The Complete Guide to Synthesizers

by Devarahi

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John Duesenberry and Ken Perrin,
and to the memory of the
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Preface

This book has been written for the musician or student who knows nothing of either synthesis or electronics and who wants to understand how analog synthesizers work. It is a combination of theoretical explanation and hands-on experience: 99 experiments are offered and discussed in detail.

All instantaneous sounds, no matter how complex, can be defined in terms of only three parameters: pitch, timbre (tone quality), and loudness. Analog synthesizers consist of units (or modules) that create these parameters separately and combine them into one final result. The VCO creates waves that sound unlike one another because of their differing harmonic content. The original sound (timbre) of the wave may be modified by the VCF. The wave's loudness is determined by the VCA. The way these three parameters vary over time (as opposed to instantaneously) is typically governed by the settings of envelope generators.

Synthesists build a rich palette of sounds by varying the controls on their VCOs, VCFs, VCAs, and envelope generators. This is done by a process known as voltage-control, the most critical concept necessary to an understanding of analog synthesis. Voltage-control is typically the change of pitch, timbre, or amplitude over a period of time that is caused by a changing electrical signal.

The bulk of this book is concerned with helping you to understand the preceding two paragraphs.

Chapter One of this book is designed so as to have you making sounds immediately, and is meant to be completed in about an hour. That doesn't mean you'll understand much in that hour. It does mean you'll be able to make your synthesizer sound like a typical synthesizer rather quickly. If you want to randomly move attenuators, push buttons, etc., please do; if you have creativity and enthusiasm but lack knowledge you can end up with some interesting effects. However, once you decide to begin Chapter One, please make a pact with yourself to follow it *exactly* until you have finished it. If in the middle of the chapter you get side-tracked by all the possibilities of some spacey sound, you'll find it very hard to come back and understand where you were. In other words separate, at least at the start, those times when you just want to intuitively play from those times when you want to learn in a rational manner. Both are important and necessary and hopefully by the end of this book you will have integrated them. Of necessity the book emphasizes the rational and linear mode of thought, but just "being with the synthesizer" is equally important. It's just that at first you'll learn more quickly if you do one *or* the other at a given time.

You will learn in paragraphs 6 and 29 of Chapter One what it means to say the synthesizer is "in neutral." Beginning with Chapter Two, unless otherwise directed, be sure that the synthesizer and keyboard are in neutral before you begin any experiment. If they are not you may get confused, because some effect you hadn't planned on may be happening.

Chapter Two introduces you to the important concepts of signal routing, block diagrams, wave parameters, and basic voltage control. In Chapter Three you'll meet the modules common to all synthesizers; the VCO, VCF, VCA, and envelope generators. Because these are so basic they are discussed at great length. Chapter Four is a short discussion of approaches to synthesis. Other modules that synthesists typically use are discussed in Chapter Five. Chapter Six, the driest and most technical of the chapters, briefly covers amplitude and frequency modulation. These techniques are used only occasionally[1] in the production of tonal music but frequently in the production of atonal music. Chapter Seven discusses the hardware, giving you information about most commercial synthesizers.

Beginning synthesis students commonly make three mistakes: 1) They fail to pay strict attention. If the instructions tell you to insert a plug in an attenuated input and you insert it in an unattenuated input, you're going to get a very different effect from the one intended. Please pay attention. 2) They fail to plug the ends of patchcords *tightly* into jacks. If you don't get sound, check those connections. 3) They don't open (generally by raising or turning to the right) attentuators. A closed attenuator is like a closed bathtub faucet; it will not let anything pass. All attenuators in signal and control paths must be open. Check them if you get no sound.

This book is meant to be universal and its contents accurate for almost all analog synthesizers. Because the many experiments show a wide variety of ways in which synthesizer modules may be used, you will be able to do more experiments—have more flexibility—if your synthesizer is patchable. However, few synthesists will have all the modules necessary to do each experiment. If you can't do a particular experiment because your synthesizer doesn't have the necessary flexibility, just read the description and study the patch: You'll at least understand the point of the experiment. As a reference point I have assumed you have available a one-voice synthesizer with one or more VCOs, a low-pass VCF, a VCA, envelope generator, LFO, and a keyboard.[2] No matter what synthesizer you have, or even if you have no synthesizer at all, you will understand the secrets of these exciting new instruments when you have completed this book. You will be able to look at a

[1] FM techniques are commonly used in instrumental simulation with computers, but such computer simulation is rare at this time.

