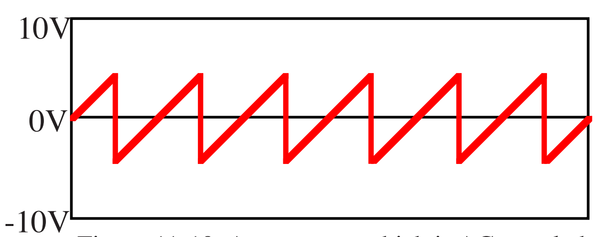


Figure 11-10: A saw wave which is AC coupled

If the switch is set to the DC position, the circuit is *DC coupled* and waveforms are passed into the ring modulation circuit 'as is.' The change in sound between these two settings is difficult to describe, but it would be generally accurate to say that sounds using the DC position are more clangorous and have an even greater level of complexity than those generated in the AUDIO position. Listen to a few sounds which start with the switch in the audio position, then move it to the DC position. **CD track 64**

THE RING MODULATOR IN PRACTICE

The two inputs of the ring modulator are dependent upon each other. This means that attenuating one of the two inputs would cause a reduction in the overall volume being output by the ring modulator. Both sliders on the ring modulator are generally set full open for this reason. One may wish to decrease the level of one input if the source is external (e.g. a guitar coming through the preamp) and it is louder than the other incoming signal.

The output of the ring modulator is sometimes used alone, but more often than not, the ring modulator's output will be added to patches using the VCO's, noise generator, etc. There are times when it is desirable to use its output alone, however. This is most effective when the ring modulator is generating lots of harmonics. Because of the way it creates harmonics, the ring modulator will generate more harmonics if the two incoming waveforms aren't perfectly in tune with each other. In fact, the farther out of tune they are, they more harmonics it will produce. This can be heard by raising both of the ring modulator's sliders (its inputs are normalled with the outputs of VCO-1 and 2) and then sweeping the frequency of one of the oscillators. **CD track 65**

One should not forget the possibility of using the ring modulator on external signals. Guitars, in particular, sound wonderful when processed with an oscillator. In fact, some companies sell ring modulators just for the purpose of being used with a guitar. Any signal is really fair game for use with the ring modulator, though. Again, experimentation is the key to mastering this module's potential.

Experiments for Section Eleven:

1. Start by closing both of the mixer inputs completely. Connect a microphone to the preamp's input, making sure the gain is fully closed, and the preamp's range is set to 10x. Connect the output of the preamp to an input on the VCF. Fully open the slider above this input. Open the filter completely. Open the filter's input on the mixer to the first mark. SLOWLY increase the level of the preamp's gain while talking loudly into the microphone. When the microphone reaches a comfortable level, stop increasing the gain. One may find that depending upon the microphone being used, 10x amplification is not sufficient for this experiment. If the preamplifier's range needs to be increased, turn the preamplifier's gain all the way down before switching to 100x range. While speaking into the microphone, try changing the filter's Fc. Now add a small amount of resonance. While conducting this experiment, be careful not to point the microphone at the cabinet's speakers or get too near the cabinet, since feedback can result. Close the preamp's gain control when finished with this experiment.

2. Conduct experiment #1 again, but this time, replace the microphone with an output from a CD player. Sweep the VCF's Fc again. Now, automate this sweep using an LFO. Which is better for this task? A square wave or a saw wave? Try adding resonance to this sound. Next, try increasing the gain until some distortion can be heard. After fully closing the gain knob, set the preamp to 1000x and try creating distortion again. Describe this sound. **CD track 66**

3. Use the preamp as a waveshaper. Connect VCO-2's sine wave output to the preamp's input and connect the preamp's output to the mixer. (Be careful to set the level in the mixer low, as this can get rather loud.) Increase the preamp's gain until the sine wave sounds close to a square wave. Try a higher range of 10x is not enough. **CD track 67**

4. Use the output of a CD player to generate an envelope using the envelope follower. Connect the CD player's output to the preamp, and set a level where no distortion is present. (Monitor the preamp's output in the mixer to confirm this.) Raise the preamp's level in the envelope follower (the preamp's output is already normalled here). Connect the envelope follower's output to the VCOs, VCF, and VCA in turn. Repeat this experiment using a microphone. Is the envelope follower tracking pitch or amplitude in the incoming signal? Take a copy of the preamp's signal and route it to the filter while the filter's Fc is being modulated by the envelope follower. Is this an effective technique? **CD track 68**

5. Observe what is normalled to the envelope follower's input, and where the outputs of the envelope follower and preamp are normalled. Normals represent the most often used patches. Why were these modules normalled the way they are?

6. Observe where the ring modulator's output is normalled on the ARP 2600's cabinet. Notice what is normalled to each of the ring modulator's inputs.

7. Create metallic sounds using the ring modulator. Connect the outputs of two different VCOs to the ring modulator's inputs. Raise the ring modulator's slider in the VCA, set the VCA's gain fully closed and use an EG to open and close the VCA. How should the VCOs be tuned to produce the most harmonics and thus the most metallic sounds? **CD track 61**

8. Create a patch in which all three VCO's are tuned in unison, routed to the filter controlled by the ADSR EG, and then to the VCA controlled by an EG. Patch any two waveforms from any two oscillators to the ring modulator and raise both sliders fully. Raise the ring modulator's level in the filter and/or in the VCA. Notice how this patch will accentuate the high frequencies in the patch. **CD track 62**

9. Patch the ring modulator's output to the mixer. Patch nothing to the ring modulator (use the VCO outputs normalled there). Set VCO-1 at a lower frequency; about 100 Hz. Start with VCO-2 as low as possible while still in the audio range, and sweep its frequency up. Listen for the change in harmonics as VCO-2 continues upwards. **CD track 65**

10. Repeat experiment #9, but this time, use a square wave from VCO-1, and set VCO-1 to LF mode. While conducting this experiment, flip the DC/AUDIO switch. The difference between the two modes is easily audible in this experiment. **CD track 64**

Review Questions for Section Eleven:

1. What does the preamplifier do? Why is this simple job so important?

2. Explain what happens when the dynamic range of any circuit is exceeded. Is this a good thing or a bad thing? When is this effect desirable?

3. What aspect of an incoming signal does the envelope follower track? What kinds of signals are usually used with the envelope follower?

4. Is the envelope follower's output always a control signal or always an audio signal?

5. What module can the envelope follower replace when external signals are being used?

6. How does the ring modulator add harmonics to sounds?

7. What kinds of sounds is the ring modulator good at producing?

8. Explain the difference between DC and AC coupling on the ring modulator.

Terms To Know:

AC Coupling
AM
Amplify
Amplifier
Amplitude Modulation
Clipping
DC Coupling
Distortion
Dynamic Range
Envelope Follower
Frequency Content
Gain
Harmonic Content
Preamplifier
Range
Ring Modulator

SECTION 12

Voltage Processors

Introduction

The voltage processors are an important part of the ARP 2600, but are often neglected and misunderstood. While their operation is not difficult to understand, their applications are rather specialized. As such, they are not used as frequently as a VCO or the VCF. The voltage processor module contains three separate processors. One of the processors is a lag processor and the other two are mixing and inverting processors. The voltage processors are located between the noise generator and the S/H module. (See Figure 12-1) Two of the three voltage processors lack formal names, so the top one will be called ‘the first voltage processor’ and the one below it will be called ‘the second voltage processor.’

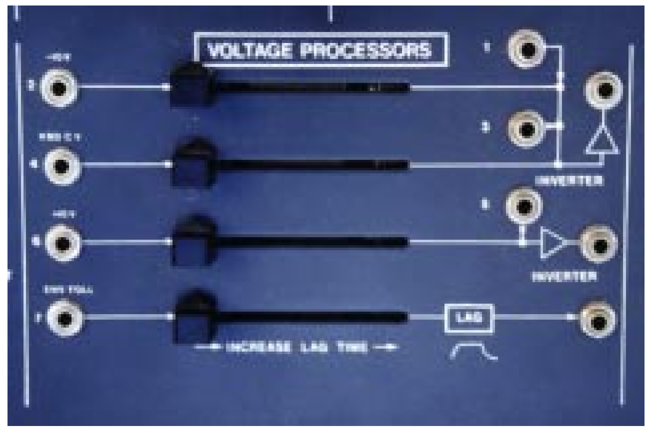


Figure 12-1: The voltage processors

The First Voltage Processor

The first voltage processor mixes and inverts. Five jacks and two sliders comprise this processor. The inputs are labeled one through four. All of these inputs are summed or mixed together, and come out of the remaining jack which isn’t labeled. In this way, this processor acts like a mixer. Inputs two and four are located on the left hand side of the module, to the left of the sliders. (See Figure 12-1) The incoming signals on each of these jacks can be attenuated using the sliders which are to the right of them. Inputs one and three have no sliders, so their levels cannot be attenuated, and nothing is normalled to them. The other normals will be discussed in a moment.

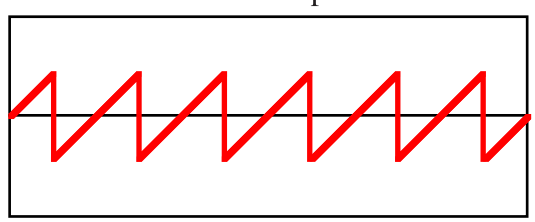


Figure 12-2: An uninverted saw wave

It is important to note that both control signals and audio signals work equally well. This processor can even be used to mix raw direct current or DC. Since this voltage processor can be used as a mixer, it will almost always be chosen over the mixer module (Section 9) for the purpose of mixing control signals. It is a better choice because this voltage processor does not feed the synthesizer’s main outputs.

Inverted Fun

Before the signals reach the output of this processor, they pass through an *inverter*. This circuit is represented on the cabinet’s panel with a triangle and the word INVERTER. An inverter is an electronic circuit which essentially flips a waveform or incoming voltage upside down. For instance, if one inputs a saw wave such as the one shown in Figure 12-2, it will be turned upside down at 0 volts as shown in Figure 12-3 on page 92. Another way to describe this is to say that the signal has been moved 180 degrees out of phase with the original signal.

Inverting the Keyboard

Several interesting possibilities present themselves when inverters have been discovered. One of the oldest tricks in the book is to invert the keyboard's control voltage so that the pitch will get lower as higher notes are played on the keyboard. This effect becomes even more interesting when one VCO is controlled with the inverted voltage and one VCO is controlled with normal keyboard CV. **CD track 69** The keyboard's control voltage is actually normalled to input four on the voltage processor for this very purpose. Observant readers will notice that ARP did away with the fancy white boxes to indicate these normals.

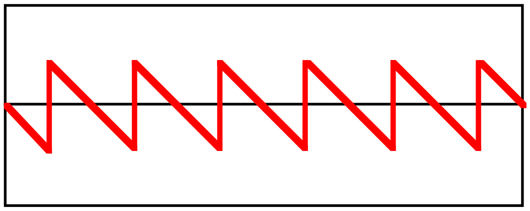


Figure 12-3: An inverted saw wave

It is important to note that when using the keyboard's voltage inverted along with an uninverted copy (which would control a different oscillator) some thought must go into how these two oscillators will be tuned. One must play a note on the keyboard, and then tune the oscillators in unison. This could be any note, but it is best to pick a note that will be in the melodic line to be played. As soon as any note other than the tuning note is played, the oscillators will no longer sound in unison. Thus, some thought must be given to which note will be chosen for tuning. Experimentation is key to understanding which note to pick. Synthesists often choose the first note in a melody so that the notes seem to branch out from the starting point. There are many other interesting possibilities, however.

Other Inverted Tricks

Another wonderful possibility is to invert the output of an envelope generator. All sorts of interesting effects can be created with this patch. VCO's can drop in pitch whenever a key is played **CD track 70** or the filter or VCA can close a bit whenever a key is played. Many modern synthesizers will allow users to invert the polarity of the instrument's envelopes.

An LFO's output can be inverted as well. For instance, if one wanted to use the saw wave shown in Figure 12-2 on page 91 to FM another oscillator, but wanted the pitch to continually descend instead of ascend, the inverter would be the correct tool for the task. **CD track 71**

Finally, audio signals can be inverted. Of course, the human ear is not able to detect whether a waveform has been inverted, but if a copy of an inverted waveform is combined with the original waveform, cancellation will occur. While cancellation is not usually the goal, this effect can work to a synthesist's advantage. Recalling from Section 3 that some of VCO-2's waveforms are naturally 180 degrees out of phase with each other, it would be problematic to attempt to feed triangle and sawtooth waves to the audio inputs on the VCF simultaneously. However, if one of the two waves is inverted, then they will be in phase with each other and will reinforce each other.

One final item about the first voltage processor bears mentioning at this time, and that is the item normalled to the input labeled "2". It is a -10 volt direct current. Of course, this voltage must pass through the inverter on its way out of the processor, so it comes out as +10 volts. All that one needs to do is to open the slider on this input to increase the voltage coming out. This voltage can also be used as an offset voltage as well. An *offset voltage* is voltage added to another signal such as a waveform to raise or lower it in the 'dynamic range' without changing the actual amplitude of the waveform. Notice how the saw wave shown in Figure 12-4 has been raised and lowered (shown in different colors at different positions) by changing the voltage offset. The ability to offset a voltage is highly useful when using a control signal to FM a VCO or to control the Fc on the VCF, since the range in which modulation will occur can easily be set and controlled.

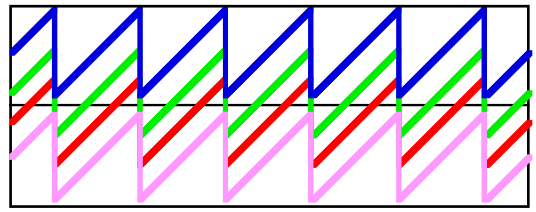


Figure 12-4: A saw wave which has been offset

The Second Voltage Processor

The second voltage processor is very similar to the first one, but with fewer features. It has two inputs, and just one output. Just as before the left most input (labeled 6) allows the user to attenuate the incoming signal with a slider. This input also has +10 volts normalled to it, and this normal is indicated just above the input in small print. Like the first voltage processor, the second voltage processor has an inverter just before its output, so the +10 volts emerges as -10 volts. The second voltage processor has a second input whose level cannot be attenuated. This input is labeled 5 and has nothing normalled to it.

The Lag Processor

The lag processor has only one input and one output. It performs an interesting duty. When the incoming voltage changes value suddenly, the *lag processor* increases the amount of time it takes this change to occur. In Figure 12-5, one can see a square wave before it passes through the lag processor and after it passes through the lag processor. (This example was taken from the ARP 2600 manual, page 41.)

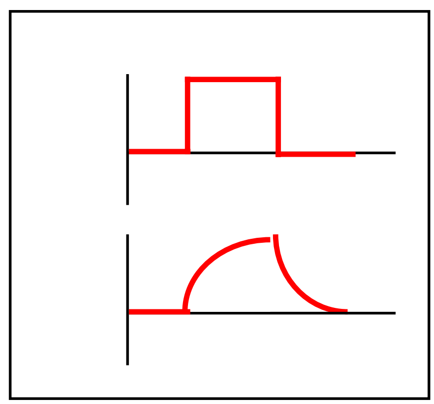


Figure 12-5: The lag processor in action

Unlike the other two voltage processors, the lag processor has a parameter. *Lag time* is the amount of time the lag processor will take to put out the full amount of voltage which is coming in. Lag time is not a voltage-controllable parameter, and thus cannot be modulated. This is not a tremendous disappointment, however, since this parameter is generally set and left alone.