[2] A word of warning: because synthesizer manufacturers have generally decided to produce keyboards, it becomes easy to think of a synthesizer as a keyboard instrument. Such thinking places an unnecessary limitation on synthesis and needlessly excludes non-keyboardists. *One need not be a piano player to be a synthesist.* The only reason a keyboard is needed is to play melody in real time (and even then it's not needed if you have one of the devices described in Chapter Five, section III-G). To fully appreciate the capabilities of a synthesizer you must expand your definition of music to encompass all of sound; don't limit yourself by thinking you must have or play a keyboard.

synthesizer you've never seen before and, within a short time of studying its face panel, understand what it can and cannot do. You will know how to create the effects you want. If you have accidentally created a great sounding patch you'll be able to go back over it and understand what you did. You will understand the advantages and disadvantages of monophonic and polyphonic, patchable and hard-wired, and analog and digital synthesizers.

Some of the experiments in this book require more than the basic aforementioned modules; in such cases the word NEEDED appears at the beginning of the experiment, and following it are listed the additional modules needed to complete the experiment. Those NEEDED modules are found on many, but not all, patchable synthesizers. You will be able to do many of the experiments in this book, and you can still learn how synthesizers work, even if your synthesizer has only the basic modules and is not patchable. In fact, this book will teach you how synthesizers work even if you have no synthesizer at all.

Obvious differences in makes of synthesizers are frequently pointed out in footnotes, but I cannot point out every difference in every synthesizer; that's why you should carefully read the manual that came with your synthesizer. This book is complementary to that manual.

The ARP 2600 is a patchable synthesizer widely used in teaching environments, and it has many features unique to it. For this reason Appendix A deals particularly with the 2600, and there are numerous references to it throughout the book (such as that on page 16, which directs the reader to section I of Appendix A).

There are *many* more modules available, both voltage- and manually-controllable, than those discussed in this introductory book. Serge-Modular's analog shift register, Aries' complex electronic switches, E-mu's voltage-controllable envelope generator timing parameters, and Polyfusion's many keyboards are but a few examples.[3] Generally these modules are compatible with the equipment of competitors, but check to be sure. Familiarize yourself with the products of these and of other companies.

I recommend thorough and constant review of this book as you learn. When you have finished Chapter Two your understanding will increase if you re-do Chapter One. When you complete section II of Chapter Three go back and review section I, as well as Chapters One and Two. In this way you will be able to integrate the many things you'll be learning, and to understand them as part of a totality rather than as isolated concepts and modules.

A final note: your reason for being interested in synthesizers is most probably the creation of music, whether tonal or atonal. The concepts you'll be learning in this book are of necessity scientific in nature. As much as possible I've given examples of how to translate that scientific understanding into musical value, but this book cannot teach you to be creative. It can only give you the tools with which to broaden your own artistry. If you let go of any limitations you may have placed upon yourself by prior definitions of what music is or what a synthesizer sounds like, your life can be greatly enriched, as mine has.

ACKNOWLEDGMENTS

I would like to publicly acknowledge and thank some of those people who have helped in the completion of this book:

My parents, Sybil and Stanley Rappaport, for a lifetime of devotion and support;

Yana Breeze and Michael Shaw, for constant encouragement and love;

[3] Addresses of synthesizer manufacturers appear on page 190. Almost all will be happy to supply you with a free catalogue. Write to them.

Clark Spangler and David Phillips, for teaching me how to simulate instruments;

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and especially Noodle Pudding, for being.

Devarahi
October, 1981

chapter one

Becoming Familiar With A Synthesizer

This chapter is a series of numbered paragraphs, most of which are in two parts. The part of the paragraph in CAPITAL LETTERS indicates something you are physically to do with the synthesizer. The part that is in lower case type is a beginning and brief explanation of what's happening. *You are not expected to understand this brief explanation of theory at this time.* If you get it, wonderful; if you don't, don't worry. Rather than just moving controls having no idea what they do, or dealing immediately with technical theory that won't seem like fun, you will get some theory along with hands-on application. Everything will be explained in greater detail later on. When you have completed this chapter you should have created some typical synthesizer sounds. Let's go!

1. If possible listen to the synthesizer through an external PA system. Internal speakers sometimes make such a system unnecessary, but to appreciate how a synthesizer can sound, you will find an external PA extremely helpful.

   If your synthesizer has one final output jack (sometimes called "Line Out" or "Audio Out") and your amplifier is monaural, or if your synthesizer has stereo outputs and your amplifier is stereo, connections between the two are obvious. If you have access to a monaural amplifier and your synthesizer has stereo outputs, connect one synthesizer output jack to the input of your amplifier and slide any "Pan" control toward that output jack.