If the lag processor's lag time is short enough, and the incoming voltage increases or decreases rather gradually, no lag will be noticeable in the signal until the voltage makes a large enough change that the difference can be perceived. Another factor is the distance between the notes which are being played. Portamento is less obvious when notes which are adjacent to each other are played. When notes far apart from each other are played, the difference becomes much more obvious. So, the smaller the interval played, the less lag time is needed. The lag time on the ARP 2600 can range from 0.5 milliseconds to about half of a second.

The Lag Processor in Practice

One of the more common applications of the lag processor is *portamento*. Portamento occurs when the keyboard's CV is passed through the lag processor. This causes notes to slide gradually from one to the next. It was a popular effect in the 1970's and early 1980's. It became so popular that ARP later incorporated a full-featured portamento circuit into the keyboard. **CD track 72**

Another very common application for the lag processor has to do with a flaw in the ARP 2600's design. It would seem that the filter's output stages are not quite properly matched electronically to the inputs on the VCA. On many 2600's, when the filter closes too quickly, a loud 'THUMP' can be heard. This is highly undesirable. This problem can be remedied by installing an additional capacitor in the filter module, but this is best left to a qualified technician. However, a second solution to this problem is to route the control signal controlling the VCF's Fc through the lag processor and increase the lag time just enough so that the sudden change of the filter closing is eliminated. This is really not a perfect solution to this problem as it drastically alters the sound of the filter.

The lag processor is also commonly used to process control signals from LFOs, envelope generators, and the envelope follower. In fact, the envelope follower's output is normalled to the lag processor's input. It's ability to reshape incoming signals can lend another element of control over control voltages on the ARP.

Experiments for Section Twelve:

1. Discover the first voltage processor's ability as an audio mixer. Connect the square wave output from VCO-1 and 2 to inputs two and four on the first voltage processor. Connect the processor's output to the mixer. Raise the attenuation sliders on the voltage processor to control the level of each VCO. Is it possible to hear that the waveforms are being inverted?

2. While conducting experiment #1, bring up these VCOs in the filter, open the filter completely (no resonance is needed), and then route the VCF's output to the remaining input on the mixer. Compare the uninverted signal with the inverted signal by bringing up the level of one then the other in the mixer. What occurs when both are up at the same time? Why is this occurring?

3. Invert the keyboard's CV. Open the slider to the right of input four, and connect the first voltage processor's inverter to the keyboard CV input on VCO-2. Bring up both VCO-1 and 2 in the filter. Play a note around which a melody can be centered on the keyboard, then tune the oscillators in unison. Now play a melody. What occurs and why? Try playing a different note on the keyboard, tuning the oscillators, and playing the melody again. Why does this sound different? **CD track 69**

4. Put VCO-1 into LF mode and connect its sawtooth output to the second voltage processor's input. Fully open the slider on voltage processor 2, and connect its output to an FM input on VCO-2 and raise the corresponding slider. While monitoring VCO-2 in the mixer, does the VCO's pitch ascend or descend? Why does this occur? **CD track 71**

5. While VCO-1 is still in LF mode, connect its sawtooth output to input 4 on the first voltage processor. Fully open the slider next to input four and connect the output to an FM input on VCO-2. While listening to VCO-2, raise the slider to the right of input 2. What happens to VCO-2's pitch? Why is this occurring?

6. Route the output of the ADSR generator to the input on the voltage inverter, then use the inverter's output to modulate the Fc on the filter. Route the filter's output to the VCA for gating using the AR EG. Where should the filter's initial Fc be set for this experiment to be successful? Where should the modulation depth be set? What settings on the filter give the optimal results? **CD track 70**

7. Repeat experiment #5, but this time, allow the inverted output of the ADSR EG to modulate the VCA's gain. Route a VCO to the VCA, and then route the VCA's output to the filter for gating. Use the AR EG to control the VCF's Fc. Where should the VCA's initial gain be set to complete this experiment successfully? **CD track 70**

8. Patch the keyboard's CV output to the lag processor and patch the lag processor's output to the multiple. Route the signal to all three VCOs tuned in unison. Bring up all three VCOs in the

filter and route this to the mixer. Use the VCF to gate the oscillators under the control of the ADSR EG. Create portamento by gradually increasing the amount of lag time. What settings become more obvious as smaller intervals are played on the keyboard? What settings are more obvious when larger intervals are played? **CD track 72**

9. Patch VCO-1's square wave output in LF mode to the input in the lag processor, and patch the lag processor's output to an FM input on VCO-2. Listen to VCO-2 either through the filter or directly in the mixer. Gradually increase the amount of lag time and listen to the change which is apparent. Repeat this experiment with the saw wave output on VCO-1. **CD track 73**

10. Patch a VCO in audio mode to the lag processor, and patch the lag processor's output to the mixer. What effect does increasing the lag time have on the audio signal? Why does this happen?

11. Patch VCO-1 in audio mode to the lag processor, and patch the lag processor's output to an FM input on VCO-2. Open the slider on the FM input fully. Patch VCO-2 to the mixer so that it can be heard. What happens when the lag time reaches a great enough amount? Why does this occur?

Review Questions for Section Twelve:

1. How many processors make up the voltage processors module? What are the features of each processor? What is normalled to each of the processor's inputs?

2. Which would be the best choice for mixing two control signals together: a voltage processor or the mixer module? Why is this the best choice?

3. What is the procedure for inverting the keyboard's CV? What must be taken into consideration when this patch is attempted?

4. Other than inverting the keyboard CV, how can the inverters be used?

5. Compare and contrast the first and second voltage processors.

6. How are offsets achieved, and why are they useful?

7. How should the lag processor be connected to create portamento?

8. How can the lag processor be used to eliminate the filter's thumping sound?

9. What else can the lag processor be used for?

Terms To Know:

Inverter
Lag Processor
Lag Time
Offset
Offset Voltage
Portamento

SECTION 13 KEYBOARD CONTROLS

A KEYBOARD WITH HISTORY

Since the ARP 2600 was in production for such a long time, it underwent many changes. Some of the first changes were very obvious. The color of the cabinet was changed from blue to gray and the overall shape and construction of the cabinet housing were changed drastically. The most obvious changes after the early changes were changes to the keyboard.

ARP produced several different keyboards for the 2600. The earliest keyboards were the model 3601 keyboards. These keyboards were highly distinctive since they had a long wooden handle that stretched their entire length. Their two control knobs are on the front of the keyboard rather than the top. They offered minimal control and required two cables to connect to the cabinet. These units are as rare as the blue and gray cabinets they accompanied.

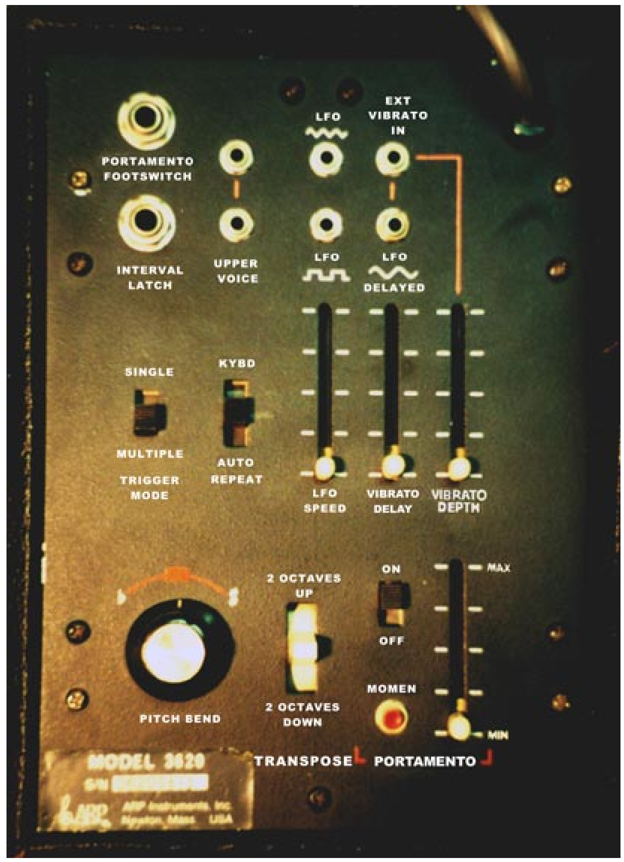


Figure 13-1: The keyboard's control panel

These 2600's (distinguished by the long wooden handles) are commonly referred to as "blue meanies" or "gray meanies" for many reasons, not the least of which is that some of the circuit boards inside them are sealed in epoxy. Repairing a circuit like this is no easy task and is considered by many to be impossible. Persons selling 2600's today will often incorrectly advertise them as 'gray meanies' simply because they have a gray cabinet or epoxied modules. However, a true gray meanie is the most rare of all of the 2600's and can easily be distinguished by its long wooden handle. Some people estimate that only five or six of these units were ever built. A true gray meanie typically brings two to three times as much as a gray-faced 2600 such as the one pictured in this book.

Notice in Figure 13-1 that the keyboard control panel has orange markings on it, indicating that it was one of the last ARP produced. This particular keyboard does not match the cabinet pictured in this book. However, many synthesists prefer the added features of a late version of the keyboard along with the sound of an earlier version of the cabinet (i.e. one with a Moog-style filter).

When ARP started producing 2600's covered in Tolex, the design of the keyboard was changed to match it. The unit's physical depth was greatly reduced, and the long wooden handle was replaced with a small plastic one. These models were dubbed the 3606, and are fairly common. They required only one cable rather than two to connect to the cabinet, and offered control over portamento and interval.

Fun With Portamento

Portamento was discussed briefly in Section 12 when the lag processor was introduced. However, the 2600's keyboard also has a dedicated portamento circuit built in. Although it is not labeled in this manner, this portamento circuit is really just a dedicated lag processor which can be switched on and off. **CD track 72**

Figure 13-2: The portamento controls

The portamento controls on the 3620 keyboard are located in the lower right-hand corner of the keyboard control panel. (See Figure 13-2) The switch labeled ON and OFF simply turns the portamento effect on and off. The button labeled MOMEN will engage the portamento circuit as long as it is held down. (I.e. it momentarily turns the portamento on.) This button only functions when the portamento switch is in the OFF position.

There is a third way to switch the portamento function on and off. A footswitch can be connected to the 1/4" jack in the upper left hand corner of the 3620 keyboard and this will allow the portamento function to be turned on and off without requiring the use of a performer's hand. Often times, performers like to add portamento to only one or two notes, and thus the momentary switch and the footswitch are useful additions.

The final portamento control is a white-capped slider in the lower right hand corner of the keyboard control panel. This slider determines the portamento time. This slider adjusts the lag time of the lag processor contained in the keyboard which actually produces the portamento. The 3604 keyboard featured a knob like the one on the preamp to set lag time, but had the same ON/OFF switch.

Interval Tuning and Other Useless Mysteries

The INTERVAL control (not pictured) allowed users to alter the amount of voltage difference from one key to the next. The practice of altering the distance between pitches is known as *microtuning*. As this amount was altered, the keyboard would no longer play chromatic half steps, but could play either larger or smaller intervals. While this had enormous possibilities in the experimental music world, few commercial artists were delighted to play melodies using 19 or more notes per octave (there are normally 12). Although a many modern synthesizers have the ability to use microtuning, most synthesists don't care to use these features.

The interval control disappeared from the 3620 version of the 2600's keyboard, which is one of the most common versions. Recall from Section 2 that this effect can be recreated by manually patching the keyboard's CV to an FM input on an oscillator and then attenuating the incoming CV signal. Since less of the keyboard's CV will modulate the oscillator's frequency, the oscillator will not change as much in pitch from note to note. Of course, attenuating the incoming keyboard CV signal can only decrease the distance from one note to the next, but the actual interval control could increase the distance as well.

Duophony

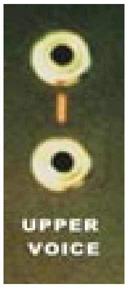


Figure 13-3: The upper voice jacks

One of the first things that strikes modern players when they play an ARP 2600 for the first time is that it is monophonic. This was all discussed in Section 1, but it takes a moment of hands-on use to really discover this for oneself. Of course, there are tuning tricks to make the 2600 sound as though it is creating chords, but there was no way to play two separate melodic lines simultaneously on early models of the keyboard.

In 1975, a man named Tom Oberheim came up with a modification kit for the 3606 keyboard which would allow it to output two keyboard control voltages simultaneously. (The keyboard is said to be *duophonic* since it can play two notes at once.) The keyboard would create a control voltage for the lowest note being played as usual and send it to the cabinet. However, when a higher note was played, this note was sent out a separate set of outputs, located just to the left of the LFO outputs. These outputs are labeled UPPER VOICE since the highest note's voltage is output here. Of course, this meant that with only three VCO's, one voice would have only one VCO and would thus be weaker sounding. However, the ability to play two notes simultaneously was such a revolutionary design that ARP was quick to recognize its potential. They integrated this design into the new version of the keyboard, the 3620.

When only one note is being played on the 3620 keyboard, that control voltage is sent not only to the cabinet in the usual fashion, but also out the UPPER VOICE jacks on the keyboard (See Figure 13-3). This results in monophonic performance.

Creating a Duophonic Patch

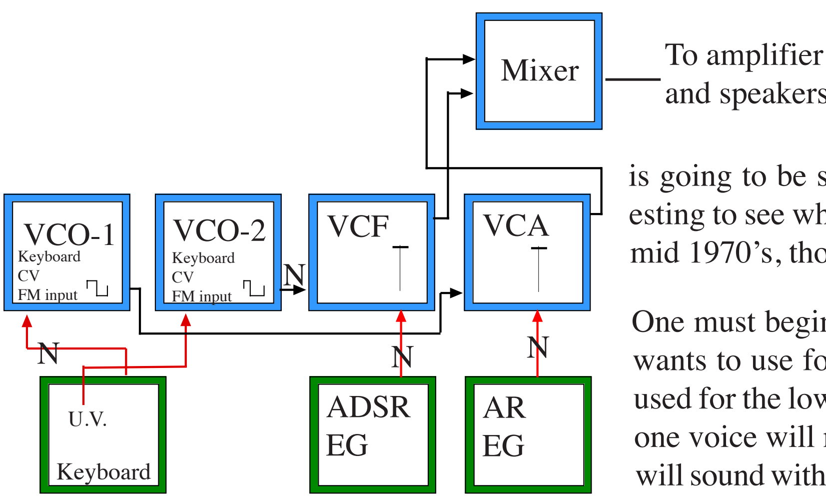


Figure 13-4: A duophonic keyboard patch

The first thing to understand about creating a duophonic patch is that if one is expecting performance rivaling modern synthesizers, one is going to be sorely disappointed. It is rather interesting to see what was considered to be hi-tech in the mid 1970's, though.

One must begin by deciding which oscillator(s) one wants to use for the upper voice, and which will be used for the lower voice. If three oscillators are used, one voice will naturally be stronger, since one voice will sound with two oscillators and the other will have only one. For the purposes of this example, each voice will be assigned one oscillator. (See Figure 13-4)

The lower voice could be assigned to VCO-1, while the upper voice is assigned to VCO-2. Since the keyboard CV is normalled to VCO-1, the lower voice is already taken care of, and no connection needs to be made. However, the upper voice signal must be manually patched to VCO-2's KYBD CV input.

This will break the lower voice CV normal to VCO-2 while patching the upper voice CV to VCO-2. Next, a timbre must be chosen. Each voice could have its own timbre, or each voice could have a unique timbre. If the user wants the 2600 to behave like a single two-voice instrument, the same waveform would be used from each VCO. Sometimes one may wish to simulate two instruments playing a duet. In this case, a different timbre from each VCO is appropriate. This same issue must be taken into account when tuning is considered. If a two-voice instrument is desired, the VCOs should just be tuned in unison. However, if one wants to simulate two different instruments, it may be more appropriate to tune in octaves or even in some other interval to create range differences between instruments. Again, this effect is further enhanced by using different timbres from each VCO.

Both oscillators could be routed to the filter for shaping and gating. However, there is a small problem with this. The filter's cutoff can be either open or closed, but cannot be both open and closed at once. If the filter is being used to gate both VCOs, imagine what might happen if one note is held down and then another is played while the first is being held. Depending upon the setting of the TRIGGER MODE switch, the EGs will either refire, causing both voices to re-attack or the EGs will not fire at all, so the second note will receive no special attack. **CD track 74**

Modern synthesizers have solved this problem by assigning each voice its own VCOs, VCF, and VCA. However, since the ARP has only one filter, and one VCA, this solution isn't helpful. It is possible to route one voice through the filter for gating while the other voice is routed through the VCA for gating, however. Since the outputs of both the VCF and VCA are present in the mixer, it is not a difficult task to hear the outputs of both. Of course, some thought must be given as to which voice will be gated by the filter and which will be gated by the VCA. This depends upon what each voice is going to be used for, of course, but bear in mind what was said in Section 8 regarding the difference between the VCF and the VCA when it comes to gating a sound.

The biggest problem with playing duophonic lines is that when only one line is sounding, the lower voice emerges from the upper voice jacks as well as following the normal route through the cabinet to the VCO's. The unfortunate result is that one must either play melodic lines in which neither voice rests at all (the keyboard will always assume that if only one note is being played, it is the lower note) or put up with the lower voice continually switching from one oscillator to two per note. This is a very obvious shift, which is really somewhat unmusical. **CD track 75** A much better solution to this problem is to have a second device which can play notes automatically for the synthesist. Such a device will be discussed in Section 15.

Latch Onto That

One alternate technique which helps to combat this problem is the INTERVAL LATCH jack, located just below the PORTAMENTO FOOTSWITCH jack. Like the PORTAMENTO FOOTSWITCH jack, the INTERVAL LATCH jack is a 1/4" connection, indicating that a foot pedal is intended to be connected here. When a foot pedal is connected and pressed on, the keyboard will 'remember' the interval between the two keys being played. The keyboardist can now play just one note, and the interval between the two notes will remain constant. This ensures that the keyboard will never switch itself back to monophonic mode by sending the lower voice CV out the UPPER VOICE jacks.

The adverse effect of using the interval latch feature is that the same effect can be created by using a monophonic keyboard and tuning the VCOs in intervals. The advantage of using the interval latch feature is that the keyboard can instantly be switched into and out of true duophonic operation. Because duophonic operation thins the 2600's sound, interval latching and duophony are best saved for performance situations rather than studio work, where tape can make up for some of these limitations. Using a tape recorder or a computer, one can record one melody using three VCOs per voice, then go back and add other melodies to it one at a time, all without sacrificing the ability to use all three VCOs at once. These techniques will be discussed in a later volume of this series.

Pitch Bending

On the 3620, the rather esoteric INTERVAL knob disappeared, to be replaced by a more useful pitch bend knob. This knob is located in the lower left hand corner of the keyboard control panel. When it is turned clockwise, the keyboard's CV rises and will in turn raise the pitch of the VCOs. When turned fully clockwise, the VCOs will sound one octave (12 half steps) higher than normal pitch. When the pitch bend knob is turned counterclockwise, the pitch will descend as far as one full octave below normal pitch.

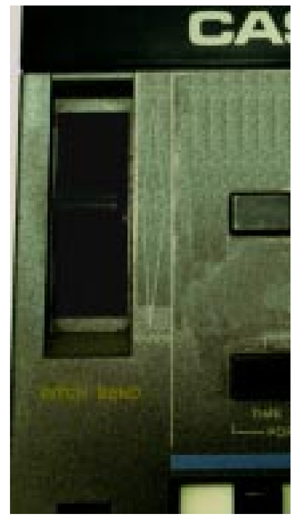


Figure 13-5: A Moog-style pitch bend wheel

Another wonderful contribution of the Moog Music company to the world of synthesizers is the *pitch bend wheel*. This is a large circular knob mounted sideways in the keyboard which can be turned forwards or backwards to bend the pitch of the oscillators up or down. These wheels are spring-loaded, so that they will snap back to normal pitch when the performer's hand leaves them. This is considered to be highly desirable, and almost every professional synthesizer on the market today employs this design. The ARP 2600 keyboard, does not, however. Instead, they provided a "Dead Area" on the pitch bend knob which was close to the normal pitch area of the pitch bend knob. Users had to manually return the knob to normal pitch, which could sometimes be a bit tricky in a darkened room. In time, the ARP design faded away into oblivion, almost never to be seen in modern synthesizers. In Figure 13-5, the pitch bend wheel on a Casio CZ-101 is shown for comparison. It employs the Moog-style pitch bend wheel.

The designers also recognized the need to be able to easily transpose the synthesizer's pitch up and down without retuning the oscillators. The lever labeled TRANSPOSE allows the keyboard to generate control voltages two octaves lower than normal when it is in the 2 OCTAVES DOWN position, and two octaves higher when in the 2 OCTAVES UP position.

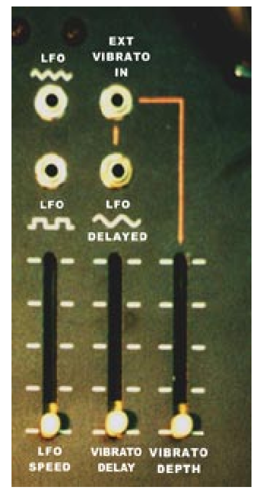
Dedicated LFO

The ARP's designers added another huge advancement to this version of the keyboard: a dedicated LFO. Up to this point in time, when vibrato was needed, one had to sacrifice one of the three VCOs, rendering the 2600 a 2-oscillator unit. This really thinned the sound, but it was too sterile without vibrato, so it was deemed a necessary evil. The 3620 offers a dedicated LFO. Synthesists can then use all three VCOs in the cabinet for producing sound and use the keyboard's VCO for producing vibrato.

The keyboard's LFO is normalled into the patch of the keyboard's CV. More accurately, the keyboard's LFO actually adds vibrato to the keyboard control voltage itself. Thus, the voltage coming into each VCO via its normal to the keyboard CV actually fluctuates to create vibrato. This is an important design concept. Because the vibrato is actually added to the keyboard CV signal before it reaches the cabinet, it is possible to use a newer keyboard such as the 3620 with an older cabinet. This is considered to be a highly desirable combination, since one can have the benefits of a duophonic keyboard and a built in LFO with delay while still retaining the older Moog filter design in the cabinet.

The LFO's Parameters

The keyboard's LFO offers four parameters. First, the rate and depth of the vibrato can be set. This is no different from the FM patch which was created in Section 3. The vibrato rate is adjusted using the LFO SPEED slider, while the vibrato depth is adjusted using the VIBRATO DEPTH slider. (See Figure 13-6) However, a delay can also be set for this vibrato. *Delayed vibrato* can be defined as the amount of time from the time a key is pressed to the time the vibrato starts in. It is adjusted using the VIBRATO DELAY slider. (See Figure 13-6) This effect is highly desirable since it better simulates real-world vibrato. Before this parameter was made available on the keyboard's LFO, synthesists would often route the output of the VCO being used as an LFO to an audio input on the VCA. The VCA would then be modulated by an EG and the output of the VCA would be fed back to the FM inputs on the VCOs. This patch is illustrated in Figure 13-7. In this patch, one VCO, the VCA, and one EG are sacrificed to create delayed vibrato. A huge sacrifice, to be sure, but the effect was desirable enough to make it worthwhile for many synthesists.



13-6: The keyboard's LFO section

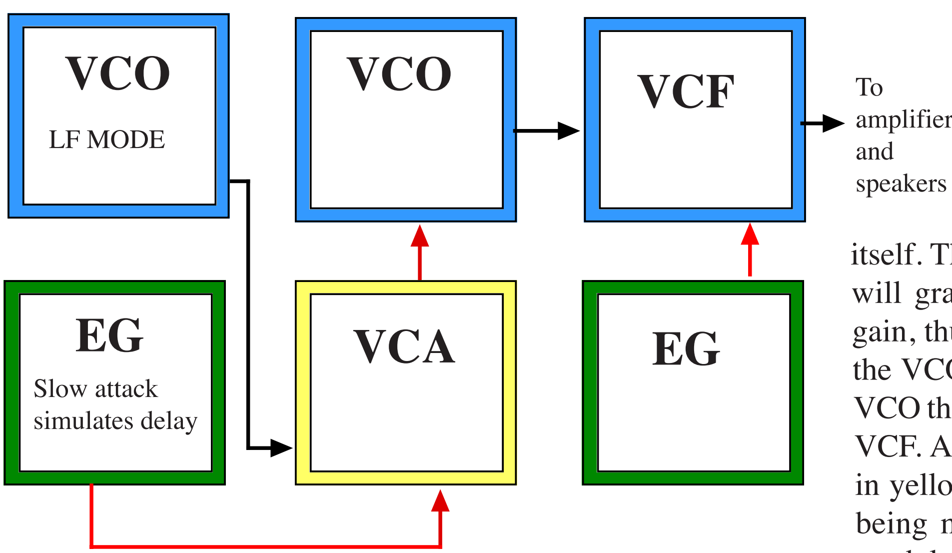


Figure 13-7: Delayed vibrato without using the keyboard's LFO

Notice that the VCO in LF mode going to the VCA is a control but, since its amplitude is actually being changed by the VCA, it is not modulating the VCA itself. The EG modulating the VCA will gradually increase the VCA's gain, thus increasing the amount of the VCO in LF mode reaching the VCO that is being heard through the VCF. Again, the VCA is illustrated in yellow since it is a carrier while being modulated by an EG and a modulator (the VCO's signal exiting it to modulate the audio VCO).

With the advent of the 3620 keyboard, synthesists could have their cake and eat it too! Delayed vibrato could be achieved without sacrificing a VCO, the VCA and an envelope generator. Delayed vibrato is a feature which can be found on almost every modern synthesizer.

The final parameter of the keyboard's LFO is its waveform. The keyboard LFO offers users three different waveforms, but the sine wave is the one which the LFO uses by default. One may recall from Section 3 that sine waves are most commonly used for the purpose of creating vibrato. If one wants to use a waveform other than the sine wave, either the triangle or square waves can be used. However, these must be patched by hand to an FM input on each VCO. Separate outputs for each of the three waveforms are available in the upper center part of the keyboard is control panel. It is interesting to note that the ARP's designers also felt that it was important to provide users with a way to add vibrato from an external source, such as an LFO on another synthesizer, or one of the 2600's own VCOs. Thus, a fourth jack is provided (EXT VIBRATO IN) which allows a control signal to be input. This control signal is added to the keyboard's CV and will be output to the VCOs in audio mode as vibrato.

Am I Repeating Myself?

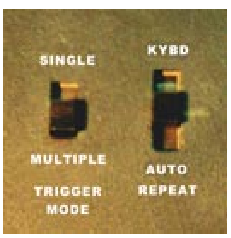


Figure 13-8: The trigger mode and repeat switches

The 3620 has a nice little amenity built in called *repeat*. Repeat affects the way the keyboard sends out trigger signals. When this three-position switch (see Figure 13-8) is in the center position, the keyboard will put out a trigger signal when one would normally expect it to (more on this in a moment). When the repeat switch is moved to the upper position, labeled KYBD, the keyboard will send out trigger pulse after trigger pulse as long as a key is being held down. When the switch is set to the lower AUTO setting, the keyboard will send out a stream of trigger pulses regardless of whether a key is being played. Oddly enough, the rate at which the trigger pulses are sent out is dictated by the keyboard's LFO. This means that when an external signal such as a square wave is input to the EXT VIBRATO IN jack, that signal will dictate the rate at which the keyboard repeats. This is useful for synchronizing the repeat to other signals.

The repeat feature lends itself to creating many special effects, but since the trigger signals are normalled to the EGs, one must remember to use the EGs to modulate something. For instance, the EGs could cause the VCA to function as a gate, thus causing a pulsing sound. Alternately, the EGs could change the pitch of the VCOs. Again, there are many possibilities available to the user here.

Trigger Mode

The trigger mode switch allows the keyboard to function in what is sometimes referred to as *legato mode*. Legato is an Italian musical term which instructs players to play an attack only on the first note in a series of notes. By moving the TRIGGER MODE switch to SINGLE, the keyboard will only put out a trigger pulse when a key is played and no other keys are being played. For instance, if one plays and holds a C, the keyboard will send out a trigger pulse. If while holding the C, one plays another key, no additional trigger pulse will be sent out. While this is very musically useful, it takes a bit of getting used to. This is particularly useful in creating the sounds one hears in older synthesizer solos. Early synthesizers (including early versions of the 2600's keyboard) were stuck in single mode, and as such, great care had to be taken in playing melodic lines to ensure that each note would sound as the performer intended it to.

When the TRIGGER MODE switch is set to MULTIPLE, the keyboard puts out a trigger pulse every time a key is played, as one would expect. However, the keyboard will still give the highest note being played priority over lower notes. This means that if one note is played, and then another note below it is played even slightly before the first note was released, the keyboard will not put out a separate trigger pulse for the second note as one might like.

This problem can be solved without making any drastic modifications to the keyboard. If the portamento circuit is turned on, but the portamento time is set to minimum, no portamento will be heard. However, the keyboard will now put out a trigger pulse every time a key is played. Because most players have been reared on modern keyboards, this is often the most comfortable way to play the ARP keyboard since this most closely resembles the way modern synthesizers respond.

One interesting trick that can be done in multiple mode is to play and hold a key, and then tap lower keys. While the lower keys won't sound, the keyboard will still put out trigger pulses, which can in turn reopen the filter or VCA. This is a wonderful feature which is available only on a precious few modern synthesizers.

The Keys to the Kingdom

To this point, little has been said about the actual keys of the keyboard. The ARP keyboard features 49 full-sized keys. They are made of a durable plastic, which generally resists discoloration unless exposed to high levels of tobacco smoke. Like the keyboard itself, ARP used several different internal designs. These designs sometimes differ radically from each other, but the important thing to understand about the keyboard is the way it works.

As each key is played, a set of very weak springs are pulled across a set of metal bars which then create the trigger, gate and CV signals. These springs are extremely fragile, and if they become bent, the keyboard may not produce gate, trigger and CV signals at the same time, or may not produce them at all. Thus, it is important to exercise some care when playing the keyboard. Although this design seems very fragile, it seems to hold up fairly well over the years. It is very important that nothing is left sitting on top of the keys as the tiny springs inside could become permanently stretched out and lose their shape.

Experiments for Section Thirteen:

1. Use the portamento controls to create slides from one note to the next. Be sure to use all three methods to turn the portamento circuit on and off. **CD track 72**

2. Create the duophonic patch shown in this section (Figure 13-4 on page 93). Try playing two single note melodies at once, and note what occurs when only one note is being played. **CD track 75**

3. While conducting experiment #2, connect a footswitch to the INTERVAL LATCH jack, and practice using the pedal to ‘latch’ intervals.

4. While playing the keyboard (monophonic patch), try moving the pitch bend knob. How far up or down can the pitch be bent? Try finding the “dead area” without looking at the knob.