2. With each experiment in this book, and with each new step of Chapter 1 (which is itself one major experiment), there will be a *block diagram,* a symbolic representation of what is happening with

the different parts of the synthesizer. Block diagrams are explained in the first part of Chapter 2. Don't expect to understand them immediately. Study them in the context of the connections you have actually made on the synthesizer and they will become clear.

3. Important words or concepts will be *italicized* the first time they are used. Try to get their meaning from the context in which they are used. There is also a glossary at the back of the book, and every new word that is italicized appears there.

4. Every control, every attenuator on the synthesizer, has some effect. Therefore it is important that any time you begin work with the synthesizer you know where every control is. Otherwise an attenuator or switch might be changing the sound you expected to get. Fortunately there is a way to make sure that no control will secretly affect what you are doing and confuse you.[1]

5. MAKE SURE THE POWER IS *OFF*. PLUG THE SYNTHESIZER INTO AN ELECTRICAL OUTLET.

   Many synthesizers come with a 3-way grounded plug. If it is at all possible, the synthesizer should be plugged into an outlet that accepts this kind of plug. If such an outlet is unavailable, buy an adapter which allows you to insert a plug like this into a regular electrical outlet. Do not cut off the third (or grounding) prong from the synthesizer power cord in an attempt to plug the cord into a regular electrical outlet.

6. MAKE SURE THAT ALL THE CONTROLS THAT MOVE UP AND DOWN ON THE SYNTHESIZER ARE DOWN; MAKE SURE THAT ALL THE CONTROLS THAT MOVE TO THE RIGHT OR LEFT ARE TO THE LEFT; MAKE SURE THAT ANY SWITCHES (EXCLUDING THE POWER SWITCH) ARE UP OR TO THE RIGHT.

   EXCEPTIONS: SLIDE CONTROLS (ATTENUATORS) FOR THE SPEAKERS, IF ANY, SHOULD BE UP; A PAN CONTROL, IF ANY, SHOULD BE IN THE MIDDLE, UNLESS YOU ARE USING AN EXTERNAL MONAURAL AMPLIFIER (IN WHICH CASE FOLLOW THE DIRECTIONS IN PARAGRAPH 1); A CONTROL FOR A HIGH-PASS FILTER, IF ANY, SHOULD BE ALL THE WAY TO THE RIGHT OR UP.

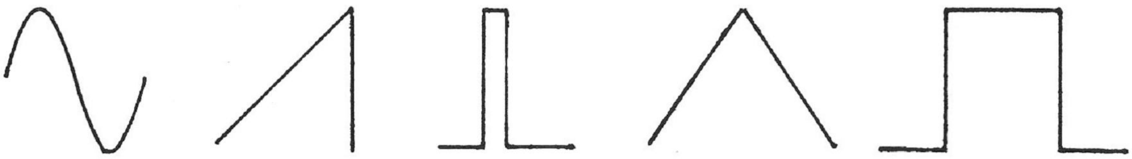
7. When the synthesizer is in the configuration described in paragraph 6 we say it is "in neutral," just like an automobile. It's ready to go.

8. TURN THE POWER SWITCH ON.

9. Your synthesizer has one or more signal generators called *voltage-controlled oscillators* (VCOs for short). A VCO is a device that generates assorted waves of all *audio frequencies* (frequencies you can hear) and (often) *low frequencies* (also called sub-audio frequencies, or frequencies you generally cannot hear). AF is an abbreviation of audio frequency, LF of low frequency. You can decide which fre-

[1] These instructions are generally true. However, you should read your synthesizer's manual in the event of particular differences. If your synthesizer has microprocessor-assisted programmability (e.g., Sequential Circuits Prophet 5, Oberheim OBX-a, Roland Jupiter 8, Yamaha CS40-M, etc.), it should be in the non-preset mode (Manual Mode in the OBX-a and Oberheim OB-1).

quency a particular VCO will generate. A VCO cannot generate more than one fundamental frequency at any given instant in time; that is, it cannot generate two fundamental frequencies simultaneously, although each VCO can generate at least two different periodic waveforms (which have the same fundamental frequency) simultaneously. The waveforms that synthesizers generate are called sine, sawtooth, pulse, triangle and square; their symbols, respectively, are:



Most voltage-controlled oscillators have two associated attenuators that control the initial frequency at which the oscillator will generate a wave and, depending on that frequency, you will hear either a pitch (for example, middle C) or, if the oscillator is in low-frequency range, an event which can recur again and again (for example, periodic clicks).

10. LOCATE A VCO THAT HAS AT LEAST TWO DIFFERENT WAVE OUTPUTS (e.g., SINE AND SAWTOOTH). OPEN THE COARSE FREQUENCY ATTENUATOR OF THAT VCO, SO THAT THE VCO OUTPUTS A FREQUENCY OF ABOUT 300 HZ. (APPROXIMATELY ~~E~~ t ~~FLAT~~ ABOVE MIDDLE C.) MAKE SURE THAT IF THERE IS A RANGE SWITCH IT IS IN THE AUDIO FREQUENCY (OR KEYBOARD) POSITION.[2]

One attenuator associated with the initial frequency of the VCO is called the *coarse tuning* attenuator to distinguish it from the other attenuator associated with the VCO, the *fine tuning* attenuator. The former has a range of about 4 to 10 volts, and can cause pitch changes of 4 to 10 octaves. The latter generally has a range of less than one volt, and can cause a pitch change of less than one octave.

If there is a range switch, when it is in its audio frequency position the frequency generated by the oscillator will be within audio range, that is, you can hear it. When it is in the low-frequency (LF) position, the oscillator will generate low, sub-audio, frequencies.

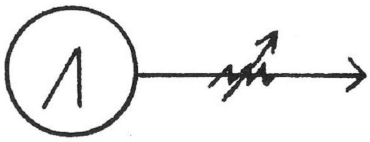
At this moment the VCO is generating as many waves as there are outputs—up to five: a *sine,* a *triangle,* a positive-going *sawtooth,* a negative-going *sawtooth,* and a *pulse,* each having a frequency of about 300 *cycles per second.* If the waves being generated were patched directly out you would be able to hear them.

11. Virtually all synthesizers have either an output mixer or a VCA as the last stage of the signal path. Whenever you are told to patch a wave "out," patch the wave into the output mixer or the VCA and open the attenuator associated with the jack into which you patched

[2] Hard wired synthesizers (Chapter 7, section I-B) usually have VCOs that generate only audio frequencies, and separate LFOs that generate only low frequencies. Thus their VCOs do not usually have range switches. Exception: Oscillator B on the Sequential Circuits Prophet 5 and Prophet 10.

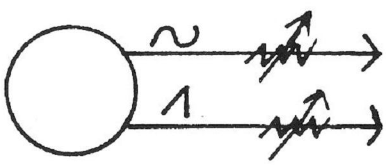
the signal. If the last module on your snythesizer is a VCA, you will have to open the gain attenuator of the VCA to let the audio signal pass through. The output mixer is generally the rightmost module on a synthesizer.[3]

12. IF IT IS PATCHABLE, YOUR SYNTHESIZER WAS SUPPLIED WITH SEVERAL PATCHCORDS, SHORT CABLES WITH "MALE" PLUGS AT EITHER END; THESE ARE USED TO CONNECT *OUTPUT* AND *INPUT* JACKS. OUTPUTS DO NOT CONNECT TO OTHER OUTPUTS; INPUTS DO NOT CONNECT TO OTHER INPUTS. IF YOU CONNECT AN OUTPUT TO AN OUTPUT OR AN INPUT TO AN INPUT, YOU CANNOT HURT THE SYNTHESIZER; YOU JUST WILL NOT GET ANY SOUND. PATCH AN AF SAWTOOTH WAVE OUT.



You should hear the sound of a sawtooth wave whose frequency is about 300 cycles per second. Note that the strength (*amplitude*) of the signal increases as the attenuator is opened. In general the more an attenuator is opened, the greater the amount of signal that can pass through it.

13. LEAVE THE PATCHCORD WHERE IT IS, BUT CLOSE THE ATTENUATOR. INSERT ANOTHER PATCHCORD INTO ANOTHER OUTPUT OF THE VCO AND PUT THE OTHER END INTO ANOTHER INPUT TO THE OUTPUT MIXER. NOW OPEN THE ATTENUATOR ASSOCIATED WITH THAT INPUT. FOR EXAMPLE:



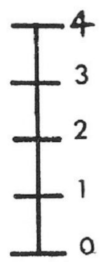
Note the difference in quality of sound between two different waves of equal frequency. Different types of waves have different characteristic sounds because they have different *overtones,* or *harmonics.*

14. ALTERNATELY OPEN AND CLOSE THE TWO ATTENUATORS TO HELP YOU COMPARE AND CONTRAST THE DIFFERENT SOUNDS. WHEN YOU FEEL YOU HAVE HEARD ENOUGH, REMOVE BOTH PATCHCORDS AND CLOSE BOTH ATTENUATORS.