5. Use the keyboard’s dedicated LFO to produce vibrato in a VCO. Experiment with the depth control. What seems to be an optimal depth setting for light vibrato? How heavy can the vibrato be? Experiment with the LFO RATE control. How fast or slow can the LFO oscillate? While experimenting with both the LFO RATE and VIBRATO DEPTH controls, is it possible to create sidebands? Experiment with the VIBRATO DELAY control. How long is the longest delay? How short is the shortest? **CD track 76**

6. Manually patch the LFO’s square output to each VCO and experiment with the depth, speed and delay controls again. Practice tuning the LFO’s depth so that it produces leaps of different intervals. **CD track 10**

7. Manually patch the LFO’s triangle output to each VCO and experiment with the depth, speed and delay controls again. Of the three waveforms available (sine triangle and square) which is the best for natural sounding vibrato? Is it still important to have the other waveforms available? Why?

8. Put VCO-2 in LF mode and patch one of its outputs to the EXT VIBRATO IN jack. Experiment with the depth, speed and delay controls again. What effect do they have now? Why is this?

9. Create the delayed vibrato patch shown in Figure 13-7 on page 103. Experiment with different settings in the VCA’s EG, the VCA’s gain, and the level of the LFO coming into the VCA until the sound of the keyboard’s LFO is closely simulated. Which is easier to use? This patch or the keyboard’s LFO? Which patch is more flexible?

10. Try the repeat function in both KEYBOARD and AUTO mode. Notice how easy it is to make changes to a patch in auto mode since one doesn’t have to play the keyboard. Try adjusting the keyboard LFO’s rate and observe the change on the repeat function. What additional use does this suggest for the EXT VIBRATO IN jack? (Think of synchronizing external equipment.) **CD track 77**

11. Set the trigger mode to SINGLE (be sure the portamento control is switched off) and play legato notes on the keyboard. Is this difficult to control? When might this be useful? **CD track 78**

12. Set the trigger mode to MULTIPLE and play legato. Is this any easier? What happens when notes are played down the keyboard in a legato fashion? What happens when notes are played up the keyboard in a legato fashion? When might this be useful? **CD track 79**

13. While the trigger mode is still set to MULTIPLE, switch the portamento controls back on, but set the portamento time to MIN. How does the keyboard respond when legato passages are played? Is this easier to control than the other modes? **CD track 80**

Review Questions for Section Thirteen:

1. Discuss the major differences between the early ARP 2600 keyboards and the last models produced.

2. List three ways the portamento circuit can be engaged.

3. What did the interval tuning control do on early keyboards? Why is this effect useful? How can this effect be simulated today?

4. What are the two main problems one might encounter when attempting a duophonic patch?

5. What are the parameters of the keyboard's dedicated LFO? Why is delayed vibrato desirable? How does the keyboard's LFO actually create vibrato in the cabinet?

6. How was delayed vibrato created before the keyboard had a dedicated LFO?

7. Explain how the repeat function works, and what controls its rate.

8. Explain the different trigger modes, how portamento relates to them, and how each mode could be useful.

9. What is the actual physical mechanism that makes the keyboard work? What must one do to keep these parts from being damaged?

10. What is the alternative design to the pitch bend knob and who developed it?

Terms And Names To Know:

Delayed Vibrato
Duophonic
Interval Latch
Legato mode
Microtuning
Pitch Bend
Pitch Bend Wheel
Repeat
Tom Oberheim
Trigger Mode
Upper Voice
Vibrato Delay

SECTION 14

PATCH DIAGRAMMING

THE MEANING OF LIFE

Why devote an entire section of this book to patch diagramming and analysis? This question can best be answered by discovering the purposes of diagramming and analyzing patches. One reason which has probably become obvious throughout the course of this book is that patch diagrams can be an excellent way to learn about how each module functions in a particular patch, and how modules are connected together to make a patch. Another useful aspect of patch diagramming is that it allows users to plan sounds ahead of time, as well as notate the sounds they have created when they are finished.

PATCH DIAGRAMMING

There are many different styles of diagramming patches. There is not a single 'best' way to diagram patches because the purpose of the diagram should dictate how the diagram looks. For instance, this book uses a very complex diagramming scheme, complete with colors to indicate which modules are carriers and modulators, and specially colored arrows to indicate control and audio signals. This style of diagramming is great for relating a lot of information about a patch, and helps people who are trying to understand the relationships of different signals coming and going in the 2600. However, for a seasoned professional who is trying to quickly document a patch he or she has made, using special colors and paying close attention to which side of a module a particular signal exits from is a colossal waste of time.

THE 2600 MANUAL

The 2600's manual contains many patches to illustrate its abilities and to introduce the reader to what was presumably a new concept. The manual presents these patches in two general styles. First, simple blocks are connected with lines and arrows. Audio signals enter and exit from the side of these blocks, while control signals enter and exit from the top or bottom of these blocks. One example can be seen in Figure 14-1. The writers of the manual preferred to indicate those connections which were not normalled by notating "P cord" next to each (not shown in Figure 14-1) rather than indicating those which are normalled with a large "N" as this book does.

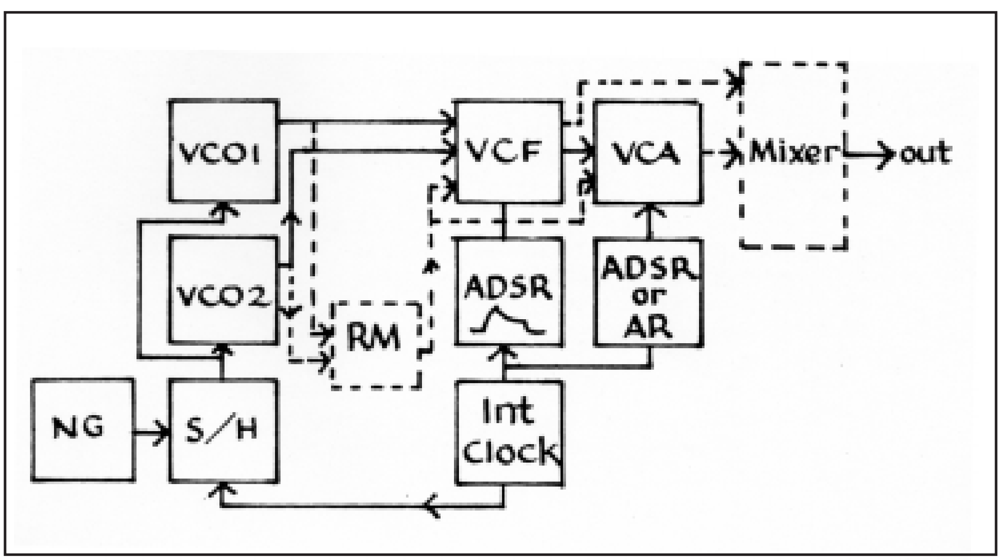


Figure 14-1: Block-style diagramming from the ARP 2600

The second style of patch diagramming was in which the ARP manual indicates patches is even more simple than the block diagrams. The manual simply names modules and draws arrows from word to word, indicating how the modules are to be patched. (See Figure 14-2) While this method of diagramming patches conveys a minimal amount of information, it also requires a minimal amount of effort and thus little time.

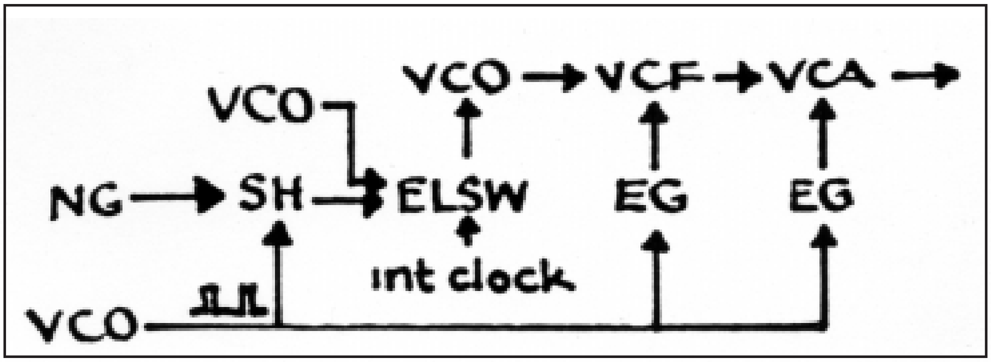


Figure 14-2: The text-style of programming found in the 2600's manual

Another highly effective way to diagram patches is to simply use a patch sheet. On some synthesizers, this would not be as useful, but since the 2600 is laid out to essentially let signals all flow towards the right, patches are usually fairly clear when drawn on a patch sheet. The advantage to this option is speed. There is no need to stop to draw or label boxes. One just has to draw in patch cords and mark the positions of sliders, switches and knobs. When one isn't as concerned about the specific relationships of one module to the next, it is probably the best of all these methods. Since this is an excellent way to convey a lot of information very quickly, this method has been chosen for the example patches analyzed in the next part of this section. Again, there is no right or wrong way to diagram a patch. Again, the kind of patch diagram should be selected based on the amount of information one needs to convey and the amount of time one has to convery it.

Patch Diagram Analysis

The ability to analyze a patch diagram and to predict how a patch will sound is an important one. While this is not an easy skill to learn, it provides the final step in putting all of the information in this book together. The ability to read a patch diagram is good, but to be able to reverse the process and predict how the sound is going to sound before actually patching the synthesizer is even more important. There are many ways to go about analyzing a patch, but these three steps may provide some direction.

1. Find out what is creating sound
2. Find out what is modifying the sound
3. What else is happening in this patch

Find Out What is Creating Sound

The first and easiest of the three steps just entails figuring out what is making the sound that will be heard. This is done by looking to see what is connected to the mixer. Of course, the VCA and VCF are normalled here, but other things may be patched in directly. If other things are patched in, it is just a matter of following the patch cords back to their source. If just the VCF or the VCA's levels are raised in the mixer, then their audio inputs must be checked. Do not forget that the VCF can also make sound on its own if the resonance is set high enough.

Once the sources of the sound are determined, this should give an idea of how the patch will sound. For instance, if the audio signals are coming from the pulse waves of VCO-2 and 3, and both pulse widths are set at 50%, chances are good that the sound will be hollow. One can also look at the positions of the frequency sliders when this information is given and make judgement about how low or high the sound will be. It is important to be able to draw on one's knowledge of every module in the 2600, and how (or if) each one sounds.

Find Out What is Modifying the Sound

This step can be a bit more tricky. It is important to check every modulation input on the entire 2600. Do not be hasty and try to judge just by looking where patch cords are. Remember that there are many normalled connections on the 2600 which can bring about huge change without the use of patch cords. It can be helpful to try to determine first which modules are actually being used as modulators, and then try to determine the effect they are having on the sound. One must stop to consider the depth and rate of the modulator, and the contour or waveform it is putting out.

Things become a bit more tricky when one has to stop to double check that none of the modulators are being modulated themselves. This is a rather complex situation, and it becomes much harder to predict the outcome of such a patch, but it is still possible.

What Else is Happening in this Patch

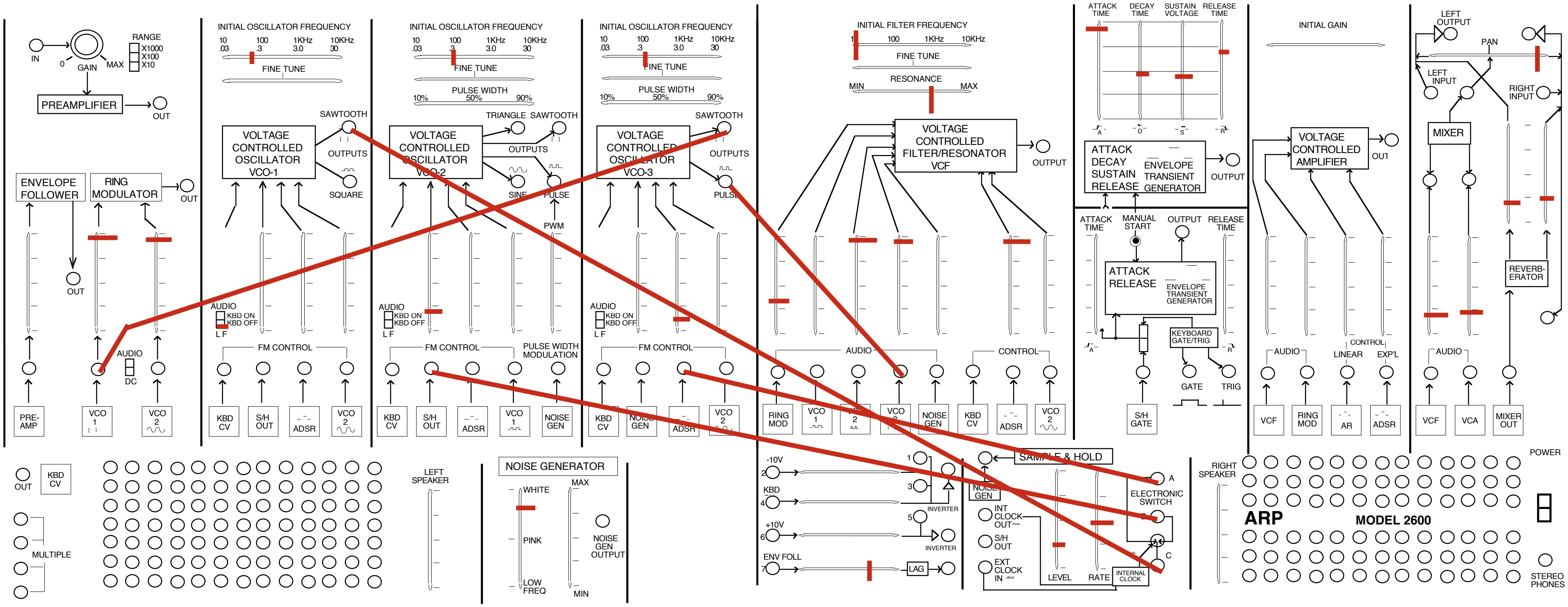
The third and final step is to look for modules which are neither producing sound, nor acting as modulators. The reverberator is a perfect example. One might stop to consider the effect of the filter and amplifier on the signal at this point. Consider the effects of the internal clock, or something feeding the S/H unit. Perhaps the envelope follower is shaping a signal, or the portamento control is engaged. Be able to account for the setting of every switch, knob, slider, jack, patch cord and dummy plug before coming to a conclusion about how the patch will actually sound.

Rather than rambling on about more specific points to look for in patches, this section will proceed to analyze two patches which will demonstrate many of the techniques one can use to analyze patches.

Patch One: The FM Special

When looking at a patch like the one diagrammed on page 112, one can become intimidated and overwhelmed. This is a natural reaction, but one must take a deep breath and remember that the first step is: Find out where the sound is coming from. (Also remember that if a slider isn't marked, it is set at the zeroed position.)

Of course, the sound has to pass through the mixer on its way out, so this is the place to start. Indeed, both the VCA's and VCF's levels are turned up in the mixer. The next step is to check both of them to see what is coming into each. In this sense, one works backwards through each module in the patch to see where the sound is coming from. The VCA is a logical place to look first, but one may be either



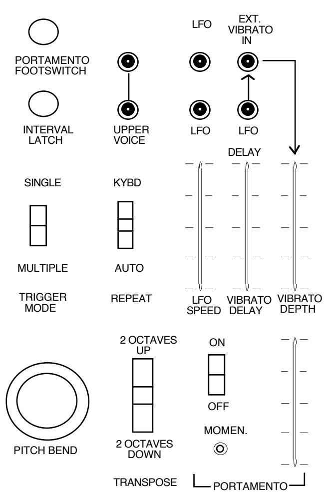


Figure 14-3: The FM special patch diagrammed

disappointed or confused looking here. None of the sliders are marked, indicating that all of them are in the zeroed position, and what is more, no patch cables are connected here. The VCA is not involved in this patch. Why would the VCA's level be up in the mixer then? Two possibilities present themselves here. If this were a real-world patch sheet, the synthesist had probably just left that slider in place from an earlier experiment and hadn't taken time to fully trace out every part of the patch before notating it. The second possibility is that in an academic situation, teachers love to insert the occasional 'dead end' on a patch on tests and quizzes.

So, the VCA had no sound coming into it, and thus has no sound coming out of it. One must then turn to the VCF, since all of the sound must be coming through it. Indeed, there are some things to look for here. VCO-2 and 3 have their levels fully raised, and the ring modulator's level is raised a little. So any sound that is going to be heard must be coming from one of these three places. Notice that a patch cord has been inserted into the VCO-3 audio input on the VCF. This cord comes from VCO-3, though, so it doesn't matter. As a final check before completing the first step, check to see that no other sources have been plugged into the ring modulator. Close inspection shows that VCO-3 has been connected to the ring modulator in place of VCO-1.

This is a good point to begin to fill in some of the blanks about this patch. The question of timbre can be answered now. VCO-2's pulse wave is normalled to the VCF, so that will be the timbre it is producing. A patch cord from VCO-3 was used to bring VCO-3's pulse wave to the VCF as well. Neither of the pulse width sliders have anything marked on them, so one is left to assume that both are centered and putting out square waves. Thus, the timbre of this patch will be a square wave.

Do not be too quick to discount the ring modulator, however. It will add extra presence to some of the high frequencies. Because VCO-2 and 3 appear to be very close in tuning, it is fairly safe to assume that they are either in tune or very near in tune, and thus very few harmonics would be produced by the ring modulator, and it would just add a little bit of fuzziness and extra high-end frequency to the sound.

Step Two

Next it is time to move on to the second step and ask "what is modifying the sound?" In this particular patch, one can see that something is patched into an FM input on both VCO-2 and VCO-3. Each FM input has the appropriate slider raised to the first mark, and these patches seem to be leading to the electronic switch. One can see that jacks A and B are being used to send the signal to VCO-2 and 3, and thus, they are being used as outputs. When jacks A and B are used as outputs, then the signal coming into jack C must be input from someplace so that a distribution patch can be created. It turns out that VCO-1's sawtooth output is feeding the electronic switch and that VCO-1 is in LF mode. VCO-1 is busy producing a saw wave at a slow pace. This signal is going to the electronic switch, which is then distributing it alternately to VCO-2 and VCO-3. Of course, the electronic switch is controlled by the internal clock, so one must check its frequency. It is set about halfway up which is usually a frequency of 60-120 Hz, or one to two cycles per second.

The ADSR EG is also busy in this patch. It is controlling the VCF, which is gating the sound, since the VCA is not in use. One can tell that the VCF is acting as a gate since its Fc is set fully closed. The settings of the ADSR generator indicate that when a key is pressed, the filter will slowly open, take a short time to decay, and will sustain at a moderate level before taking a long time to close again. There are no obvious indications that signal is coming from or going to any other modules, so it is time to now move on to the last step: What else is going on in this patch?

Step Three

One way to check to see what else is happening in a patch is to look to see where unaccounted items are marked on the patch sheet. For instance, the reverberator's level is up a little bit. Another thing that hasn't been accounted for is the resonance level of the filter. There is just enough Q to create a filter sweep when a key is played. Finally, the pan slider has been set to the right, so the patch will sound mostly from the right speaker.

There are also several dead ends which must be explored, just to make sure that they are not producing or modifying the sound. First, the noise generator isn't zeroed. However, it is not introduced to the audio path at any point. It is used as a control signal in the S/H unit, however. The S/H unit's level is turned up a bit, and it is busy sampling the noise generator. However, the S/H's output is not connected anywhere at the moment, so this too is a dead end.

Finally, the lag processor's level has been set. However, nothing is patched into it, and the only thing normalled to it (the envelope follower) is doing nothing at all since its level is set completely down. Thus, this is another dead end.

Summing up Patch One

So, when asked to sum up this first patch, one could say that when a key is played, one will hear a slow resonant filter sweep which will eventually settle down to a moderate level. This patch's timbre is a square wave, and frequency modulation is occurring. The frequency of each of the two square waves will ramp upwards then drop back to normal pitch in alteration, first VCO-2, then VCO-3. This switching will occur about once or twice per second. When the key is released, the filter will slowly close until the patch is silent. This patch will sound mostly from the right speaker, and also has a small amount of reverberation on it. This patch can be heard in **CD track 81**.

If any of the conclusions drawn during the analysis of this patch were unclear, take a moment now to return to the pertinent section to quickly review it before moving on to the second patch.

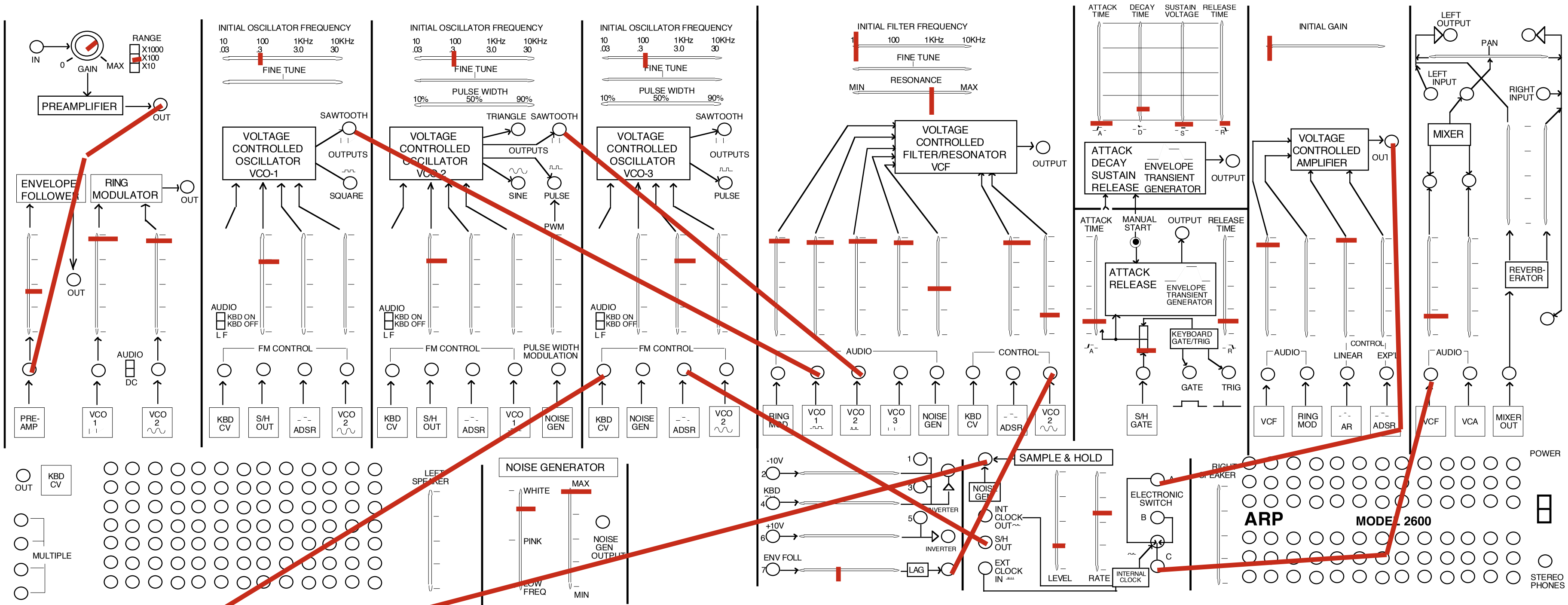
Patch Two: S/H With A Twist

While the second patch (see Figure 14-4 on page 115) is significantly more complex than the first, it is still easy to predict how it will sound. Once again, one must begin by trying to determine what will be producing the sound. In this particular patch, the VCF slider is raised in the mixer, but something has been patched to it. Jack C of the electronic switch has been connected, which would indicate that jack C is functioning as an output. When jack C is an output, the electronic switch is being used to create a switching patch.

Following the signal path backwards, one can see that there is a patch cord connected to jack A of the electronic switch, but nothing is connected to jack B. Thus, the electronic switch will be switching between the sound coming in jack A and silence. The VCA's output is connected to jack A. The VCA's gain is set fully closed which means that no sound can come through for the time being, but since one of the control inputs has something connected to it and the appropriate slider is raised, it is still possible that sound could be produced.

One of the VCA's two audio inputs has its slider raised, allowing the output of the filter to flow in. Like the VCA, the filter's Fc is completely closed. Once again, something is connected to one of the control inputs (actually both on the filter), so a provision has been made for the Fc to be raised, thereby letting sound through.

The VCF is being fed by a host of sources. All three oscillators are connected, all using their sawtooth outputs. Their levels are all raised in the filter, and they all seem to be tuned in unison since the INITIAL FREQUENCY sliders are all roughly in the same place. In addition to the VCOs, the ring modulator and the noise generator are also present, although the noise generator's volume is being attenuated a bit.



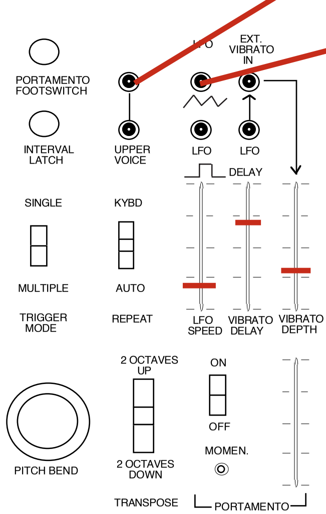


Figure 14-4: A more complex patch

This is a good time to be sure that signal is really coming from the oscillators, ring modulator, and noise generator. One can see that all of the oscillators are indeed in the audio range, both sliders are up on the ring modulator, and the noise generator's level is set at full. Indeed, all of these sources are producing sound, and this patch will produce some sound.

Step Two

One can see that there is some serious modulation taking place in this patch. To begin, the S/H module is frequency modulating all three oscillators, and at a fairly good depth. The S/H module is sampling the keyboard LFO's triangle output. It is sampling at a fairly good rate; probably three to four times per second. This means that it is producing stepped control voltages which are gradually increasing and decreasing.

In addition to being frequency modulated by the S/H module, VCO-3 is being frequency modulated by something else: the keyboard's UPPER VOICE output. This patch is a duophonic patch, which is easily discernible because of the patch to the UPPER VOICE jack.

Above and beyond all of this, the keyboard's LFO is elbowing into the picture, adding some vibrato at a slow speed. There will be a long delay between the time the key is pressed and the time this vibrato becomes apparent, however.

The VCA's gain is being modulated by the AR generator, which is set for a very short attack and release. These settings are mirrored in the ADSR generator, which is controlling the VCF. It seems that this patch is going to consist of short bursts of sound as the EG's momentarily open and close both the VCF and VCA.

VCF appears to have another source controlling it's cutoff, however. This patch comes from the lag processor, whose slider is set to about 1/3 open, which translates to about 1 second of lag, perhaps a bit more. The lag processor's input has the envelope follower normalled to it. The envelope follower has its level raised this time, and the preamp has been patched to its input. This is a redundant patch, and should throw up a red flag for test-taking purposes that someone is going to great trouble to convince the reader that this patch is authentic. In this particular patch, there is even a level set on the preamp, although nothing is connected to it, so all in all, it is just one very long dead end. This brings up an important point: If something is not drawn on a patch sheet, do not assume it is there.

Step Three

In this patch, there is very little left to examine, but the patch deserves looking over one more time. Upon closer inspection, one can see that the S/H GATE switch has been moved to its lower position, so that the keyboard will no longer trigger the EGs and now the internal clock will cause the EGs to fire. This makes sense, because the clock is going at a fairly fast rate, and a long attack or release time wouldn't have much effect on the VCA and VCF, so they had to be kept short. Many times, different parts of a patch will reinforce each other and make it obvious that the reader is on the right track. If things seem to conflict or contradict each other, stop and question the authenticity of each signal being followed.

One final item is the resonance level on the VCF. Normally, this could help to create a filter sweep sound, but the filter is going to be opening and closing too quickly to perceive an actual sweep. Instead, this level of resonance will produce a rather bubbly, rounded tone.

Summing up Patch Two

If one was asked to describe the sound of patch two, one might say that patch two is a duophonic patch with a sawtooth timbre with some noise and a bit of the high-frequency ring modulation peeking through. The pitch will continually rise and fall, but will do so in steps rather than a continuous smear in frequency. If the keyboard is played, vibrato will be introduced after a few moments, but the internal clock is causing the EGs to fire in short bursts, which are rather 'burbly' because of the resonance. Finally, the sound pulses on and off in time with the S/H module because the electronic switch is switching between the patch and silence. This sound can be heard on **CD track 82**. Again, if anything seems unfamiliar or unclear about the explanation of this patch, take a few moments to go back into the appropriate section to review.

FINAL THOUGHTS

Patch analysis is not as difficult as most people make it out to be. It is important to recall what was said in Section 3 about complex patches: they are really just a collection of simple patches which are all occurring at once. When one stops to consider a large patch such as the ones presented in this section, they seem much less intimidating when one considers them as a simple FM patch, a switching patch, etc.

There are some patches which are very difficult to predict, even if one can figure out how they work. Examples of these patches include series modulation, cross modulation, patches which use sidebands, and patches which include feedback of audio signals as control signals into the patch. Sometimes several of these techniques can be combined to produce results that no one could have predicted! (**CD track 83)** In situations like these, if one is asked to describe the patch, one is much better off just describing the processes which are occurring and summing up one's observations with a guess at what the patch will sound like.

Perhaps the most important thing to remember is to state the reasons that a particular sound should sound the way it does. In a testing situation, a teacher can give no credit for a description of sound which is totally incorrect, but partial credit could be given if the processes which are at work are described along with the sound.

If one is learning the art of patch analysis for an academic setting in which one is going to be tested, beware the "John Cage" patch. John Cage was a composer whose most notable work is one composed entirely of rests, and the players play in silence. It is very easy to create a very elaborate-looking patch which will produce no sound at all. This sort of trick question is easiest to pull off when a patch is dictated as text rather than in diagram form, but a clever person can make a diagram look convincing when it is really not.

Experiments for Section Fourteen:

1. Practice creating several patches on the ARP, and then diagram them using at least two different methods. The methods described here in the book can be used, or better yet, new ones can be created.

2. Create several patches on patch sheets, and then analyze them. Try to determine how they will sound, and then check your work by trying them on the 2600.

Review Questions for Section Fourteen:

1. What is the purpose of patch diagramming?

2. Name three ways to diagram patches.

3. What are three steps to follow when analyzing a patch?

4. How can patch diagramming and analysis help a user to become a better synthesist?

SECTION
15

The ARP Sequencer

An Introduction

An in-depth look at the ARP sequencer would easily fill 40 pages, and is beyond the scope of this book. This section is intended to provide only a brief introduction to this rare instrument. Unfortunately, many of its wonderful features can not be explored at this time, but for the curious, there is a good deal of information about this instrument on the web.