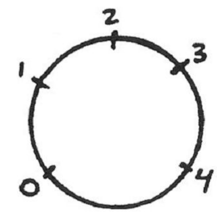
15. An attenuator has a wide operating range. When it is completely shut, it does not allow any signal to pass through; when it is completely open, it allows all the signal to pass through. At different points in the operating range it allows different levels of signal to pass through.

[3] If your synthesizer is hard-wired then you will have to open the filter and VCA as well to hear sound. You may also have to depress a key. Microprocessor-assisted programmable synthesizers do not have separate gain attenuators associated with their VCAs; to open those VCAs you'll have to open the D, S and R attenuators of the associated ADSR envelope generators.

There are times when you will want to have some specificity on how open the attenuator should be: wide open, halfway open, etc. The convention used in this book is as follows: all attenuators are assigned five numbers: 0, 1, 2, 3, and 4. If the attenuator setting is 0, it is closed; if it is 4, it is completely open. At 1 it is one-fourth open; at 2 it is one-half open; and at 3 it is three-quarters open.



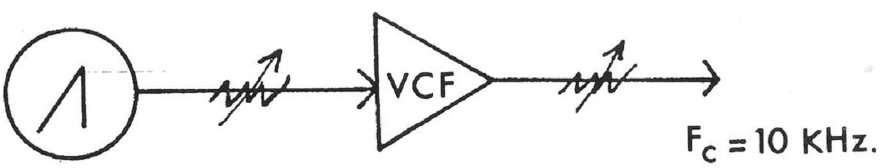
Unless otherwise stated, all attenuators appearing in a block diagram will be at maximum open (a level of 4). The only exception will be the final (rightmost) attenuator in the patch, which will control the final volume of the sound. Set that attenuator to your taste.



By way of example, right now most attenuators on your synthesizer are set at 0, although the attenuators associated with the speakers and high-pass filter, if any, are at 4, and the coarse frequency attenuator of one VCO is somewhere between 0 and 4.

16. One of the primary *modules* of any synthesizer is a *voltage-controlled filter* (VCF). A filter controls which harmonics of a wave are allowed to pass through to the final signal output. In order to have a VCF modify a wave, the wave must be input into the filter. This is done through any of the *audio* inputs to the filter. To get a filter to modify a wave from a VCO you could connect a patchcord from any VCO complex wave output to *any* audio input to the filter.

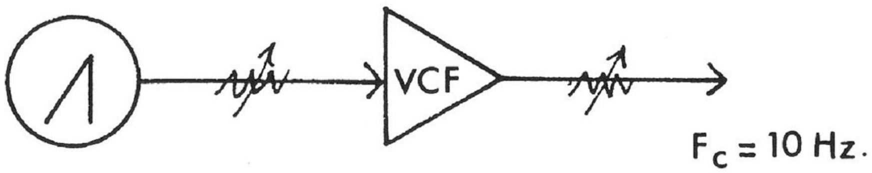
17. SET THE COARSE FREQUENCY OF A VCO TO ABOUT 300 HZ., JUST AS YOU DID BEFORE. CONNECT A PATCHCORD FROM THE VCO'S SAWTOOTH OUTPUT TO ONE OF THE AUDIO INPUTS TO A LOW-PASS FILTER. MOVE THE ATTENUATOR ASSOCIATED WITH THE FILTER, THE *INITIAL FILTER CUTOFF FREQUENCY ATTENUATOR,* ALL THE WAY TO THE RIGHT OR UP. THIS OPENS THE FILTER. PATCH THE FILTER OUT. NOW RAISE THE ATTENUATORS ASSOCIATED WITH BOTH INPUTS, THE ONE TO THE FILTER AND THE ONE TO THE OUTPUT MIXER, TO A LEVEL OF 3.



You should be hearing a sound similar to the one you heard before. At the same frequency one sawtooth wave, if unmodified, will sound like another. With the initial filter cutoff frequency attenuator all the way to the right or up, the low-pass filter is wide open, passing virtually everything that comes through it. Therefore it is not yet modifying the sawtooth wave from the VCO.

18. VERY SLOWLY CLOSE THE INITIAL FILTER CUTOFF FREQUENCY ATTENUATOR OF THE VCF. THE ATTENUATORS (VERTICAL OR ROTARY CONTROLS) ASSOCIATED WITH THE INPUTS

TO THE FILTER AND THE MIXER SHOULD BOTH BE PUT AT A LEVEL OF 4.