When ARP produced its first modular synthesizer, the mammoth 2500, one of the modules which could be included was a sequencer. A *sequencer* is a device which puts out a series of control voltages which the user is able to preset. The sequencer then repeatedly steps through these voltages. ARP later made a revised version of the sequencer module available as a separate unit. (See Figure 15-1)

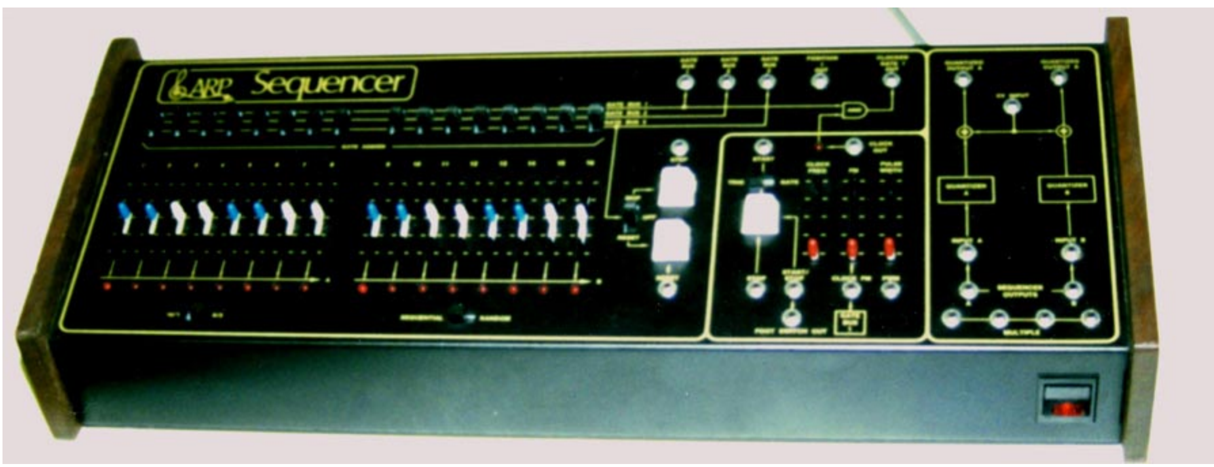


Figure 15-1: The ARP sequencer

Most analog sequencers allow users to program sequences of 1 to 16 steps. A *step* is an event in a sequence, which is usually characterized by the sequencer putting out a voltage. Sequencers put out up to sixteen steps, because there are sixteen sixteenth notes in a measure of common time, which is the most common musical meter. The voltage that the sequencer will output when it reaches any given step is set using a knob or a slider. (This varies from sequencer to sequencer.) The ARP sequencer uses sliders with blue and white slider caps, and there is one slider for each potential step. (See Figure 15-1)

The ARP sequencer is highly sought after as analog sequencers go since it offers two voltage quantizers. A **voltage quantizer** is a module which will raise or lower control voltages slightly so that the voltage will always fall on a tone from a chromatic scale. When the INITIAL FREQUENCY slider on an oscillator is moved, the frequency of the oscillator changes in a continuous smear. However, if the frequency of an oscillator is being modulated by the sequencer, and the slider on a particular step of the sequencer is moved up or down, the pitch of the VCO will change up or down in a chromatic scale. **CD track 84** Imagine how difficult it is to set particular notes on a sequencer when one must go through a tuning process for each note. The ARP sequencer's voltage quantizers eliminate this problem.

The Sequencer's Main Controls

The ARP sequencer has three large, square, white buttons which control its main functions. (See Figure 15-2 on page 121) The right most button starts and stops the sequencer's internal clock which causes the sequencer to step through its preset voltages. When the sequencer is playing, small lights along the bottom of the sequencer's panel will light one after the other under each step to indicate that that step is the active step. When a step becomes active, it sends out the voltage the user preset into it using the slider. In normal operation, the lights move sequentially from left to right from step one to step sixteen.

Figure 15-2: The sequencer's VC sliders

The bottom left hand button is labeled RESET and will return the sequencer to the first step in the sequence regardless of where it is in the sequence. It is interesting to note that the reset button can be pressed while the sequencer is playing or while it is stopped. (See Figure 15-3 on page 121)

The top left hand button is labeled STEP and will cause the sequencer to advance one step in the sequence. This is usually most useful when the sequencer is stopped, but like the reset button, this button can be used while the sequencer is playing.

Connecting the ARP Sequencer

The way the ARP sequencer is connected to the 2600 depends largely upon what one hopes to accomplish with it. One obvious application of the sequencer is to use it to control the frequency of the VCOs to produce a preset melodic line. Another great possibility is to use it to open and close the filter in a rhythmic fashion to add life to sounds. The sequencer is equally at home modulating the gain on the VCA so that sounds can get louder and softer in a rhythmic fashion.

Assuming one wants to use the sequencer to control the pitch of the VCOs, one would first connect QUANTIZED OUTPUT A on the upper right hand side of the sequencer to a multiple. The multiple on the 2600 can be used, but the sequencer also has a multiple for this purpose. (See Figure 15-3 on page 121) From the multiple, the sequencer's CV output signal can then be patched to each VCO using an FM input. It is generally recommended that one use the keyboard CV input on the VCOs, since it would not do to have both the sequencer and the keyboard trying to control the VCO at once. A second reason is that patching into any other FM input would mean having an attenuator between the input and the oscillator. In Section 2, it was discovered that even when the attenuators on the FM inputs are fully closed, they still alter the incoming control voltage. Other connections can be made to the 2600, but connecting the sequencer to the VCOs will be enough to create a basic sequence.

Programming a Basic Sequence

When programming a basic sequence, the first step is to decide whether the sequencer will produce one sequence which is sixteen steps long or two simultaneous sequences of eight steps each. This is selected using the 16/1 8/2 switch located at the lower left of the panel.

The sequencer actually has two VC outputs, labeled QUANTIZED OUTPUT A and QUANTIZED OUTPUT B. These jacks are located on the right side of the sequencer's front panel. If a sequence is 16 steps long, all of the control voltages the sequencer creates will emerge from QUANTIZED OUTPUT A. If two eight-step sequences are being attempted, the left eight steps will be output at QUANTIZED OUTPUT A while the right eight steps will be output at QUANTIZED OUTPUT B.

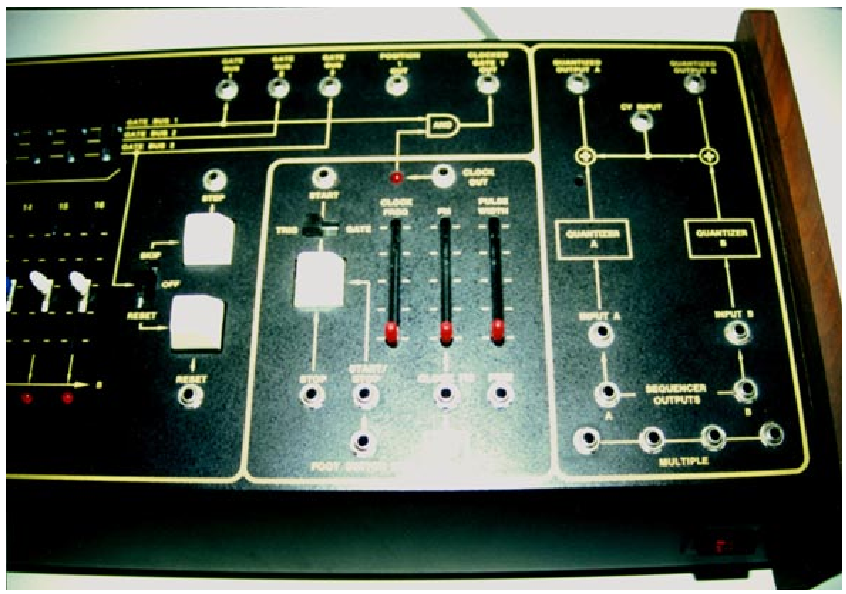


Figure 15-3: The main controls, clock, multiple, outputs, and CV input

The sequencer can step through its steps in order (sequentially), or randomly activate steps. This parameter is selected using the RANDOM/SEQUENTIAL switch located on the lower center part of the sequencer's front panel.

To create a sequence, one must first to create the sound one wants to use on the 2600, and then open the filter and/or VCA so that the patch continuously sounds. The sequencer is stopped (it always starts running when it is switched on), and the RESET button is pressed to return it to the first step in the sequence. The voltage slider on the first step is then moved into the desired position. Assuming that the sequencer is indeed connected properly to the VCOs, a change in pitch should be perceived when the slider over the first step is moved. The STEP button is then pressed, and the next step's voltage is set using the next slider. This process goes on until all of the steps have been set. The VCA and/or VCF are then closed, and the sequencer is started.

The speed of the sequence can be adjusted by increasing or decreasing the frequency of the internal clock. This is adjusted with the red-capped slider just to the right of the START/STOP button. It is interesting to note that the clock's frequency can be modulated from an external source. An input and red-capped attenuation slider just to the right of the CLOCK FREQ slider makes this possible.

Other Sequencer Connections

There are several other connections which may generally prove useful when learning to use the sequencer. It is possible to use the internal clock on the ARP 2600 to run the sequencer, rather than the sequencer's own internal clock. This is accomplished by connecting the INT CLOCK OUT jack from the S/H module on the 2600 to the jack above the STEP button on the sequencer. Now, each time the 2600's clock pulses, the sequencer will advance one step, and the two are synchronized. The advantage of this connection is that the S/H unit will sample in time with the sequencer, and the electronic switch will switch in time with the sequencer.

Another important connection can be made from the sequencer's CLOCK OUT jack to the GATE jack under the AR EG. Now, whenever the sequencer moves to a new step, the EGs will fire. Of course, if the sequencer is being controlled by the 2600's clock as in the last example, all one has to do is to move the S/H GATE switch to the lower position and the clock will automatically cause the EG's to fire. The EGs can then be set to open and close the VCF or VCA for use as a gate.

A final connection that bears mentioning is the CV INPUT jack which lies just between the quantized output jacks on the upper right hand part of the sequencer's panel. This jack accepts any control voltages, but the most common application it to patch the KYBD CV OUTPUT jack from the 2600's front panel to the sequencer's CV input jack. When different voltages are received at this jack, the sequencer automatically transposes all of the pitches up or down based on the incoming voltage. This leads to hours of endless fun, since each sequence can constantly move up or down depending upon the key played on the keyboard. For some examples, listen to **CD tracks 85-98**.

The Evolution of Sequencers

Today's sequencers have evolved tremendously from their analog ancestors. While a few companies still make hardware sequencers, many companies have turned to making software sequencers, which are computer programs that are extremely powerful.

These sequencers allow users to record musical lines in real time by playing them on a keyboard connected to the computer. This information can then be played back, edited, and stored. The number of steps they can store is limited only by the amount of memory the computer has, and this limit is usually so high that users never encounter it.

Although modern sequencers operate using a system other than the voltage control the 2600 uses, converters are available to convert the modern control signals into control voltages which the ARP 2600 can understand. This is very exciting because it means that the power of modern sequencers can be harnessed and used with the phenomenal sounds of the ARP 2600.

Experiments for Section 15: The ARP Sequencer

1. Route the output of the sequencer to all three VCOs on the 2600 using the multiple. Create a standard patch. Connect the sequencer's CLOCK OUT jack to the S/H GATE jack and set the S/H GATE switch to the lower position. Connect the keyboard's CV output to the CV input on the sequencer. Create a basic 16 step sequence using the techniques described in this section. What effect does the keyboard have on the sequence? Experiment with the START/STOP, RESET, and STEP controls, both while the sequencer is running and when it has stopped.

2. While conducting experiment #1, switch to two sequences of eight steps each. Use the first eight steps to control the frequency of the VCOs, and use the second eight steps to control the VCF's cutoff. It may be more effective to use the unquantized output for controlling the filter.

3. Repeat experiment #3, but this time, create a sequence using the unquantized output to control the frequency of the oscillators. Which procedure is easier? What is the advantage to the method described in this experiment?

4. Repeat experiment #2, but this time, use the second eight steps to control the VCA's gain. Is this as effective as #2? Which control input, works best for this purpose, linear or exponential?

5. Create a patch in which just white noise is fed into the VCF. Add about 60% resonance, and use the sequencer to control the filter's cutoff. Add a touch of reverb. What does this sound like? Could someone have gotten an idea for a drum machine in this way?

6. Use the 2600's internal clock to drive the sequencer by connecting the INTERNAL CLOCK OUT jack to the sequencer's STEP input jack. Stop the sequencer before making this connection, or both clocks will try to control the sequencer at once. Why does this work?

Review Questions for Section Fifteen:

1. Where did the design for the ARP sequencer come from?

2. How can the ARP sequencer be synchronized to the 2600's internal clock?

3. Name three modules on the 2600 which are appropriate for modulation from the sequencer.

4. Describe the process of creating a sequence.

5. How many steps can a sequence consist of on the ARP sequencer?

6. What internal source controls the speed of the sequencer? Can this parameter be modulated?

7. What are the three main controls of the ARP sequencer?

8. What allows ease of tuning of steps and is a great advantage to the ARP sequencer?

9. Describe the basic connections which must be made to the sequencer, and the purpose of each.

Terms To Know:

Reset
Sequencer
Start/Stop
Step
Voltage Quantizer

GLOSSARY

A

AC COUPLING - Incoming waveforms are centered around the 0 volts mark. Another way to think of this is that any DC offset which exists in a signal is cancelled out as the signal enters a module. Most of the inputs on the ARP 2600 are DC coupled, while the inputs on the ring modulator are switchable between AC and DC coupling since the two produce very different results in ring modulation. (See DC coupling, Ring modulation, amplitude modulation.)

ADDITIVE SYNTHESIS -The process of constructing a complex sound using a series of fundamental frequencies (pure tones or sine waves). Each of the fundamental frequencies usually has its own amplitude envelope which allows independent control of each partial (harmonic). Some organs are based on additive synthesis.

ADSR - Abbreviation for Attack, Decay, Sustain, and Release. These are the four parameters found on a basic synthesizer envelope generator. An envelope generator is sometimes called a transient generator or a contour generator. The Attack, Decay, and Release parameters are rate or time controls. Sustain is a level. When a key is pressed, the envelope generator will begin to rise to its full level at the rate set by the attack parameter, upon reaching peak level it will begin to fall at the rate set by the decay parameter to the level set by the sustain control. The envelope will remain at the sustain level as long as the key is held down. Whenever a key is released, it will return to zero at the rate set by the release parameter.

AMPLIFIER - A device which increases the level of a signal. See also preamplifier.

AMPLIFY - To make louder or increase the amplitude or height of a waveform. This task is performed by an amplifier.

AMPLITUDE - The height or volume of a waveform. It can also be represented through higher voltage.

AMPLITUDE MODULATION - (AM) A change in the level of a signal caused by another signal. Although this term can be used if a Voltage Controlled Amplifier (VCA) were being modulated by a Low Frequency Oscillator (LFO), it is usually reserved for patches utilizing the ring modulator. The abbreviation of Amplitude Modulation is AM. (See Ring modulator)

ANALOG SYNTHESIZER - A synthesizer which uses voltage-controlled modules to synthesize sound. The concept of a variety of analog modules all of which can interconnect via a standardized voltage control system was invented by Dr. Robert Moog. The three main voltage controlled modules in an analog synthesizer are: Voltage Controlled Oscillator (VCO), Voltage Controlled Filter (VCF), and Voltage Controlled Amplifier (VCA).

AR - Abbreviation for Attack-Release. A trimmed-down version of the ADSR EG. The sustain is permanently set to full open and as a result, there is no decay stage.

ARP - A synthesizer company founded by Allen R. Pearlman, for whom it is named.

ATTACK - The first parameter of an envelope generator which determines the rate or time it will take for the event to reach the highest level before starting to decay.

ATTENUATION - The act of decreasing the amplitude of any signal

AUDIBLE RANGE - The range of frequencies that the human ear can hear. A healthy young human can usually hear from 20 cycles per second to around 20,000 cycles per second (20-20,000 Hz), less after prolonged exposure to loud sounds or music.

AUDIO SIGNAL - A signal which is intended to eventually be heard.

B

BAND PASS FILTER - A filter which allows only a selected band of frequencies to pass through while rejecting all other frequencies above and below the cutoff point. Usually a bandpass filter will allow the user to set the width of the passband.

BAND REJECT FILTER - See Filter.

BANDWIDTH - The breadth of a range of frequencies. Bandwidth is usually dealt with when dealing with filters. For instance, a band reject filter might affect all frequencies within a one octave range.

C

CABINET- the main part of the synthesizer which contains all of the modules except the keyboard controls.

CANCELLATION - A reduction in volume which occurs when two identical waveforms are 180 degrees out of phase with each other.

CARRIER - A module which is being modulated by a modulator.

CENT - Unit of pitch equal to 1/100 of a semitone.

CARLOS, DR. WENDY - Most famous for her 1968 album *Switched-On Bach* and the soundtrack to the movie *A Clockwork Orange*. She studied under Vladimir Ussachevsky, and is good friends with Bob Moog. She is credited for bringing synthesizers into the public eye.

CENTER DETENT - A notch in the center of a knob or lever which allows the performer to find the home position.

CENTER FREQUENCY - The frequency around which a band reject or bandpass filter will attenuate or pass frequencies.

CLIPPING - See Distortion

CLOSED - An attenuator is closed when it is fully attenuating the signal. E.g. the volume of a signal will be completely reduced.

CONTOUR - See Envelope Generator

CONTROL SIGNAL - A signal which the user doesn't intend to hear. It is used to modulate another part of the synthesizer.

CONTROL VOLTAGE - A raw electrical current which represents a value. It can be assigned to a parameter on a module by connecting it to that module with a patch cord.

CROSS MODULATION - Modulation in which two audio range oscillators frequency modulate each other simultaneously. This often produces metallic timbres.

CUTOFF FREQUENCY - The frequency above which a low pass filter will start attenuating signals present at its input. Abbreviated Fc.

CUTOFF SLOPE - The rate at which the filter attenuates harmonics.

D

DC COUPLING - Sounds being input are left unchanged. (As opposed to having their offset set to zero volts as in AC coupling). This means that an incoming saw wave which moves between 0 and +10 volts would continue unchanged rather than being changed to -5 to +5 volts. (See AC coupling)

DECIBEL (dB) - A reference for the measurement of sound energy. The minimum change in volume that the human ear can perceive. Named after Alexander Graham Bell.

dB/OCTAVE
The unit typically used to indicate the slope of a filter, or how fast the
frequency response rolls off past the cutoff frequency. Example: A 24
dB/octave filter would attenuate an input signal by 24 dB one octave above the cutoff frequency, by 48 dB two octaves above the cutoff frequency, and so on.

DECAY - The second stage in an ADSR type envelope generator. See ADSR.

DEPTH - The amount of modulation. Sometimes called Amount, Width, Intensity or Modulation Index.

DISTRIBUTION PATCH - A patch involving the electronic switch in which a signal input to jack C will be alternately distributed to jacks A and B which will serve as outputs.

DISTORTION - A waveform's shape is changed when its amplitude becomes so great that it exceeds the dynamic range of an electronic circuit. The tops and bottoms of the waveforms are clipped off, thus changing or distorting the shape of the waveform. This results in a timbral change, often resulting in a harsher quality.

DOUBLE MODULATION - Two different modulators modulate a single carrier. Although double frequency modulation is the most common form of double modulation, any case in which two modulators modulating a single carrier is double modulation.

DUMMY PLUG - The plug from a patch cord without the cord. It allows the user to break a normal and not connect anything else to that jack.

DUTY CYCLE - The amount of time a pulse wave is 'on', and the amount of time a pulse wave is 'off'.

DUOPHONIC - The ability to play two notes at once.

DYNAMIC RANGE - The range of possible amplitudes that a given electrical circuit can handle.

E

ECHO - Sound waves which have bounced off of an object and returned to a listener.

ELECTRONIC SWITCH - A module which alternates an electrical connection between either two inputs and one output or two outputs and one input.

EMPHASIS - See Resonance

ENVELOPE - A voltage contour created by an envelope generator or the envelope follower. It is almost always used as a control signal. See ADSR, AR, and Envelope Generator.

ENVELOPE FOLLOWER - A module which can create a voltage contour or envelope from an incoming audio signal which can then be used to modulate other modules, much like the output of an envelope generator.

ENVELOPE GENERATOR (EG) - A circuit, usually triggered by pressing a key on a keyboard, that generates a changing voltage with respect to time. This voltage typically controls a VCF or VCA. An AHDSR and ADSR are two types of Envelope Generators. See ADSR.

EXPONENTIAL - A response curve in which increasingly large voltages are required to produce the same amount of change in a module.

F

FADER - A potentiometer which allows a user to select a value by sliding a small knob up or down, or from side to side. See also Slider and Attenuation.

Fc - See Cutoff Frequency

FEEDBACK PATCH - A patch in which the output of one of the modules in the patch is fed to one of the inputs on a module in the patch, thus creating a feedback loop. (This is usually done with an audio signal fed to a control signal input.)

FILTER - A device used to remove unwanted frequencies from an audio signal thus altering its harmonic structure. Low Pass filters are the most common type of filter found on music synthesizers. They only allow frequencies below the cutoff frequency to pass (Low Pass). High Pass filters only allow the high frequencies to pass, and Band Pass filters only allow frequencies in a selected band to pass through. A Notch filter rejects frequencies that fall within its notch.

FILTER SWEEP - A resonant sweep of the filter's cutoff frequency.

FINE TUNE - An attenuator found on the VCO's and VCF which allow the user to adjust the frequency and cutoff frequency respectively of a module in very small increments.

FIXED ARCHITECTURE - A synthesizer construction scheme in which the oscillators are permanently connected to the filter, which is permanently connected to the VCA, and no other connections may be made by the user. This category describes about 98% of modern synthesizers.

FREQUENCY - The number of cycles of a waveform occurring in a second.

FREQUENCY MODULATION (FM) - Occurs when the rate of an oscillator or other time-based module is controlled by a control voltage from another module.

FUNDAMENTAL - The first, lowest note of a harmonic series. The fundamental frequency determines a sound's overall pitch.

G

GAIN - The factor by which a device increases the amplitude of a signal. Negative gain will result in the attenuation of a signal.

GATE - The act of making a patch silence itself when notes are not being played.

GATE SIGNAL - A control voltage generated by the keyboard as long as a key is being held down. It tells the envelope generator to remain in its sustain stage.

H

HERTZ (Hz) - A unit of frequency equal to 1 cycle per second. Named after Heinrich R. Hertz.

HIGH PASS FILTER - See Filter

I

INPUT - a jack which only accepts incoming signals. It must be connected to an output.

INTERNAL CLOCK - A device which puts out a continuous stream of trigger pulses which can then control the timing of the S/H unit and the electronic switch. It's output is a square wave.

INTERVAL LATCH - A footswitch connector on the keyboard which allows users to 'hold' a certain interval as notes are played up and down the keyboard. Available only in duophonic mode.

INVERT - The act of flipping a signal upside down at the 0 volts mark, which effectively puts it 180 degrees out of phase with the original incoming signal.

INVERTER - A device which flips a signal at the 0 volts mark. The result is that the signal will be 180 degrees out of phase with the incoming signal.

J

JACK - A small hole which the plug of a patch cord is plugged into. It allows electrical connections to be made between modules.

K

K - Abbreviation for Kilo or 1000

KEY TRACKING - A feature of the filter in which a small amount of the keyboard's control voltage is few to the filter's control inputs for the purpose of minutely increasing or decreasing the filter's Fc as higher or lower notes are played on the keyboard. This insures that the timbre will remain constant across the keyboard's range. If key tracking is not employed, the higher notes will become duller while the lower notes will be brighter since the higher notes's highest harmonics will be attenuated by the filter as they reach the cutoff frequency.

KEYBOARD- the part of the 2600 that actually has the keys on it. It also includes the keyboard controls.

KEYBOARD CONTROL VOLTAGE - A control voltage produced by the keyboard to represent the pitch being played.

L

LAG PROCESSOR - A voltage processor which increasing the amount of time a control signal takes to change from one voltage to the next. The amount of time the lag processor takes to make this change is known as the lag time. When the keyboard CV is patched through the lag processor, portamento results.

LAG TIME - See lag processor

LF MODE - A mode in which an oscillator is made into a Low Frequency Oscillator or LFO. In this mode, the keyboard no longer controls the frequency of the oscillator, and the oscillator will usually oscillate in the sub-audio range. See LFO.

LFO - Low Frequency Oscillator. An oscillator used for modulation whose range is below the audible range (20 Hz). LFOs are commonly used in frequency modulation to create vibrato.

LINEAR - The opposite of exponential. The voltage being put into a circuit will produce a proportional amount of change in that circuit.

LINEAR TO EXPONENTIAL CONVERTER - A circuit commonly found in voltage controlled amplifiers which allows an incoming linear signal to affect the VCA's gain in an exponential manner.

LOW PASS FILTER - A filter whose frequency response remains flat up to a certain frequency, then rolls off (attenuates signals appearing at its input) above this point.

M

MANUAL START BUTTON - A small red button located in the envelope generators which produces both a trigger and gate signal when pressed and held. These signals are normalled to the envelope generators.

MASTER-SUBMASTER RELATIONSHIP - A relationship in which a master control, such as the INITIAL FREQUENCY slider on the VCO's picks a range of values, and a submaster control such as the FINE TUNE slider chooses a specific value from that range. The submaster provides a more precise adjustment of the value already chosen by the master.

MICROTUNING - A method of tuning in which the distance between pitches of adjacent keys on the keyboard is increased or decreased to produce non-chromatic scales.

MIXER - A module or freestanding device which allows the user to combine two or more signals. Mixers usually allow the user attenuate each incoming signal individually.

MODULAR - A synthesizer which makes use of modular architecture; i.e. it has its parts divided into sections which can be interconnected in any order the user desires using patch cords.

MODULATION - A procedure in which the output of one module changes the value of a parameter of another module.

MODULATOR - A module which is putting out a signal which will modulate a carrier.

MODULE - A part of a synthesizer which does a specific job. Some examples of modules include oscillators, filters, and preamps.

MONOPHONIC - A musical instrument that is only capable of playing one note at a time.

MOOG, DR. ROBERT - Often considered the father of modern synthesizers. Dr. Moog invented many of the modules which are still in use today in one form or another. Devised the system of voltage control, invented the VCO, VCF, and VCA. Built the first working envelope generator as well. Worked as an assistant at the Columbia-Princeton Institute where he met Wendy Carlos and Vladimir Ussachevsky.

MULTIPLE - A device which makes up to three identical copies of an incoming signal.

MULTIMODE FILTER - A filter which is capable of acting as a lowpass, highpass bandpass, or band reject filter. The filter may be able to perform all of these kinds of filtering simultaneously. It may also be able to perform other different kinds of filtering.

MULTITIMBRAL - The ability of a musical instrument to produce two or more different sounds or timbres at the same time.

N

NOISE - An unpitched sound. Also sound which has so many harmonics in such high amounts that it is difficult to perceive the fundamental's pitch.

NOISE GENERATOR - A module which creates noise. Noise generators often feature a built-in low pass filter which allows them to produce several different kinds of noise.

NORMAL - A connection which is premade between modules before any cords are inserted. A normal is usually made between modules which are most frequently connected together.

NOTCH FILTER- See Band Reject Filter.

O

OFFSET- The position of a waveform within a circuit's dynamic range. E.g. a saw wave might move between 0 volts and +10 volts, or might move between -5 volts and +5 volts. The difference between the two is the offset.

OPEN - A fader is said to be open when it is attenuating the least amount possible.

OSCILLATOR - A module which produces varying voltage in a pattern called a waveform. By producing this waveform at different speeds (frequencies) the oscillator can change pitch. By changing the pattern of voltage, the oscillator can change timbre. The oscillator produces the raw sound which is then shaped by other modules of the synthesizer.

OUTPUT - A jack which can only put forth a signal, but can never accept an incoming signal.

P

PAN - See Panoramic Potentiometer.

PANORAMIC POTENTIOMETER- A control which determines how much of a signal will be output to the right channel path and how much signal will be output to the left channel path.

PARAMETER - Something which can be changed.

PATCH - Referring to a particular sound created on a synthesizer. Comes from the use of patch cords on the original modular synthesizers.

PATCH CABLE - A cable used to connect different synthesizer modules together. Their color indicates the kind of signal they are carrying.

PAUL, LES - A guitarist who made great advances in modern recording. His experiments defined studio signal flow which is used today. His other inventions included multitrack recording, spring reverb, and a highly improved version of the electric guitar which is still used today.

PHASE ANGLE - A measurement of phase in degrees. Like a circle, phase angle is measure in 0 to 360 degrees.

PHAT TUNING - A tuning technique in which oscillators are intentionally put slightly out of tune to create a warmer, richer sound.

PINK NOISE - White noise which has had some of its higher frequencies attenuated with a low pass filter.

PITCH - Frequency. Determined by the rate of oscillation of an oscillator.

PITCH BEND - A control found on most synthesizers which allows the user to smoothly raise or lower the pitch of all oscillators simultaneously.

POLE - A Measure of the cutoff slope of a filter. One pole is equal to six decibels of volume reduction for every octave higher or lower the sound goes.

POLYPHONIC - A musical instrument that is able to play more than one note at the same time.

PORTAMENTO - An effect achieved when the keyboard's control voltage is sent through a lag processor. The oscillators glide from note to note.

PREAMPLIFIER - An amplifier which is used at the beginning of a signal path to raise the level of an incoming signal to that which is required by the rest of the device(s) it will be used with.

PULSE WAVE - A waveform which is in one of two states: 0 volts or +10 volts. The percentage of time it spends in the +10 volts states is expressed as the duty cycle. Square waves are a kind of pulse wave. The duty cycle can be varied using a slider, or in some case, can fall under voltage control. Pulse waves can sound nasal to hollow depending upon their duty cycle.

PULSE WIDTH- The duty cycle of a pulse wave

PULSE WIDTH MODULATION (PWM) - A kind of modulation in which the duty cycle of a pulse wave is changed by an incoming control voltage

Q

Q - The figure expressing a filter's resonance. Varying Q varies the sharpness of the filter sound.

R

RANGE - A setting of the preamplifier which determines the gross amount of amplification which will be applied to the incoming signal.

RATE - The frequency of a modulator or clock.

REDUNDANT PATCH - A patch made with a patch cord which duplicates a patch already made by a normal. These should generally be avoided.

REFLECTION - See Echo.

REINFORCEMENT - An increase in amplitude which is heard when two identical signals are in phase with each other.

RELEASE - The final stage of an envelope which begins when a key is released.

REPEAT - A switch on the keyboard which causes the keyboard to consistently output trigger pulses either all the time or when a key is pressed depending upon the setting of the switch.

RESET - One of the ARP sequencer's main control buttons. When RESET is pressed, the sequencer returns to the first step in the sequence.

REVERBERATION (REVERB) - A time based effect which occurs when multiple reflections (echoes) return to a listener in such rapid succession that the listener is no longer able to perceive individual echoes, but instead perceives a wash of sound.

REVERB TANK - A brass tank which contains several flimsy springs. When an audio signal passes through these springs, they generate and effect similar to the reverberation which occurs in natural acoustic spaces. Invented by Les Paul.

RING MODULATION - See Ring Modulator

RESONANCE - A frequency at which a material object will vibrate. In a filter with resonance, a signal will be accentuated at the cutoff frequency. See Q.