What are you hearing? Although you might think at first that the volume is decreasing, what is really happening is that the higher harmonics, the higher-frequency components, of the sawtooth wave are being removed—filtered—from the sawtooth. It is similar to turning down the treble control on a hi-fi amplifier. Listen to this effect several times. Of course, the total volume is in fact decreasing, but this is a secondary effect, caused by the loss of more and more of the high harmonics, and thus more and more of the total energy of the signal. Conversely, as you open the filter, more harmonics are allowed to pass through and the sound becomes brighter, richer, fuller. A filter changes the *tone color*—the *timbre*—of a wave.

19. When you opened and closed the initial filter cutoff frequency attenuator, you exercised *manual control* over the filter. This works just fine if you want to open and close the filter occasionally, but what if you wanted to do that evenly 6 or 7 times a second? You couldn't do it. Try it if you think you can.

However, the various waves of the synthesizer VCOs, when in their low frequencies, can control the filter, by a process known as *voltage control.* Voltages do the same thing you did manually, only much faster and more evenly than you could. Here's an example that will help you distinguish how manual and voltage control are similar and how they differ.

20. PUT THE RANGE SWITCH OF ANOTHER VCO IN ITS LOW-FREQUENCY (LF) POSITION AND OPEN THE FREQUENCY ATTENUATOR ASSOCIATED WITH THE VCO ABOUT TWO-THIRDS OF THE WAY.[4]

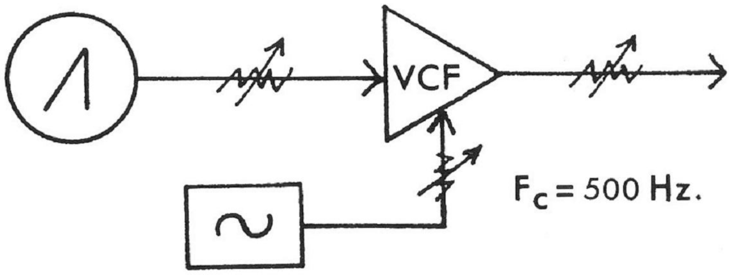
A VCO functioning as a generator of low-frequency waves is called an LFO (low-frequency oscillator). Thus the VCO set to generate a frequency of about 3 cycles per second is now an LFO. If the range switch of the VCO (if it has one) were in its audio-frequency position now, the frequency of the waves generated by the VCO would be about 1 KHz. (KHz. is defined in Chapter 2, section II.)

21. In addition to audio inputs, the filter has control inputs. Typically one of these is *unattenuated* and others are attenuated (that is, they have attenuators associated with the input jacks that control how much voltage gets through).

22. PATCH A LOW-FREQUENCY SINE OR TRIANGLE WAVE INTO AN ATTENUATED CONTROL INPUT TO THE FILTER. THE SINE WAVE

[4] If your synthesizer is hard-wired, open an LFO's attenuator about two-thirds of its range, and route it to control the filter.

IS NOW VOLTAGE-CONTROLLING THE FILTER. THAT'S WHY ITS CALLED A *VCF*. THE INITIAL FILTER CUTOFF FREQUENCY ATTENUATOR ASSOCIATED WITH THE FILTER SHOULD BE IN THE MIDDLE OF ITS RANGE. *SLOWLY* OPEN THE ATTENUATOR ASSOCIATED WITH THE CONTROL INPUT TO THE FILTER.

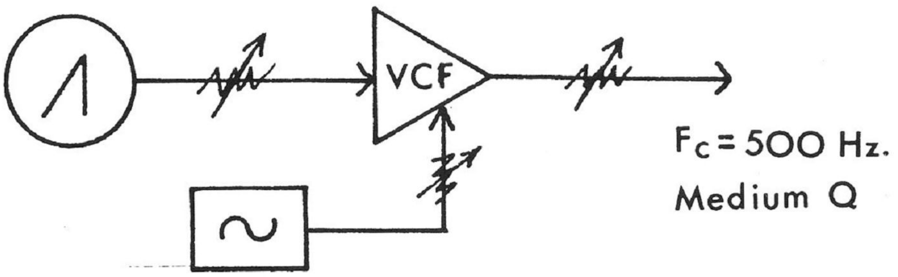


Notice how smoothly the sine wave is able to open and close the filter. It's similar to the way you did it manually, but more even.

23. NOW SLOWLY OPEN THE FREQUENCY ATTENUATOR OF THE LFO. THIS INCREASES THE FREQUENCY OF THE LFO, THE CONTROLLING OSCILLATOR. THE FILTER IS NOW BEING OPENED AND CLOSED AT A FASTER RATE AS YOU CONTINUE OPENING THE ATTENUATOR.