RING MODULATOR - A module which allows users to apply amplitude modulation (also Ring modulation) to two different waveforms. In Ring modulation, all of the frequencies of the harmonics of each of the two incoming waveforms is added and subtracted. The resulting waveform output has many harmonics and often has a metallic sound to it. (See Amplitude Modulation)

S

SAMPLE - A instantaneous measurement taken of an incoming voltage.

SAMPLE-AND-HOLD (S/H) - A Module which takes a sample of an incoming voltage to determine its value, and then holds that value, continually putting that voltage out its output until another sample is taken. The rate at which the S/H module takes samples is determined by either the internal clock or an input from an external clock.

SAWTOOTH WAVE (RAMP WAVE) - A waveform which increases gradually to +10 volts before falling sharply to 0 volts to begin again. Sawtooth (saw) waves have a large number of upper harmonics which give them a rather buzzy sound.

SELF-OSCILLATION - When some filters have their resonance increased to a high enough level, they will begin to produce a sine wave. This phenomenon is known as self-oscillation.

SEMI-MODULAR - A synthesizer which is fully modular, and each module can be addressed individually, but which also has normals.

SERIES MODULATION - Modulation in which module A modulates module B which in turn modulates module C.

SEQUENCER - A device which puts out a series of preset voltages in a rhythmic pattern.

SIDEBANDS - Frequencies which are added to a sound when modulation occurs that is fast enough and deep enough. Sidebands most frequently occur during frequency modulation, but can also occur during PWM or AM.

SINE WAVE - A waveform which has only the fundamental and no harmonics. It has a rather pure sound and is commonly used as the modulator's waveform in vibrato.

SKIP - One of the three main control buttons on the ARP Sequencer. It causes the sequencer to move to the next step in a sequence. This function works while the sequencer is playing and while it is stopped and can be voltage-controlled

SLIDER - See Fader.

SOURCE SWITCHING PATCH (SWITCHING PATCH) - A patch using the electronic switch in which jacks A and B are used as inputs and jack C is used as an output. The switch then switches between the two sources which go to one destination.

SQUARE WAVE - A pulse wave with a duty cycle of 50%. It has only the odd harmonics and a hollow sound to it.

STAGE - One individual part of an envelope generator's output. E.g. Attack. Also, any one of an envelope generator's parameters.

START/STOP - One of the ARP sequencer's three main controls. The START/STOP control starts and stops the sequencer's internal clock, thus causing it to run or stop. This function is voltage controllable.

STEP - A step is a single event in a sequence. Sequences on the ARP sequencer can have between one and sixteen steps. A step is usually characterized by the sequencer putting out a new voltage.

SUBSONIC - A tone whose fundamental's frequency is below 20 Hz and is thus not audible to human ears.

SUPERSONIC - A tone whose fundamental frequency is above 20 kHz and thus cannot be

SUBTRACTIVE SYNTHESIS - A method of synthesis in which harmonics are removed from one or more harmonically rich waveforms,

SUSTAIN - The third stage of an ADSR envelope in which the voltage holds at a given level until the gate signals is no longer applied to the envelope generator.

SWEEP - The act of moving a parameter through some of all of its possible values.

T

THRESHOLD OF PAIN - The volume of sound at which it is painful to hear the sound.

TIMBRE - The raw sound of a particular waveform, determined by its harmonic content, and thus its shape.

TREMOLO - An effect which is like vibrato. A sub-audio sine wave modulates the gain on a VCA to produce a wavering effect in the sound.

TRIANGLE WAVE - A waveform which has only the odd harmonics, slightly buzzy in sound, but more mellow than a saw wave. Increases to +5 volts before sharply changing to decrease to -5 volts on the 2600.

TRIGGER PULSE - A +15 volt spike of electricity sent out by the keyboard when a key is pressed to the envelope generators to cause them to begin their attack stage.

TRIGGER MODE - A switch on the keyboard's control panel which determines when the keyboard will put out a trigger pulse.

U

UPPER VOICE - Output jacks on the keyboard's control panel out of which the keyboard sends a keyboard control voltage representing the highest note being played on the keyboard. This is separate from the regular keyboard CV being sent to the cabinet, and it allows the ARP to play two notes at a time.

USSACHEVSKY, VLADIMIR - Modern teacher and inventor. Taught Wendy Carlos at the Columbia-Princeton Center for Electronic Music.

V

VALUE - The setting of a parameter.

VCA - See Voltage-Controlled Amplifier.

VCF - See Voltage-Controlled Filter.

VCO - See Voltage Controlled Oscillator

VIBRATO - A form of frequency modulation. Usually done by an LFO. Results in a slight rising and falling in pitch. Common parameters include vibrato depth (frequency modulation depth) vibrato rate (frequency of modulator) vibrato waveform (waveform of modulator) and vibrato delay (The amount of time between the key press and the onset of vibrato). Vibrato lends a more human sound to otherwise sterile waveforms and is considered to be highly desirable.

VOLTAGE CONTROL - A system which uses raw voltages called control voltages to represent values and to send them from module to module.

VOLTAGE CONTROLLED AMPLIFIER - A module which allows the user to change the amplitude of an incoming signal. Its gain parameter can be modulated.

VOLTAGE CONTROLLED FILTER - A module which allows the user to remove and/or accentuate harmonics from incoming waveforms. The frequency at which it begins to attenuate signals can be controlled by an incoming control voltage. Its cutoff slope can be altered by adding resonance.

VOLTAGE CONTROLLED OSCILLATOR - A Module which creates the raw sound which is shaped by all of the other modules. It creates a waveform

VOLTAGE QUANTIZER - A module which causes voltage to be changed (either increased or decreased) to a specified value. These values are usually set up so that voltages will always fall on a note of a chromatic scale, thus making it easier to select a particular note. Two of these modules are found on the ARP sequencer.

VOLUME - See Amplitude.

W

WAVEFORM - A repeating pattern of voltages produced by an oscillator whose shape determines the timbre of the sound. It's height determines the amplitude or volume of the sound. Its frequency or speed in repeating determines the pitch of the sound.

WHITE NOISE - A random amount of all frequencies simultaneously. This is usually produced by the noise generator.

Z

ZERO - The process of resetting all knobs, sliders and switches to their default location and removing all patch cords.

INDEX

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About the Author

Figure 16-1: The doer of dirty deeds himself: Sam Ecoff.

Sam Ecoff is a graduate of the University of Wisconsin in Eau Claire where he studied music theory and composition. Sam saw his first synthesizer at age 8 (a Moog IIIp!) and was completely entranced. The spell hasn't been broken yet. Today, he volunteers much of his spare time at local elementary schools to expose children to the wonders of synthesizers.

When Sam's days are not being devoured by writing books, he is composing music for television using his synthesizers. Although he has a great love of vintage instruments, his studio is stocked with many of the latest synthesizers and samplers.

Sam's great passion in life is teaching young people about music, particularly about synthesizers and music technology. He teaches applied piano and music technology lessons at a private conservatory in southeastern Wisconsin, and this book was originally written for his private students. He hopes to one day teach music technology classes at a small university interested to build a music technology program from the ground up.

When asked for a quote, he provided two: "96 simultaneous channels of synthesizers is really *not* enough," and "No, I don't think it is an unreasonable goal to attempt to collect one of every synthesizer ever made. What do you think I am going to do in retirement? I am going to open a museum and let the kids in to play these things!" At age 25, with plans for six more books in the *Fundamentals of Music Technology Series*, he is a long way away from retirement.

Appendix One:

Additional Resources

Books To Read

Digital X Book Amsco Publications
Complete MIDI 1.0 Detailed Specification, The by The MIDI Manufacturers Assoc.
Electronic Projects for Musicians by Craig Anderton
Keyfax, Omnibus Edition by Julian Colbeck
Making the Ultimate Demo Ed. Michael Molenda
MIDI Book, The by Steve De Furia
MIDI for the Professional Lehrman and Tully
MIDI Implementation Book by Steve De Furia
MIDI System Exclusive Book by Steve De Furia
MIDI, The Ins, Outs, & Thrus by Jeff Rona
Modern Recording Techniques by Huber and Rubenstein
Practical Recorsig Techniques by Bruce and Jenny Bartlett
Secrets of Ananlog and Digital Synthesis, The by Steve De Furia
Synthesizer Basics by Dean Friedman
Vintage Synthesizers by Mark Vail
What's MIDI? By Jon F. Eiche
What's a Sampler? By Jon F. Eiche
What's a Sequencer? By Jon F. Eiche
What's a Synthesizer? By Jon F. Eiche

CDs To Listen To

Tori Amos	The Choirgirl Hotel
	To Venus and Back
Björk	Homogenic
	Post
	Debut
Thomas Dolby	The Golden Age of Wireless
	Retrospectacle
	Astronauts and Heretics
	Aliens ate my Buick
Depeche Mode	Ultra
	Violator
	Songs of Faith and Devotion
	The Singles 81>85
	The Singles 86>98
Electronic	Raise the Pressure
	Electronic
Enya	The Memory of Trees
Erasure	I say, I say, I say

	Cowboy
	ABBA-esque
Garbage	Garbage
	Version 2.0
Human League	Octopus
Information Society	Hack
	Peace and Love Inc.
Madonna	Ray Of Light
	Peace and Love Inc.
New Order	Substance 1987
	Republic
Nine Inch Nails	The Downward Sprial
	The Fragile
Pet Shop Boys	Discography
	Bilingual
	Very (or Very Relentless)
	Alternative
	Behavior
Wendy Carlos	Switched on Bach
	Switched on Bach 2000
Venga Boys	The Party Album

Periodicals

Electronic Musician
EQ
Future Music
Mac Week
Mac World
Keyboard
Mix

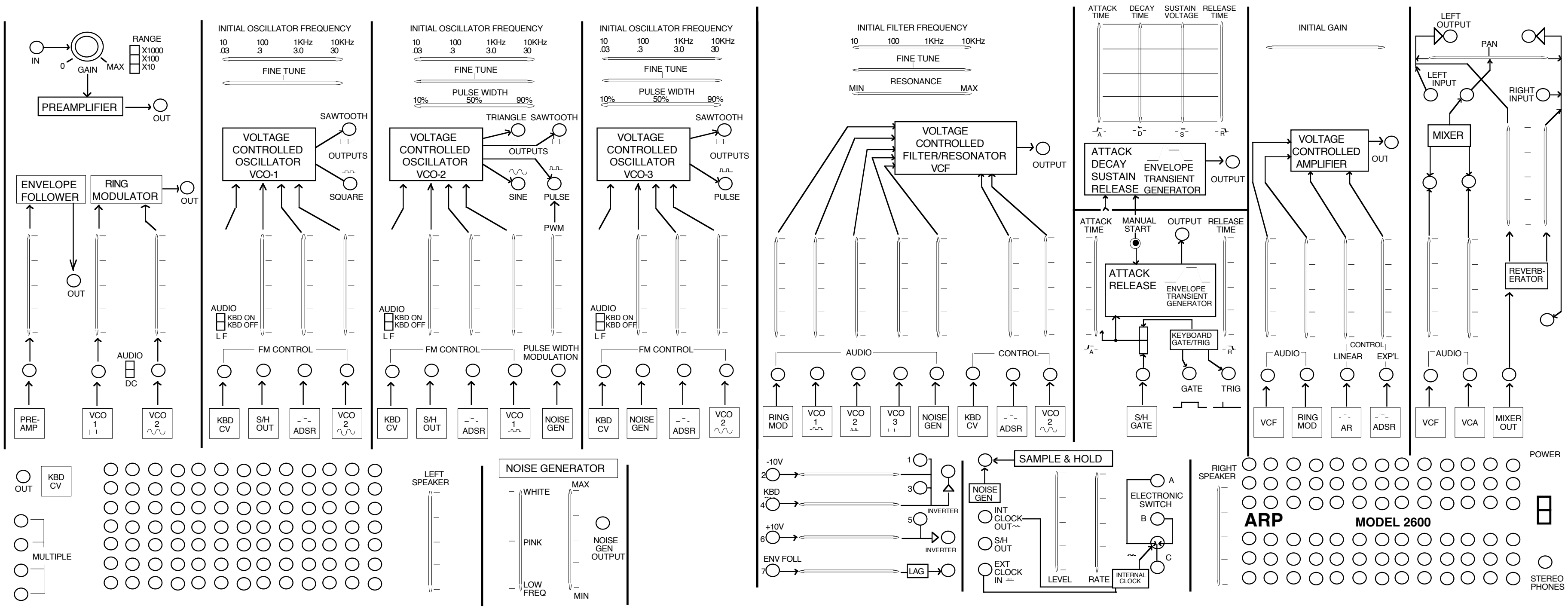
Web Pages To Visit

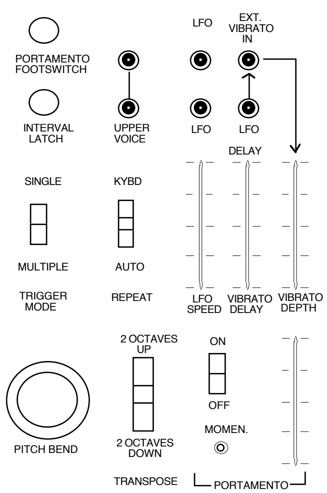
Apple Computer	http://www.apple.com/
Mac OS Rumors	http://www.macosrumors.com/
Vintage Synth Explorer	http://www.vintagesynth.com/
The Synthesizer Picture Archive	http://www.headcleaner.com/synths/
Synth Museum	http://www.synthmuseum.com/
Harmony Central	http://www.harmony-central.com/
Synth Zone	http://www.synthzone.com/
ThE aLpHa-DiaL	http://www.home.sol.no/`trukrist/a_juno/a_juno.htm/
Mark Glinsky	http://www.magicnet.net/`mglnsky/msg4.html
Super JX page	http://members.iquest.net/`twighlight/superjx.html

Yamaha Synths	http://www.obsolete.com/120_years/machines/yamaha/index.html
Temple of CZ	http://pw2.netcom.com/~bladez/Temple/index.html
E-music DIY archive	http://aupe.phys.andrews.edu/diy_archive/ (ARP 2600 Manual Here!)
Synth information	http://drum.warwick.ac.uk/music/drum/synth.html
The Easy VZ page	http://www.room101.co.uk/easyveeveezee/
JP8K page	http://www.geocities.com/SunsetStrip/Underground/Mezzanine/2252/
Keyboard Magaine	http://www.keyboardmag.com
PAiA Electronics	http://www.paia.com
MOTU	http://www.motu.com
Opcode	http://www.opcode.com
Yamaha	http://www.yamah.co.uk/
Rogue Music	http://www.roguemusic.com/
Music Recylcer	http://music.recylcer.com
Gearheads	http://www.jade-v.com/~gearhead/
Gasstation	http://www.gasstation.com
Roland	http://www.rolandus.com

Other Key Words To Try In Searches:

Kawai
Roland
FM Synthesis
Physical modeling synthesis
Synthesizer
Kurzweil
Emu-ensoniq
Subtractive synthesis
Oberheim
Rebirth
Peavey
Moog
ARP
Fatar
Mackie
Emulator
Tascam
Digidesign
Spectrasonics
Alesis
JL Cooper





Patch Name:

Date:

Created By:

Comments:

Appendix 2: ARP 2600 Patch Sheet

Some Final Words of Wisdom Quoted Directly from the ARP 2600 Owner's Manual:

"Don't forget to TURN ON THE SYNTHESIZER.
Often this is the reason why you get no sound out of it."

BY USING THIS CD, YOU INDICATE YOUR ACCEPTANCE OF THE LEGAL TERMS ON PAGE ii OF THIS BOOK.

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