Here's the real beauty of voltage control. There is no way anyone could manually open and close the filter as fast or evenly as that.

24. FOR ADDED EFFECT EXPERIMENT WITH OPENING THE *RESONANCE* ATTENUATOR. DO IT VERY SLOWLY. THERE WILL BE A VERY MARKED CHANGE IN QUALITY WHEN YOU GET THE RESONANCE SLIDER QUITE FAR TO THE RIGHT AND THE FILTER STARTS TO OSCILLATE (OR APPROACHES OSCILLATION).



25. NOTE THAT THERE ARE FIVE *PARAMETERS* WE HAVE USED SO FAR, EACH OF WHICH WILL CREATE A DIFFERENT EFFECT.

1. Moving the coarse frequency attenuator of the VCO changes the pitch of the VCO.
2. Moving the frequency attenuator of the LFO causes the frequency of the filter being opened and closed to be raised or lowered, depending on whether you open or close that attenuator.
3. Closing the filter cutoff frequency attenuator causes filtering, the subtraction of harmonics.
4. The resonance attenuator causes the accentuation of a particular frequency component (harmonic) of the sawtooth wave.

5. The attenuator associated with the control input to the filter controls the depth of effect of the LFO.

26. Take your time with these five parameters. Become familiar and comfortable with what they do. Each one is important, not only for what it does but also because an understanding of the effect of each will help you understand what other modules do.[5]

27. TURN THE POWER SWITCH OFF. PLUG THE KEYBOARD INTO THE SYNTHESIZER, IF NECESSARY.

Never plug the keyboard into the synthesizer when the power is on; doing so could damage the synthesizer.

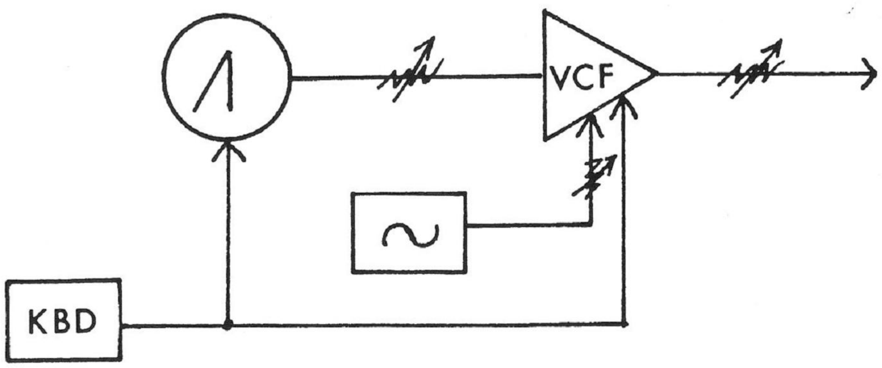
28. TURN THE POWER SWITCH BACK ON.

29. Just as it was important to put the synthesizer "in neutral" before beginning so that you were not surprised by unwanted effects, so too it is important to put the keyboard in neutral if it has controls besides just a keyboard.

LOOK AT THE KEYBOARD: ANY VERTICAL ATTENUATORS SHOULD BE ALL THE WAY DOWN; ANY *PITCH-BEND* DEVICE SHOULD BE IN THE MIDDLE; ANY *TRANSPOSE SWITCH* SHOULD BE IN THE MIDDLE; ANY *PORTAMENTO* SWITCH SHOULD BE OFF; ANY *TRIGGER-MODE* SWITCH SHOULD BE ON MULTIPLE; ANY REPEAT SWITCH SHOULD BE IN THE MIDDLE; ANY MODULATION WHEEL SHOULD BE ALL THE WAY TOWARDS YOU.

30. A synthesizer keyboard is a source of control voltage. In order for an oscillator to change pitch as you play notes on the keyboard, the voltage output by the keyboard must control that oscillator. Many synthesizers have hard-wired patches (installed at the factory) which automatically provide for keyboard control of VCOs (and VCFs).

31. IF YOUR SYNTHESIZER IS PATCHABLE, LOCATE THE KEYBOARD CONTROL VOLTAGE OUTPUT JACK. PATCH IT INTO ONE OF THE MULTIPLE JACKS (IF YOUR SYNTHESIZER HAS MULTIPLE JACKS; IF IT DOES NOT, YOU CAN BE QUITE SURE THAT THERE ARE HARD-WIRED PATCHES TO THE VCOs AND VCFs). PATCH ONE MULTIPLE JACK OUTPUT TO AN UNATTENUATED CONTROL INPUT TO THE VCO, AND ANOTHER MULTIPLE JACK



[5] You will be able to complete the steps in this chapter only if your synthesizer has patchcords; if it does not, you will still learn something by reading to the end of this chapter.

OUTPUT TO AN UNATTENUATED CONTROL INPUT TO A VCF. IN THIS CONFIGURATION THE KEYBOARD CONTROL VOLTAGE WILL CONTROL BOTH THE VCO AND THE VCF. AS YOU PLAY THE KEYBOARD HIGHER, MORE VOLTAGE WILL BE INPUT TO THE VCO, CAUSING ITS FREQUENCY (PITCH) TO RISE, AND TO THE VCF, CAUSING IT TO OPEN. CONVERSELY, AS YOU PLAY THE KEYBOARD LOWER, THE VCO'S PITCH WILL FALL AND THE VCF WILL CLOSE.

32. CLOSE THE LFO'S CONTROL ATTENUATOR INTO THE FILTER; OPEN THE FILTER; MAKE SURE THE ATTENUATORS ASSOCIATED WITH THE AUDIO INPUT TO THE FILTER AND THE INPUT TO THE MIXER ARE OPEN. NOW PLAY THE KEYBOARD A BIT.

33. At this point it seems like you are playing a regular keyboard; that is, the interval between C and D is one whole tone, between E and the next higher E exactly one octave, just as you would expect. This effect is misleading and is only one (albeit the most common) use of the keyboard: to control an oscillator at the rate of *1 volt per octave.*

The keyboard is, in fact, a *voltage controller.* We will have much more to say about that later.

34. You will also notice that you can't turn the sound off. This is because the filter is always open, always allowing the signal to pass. When you take your finger off a note on the keyboard, the synthesizer tracks and holds that note until you play another note, and then it tracks and holds that one. You need some way to turn the sound on and off: on when you play the keyboard, off when you stop. There are several ways to do this, and a common one is to use a *VCA* and an *envelope generator.*

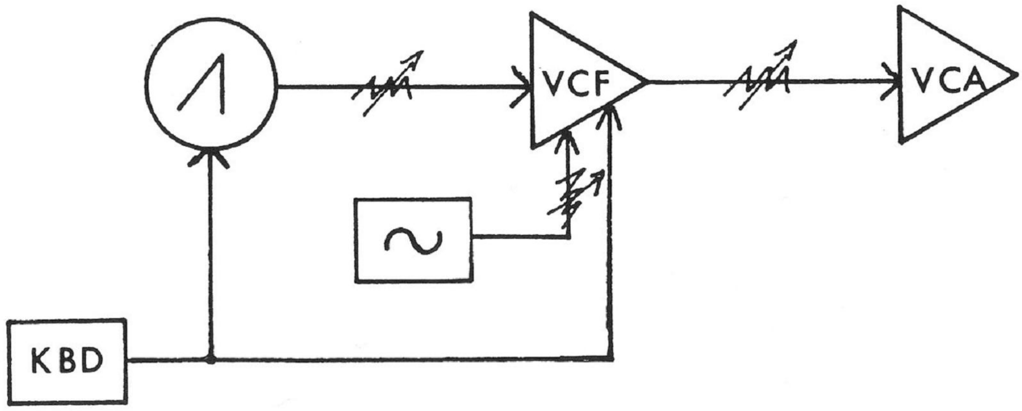
35. Many synthesizers have two types of envelope generators. The more versatile is the *ADSR,* the nature of whose output is determined by four vertical or rotary attenuators; the other is the *AR* generator, which has two vertical or rotary attenuators. An envelope generator typically might control a VCF or a VCA. Pushing down a key on the keyboard sends a *trigger* signal to the envelope generator,[6] telling it to start allowing the VCF or VCA to open. It also sends a gate signal which tells the envelope generator how long to output a control voltage. An envelope generator also determines the "shape" of the sound.

36. The VCA determines the strength of the signal which is output. If it is wide open, all signal input to it comes through; if it is closed, none comes through.

37. You are now going to change the route the signal takes to get out of the synthesizer. Instead of having it go directly from the filter out, you will take the signal coming out of the filter and put it through the VCA. Since it is not controlling the VCA, you must first put it in one of the audio inputs to the VCA.

[6] Actually the AR generator requires only a gate signal; no trigger signal is needed. The ADSR requires a trigger if it is to function as a four-stage envelope generator, but it will work as a three-stage envelope generator if it receives only a gate signal. Moog ADSRs do not require triggers. More on this in Chapter 3, section D.

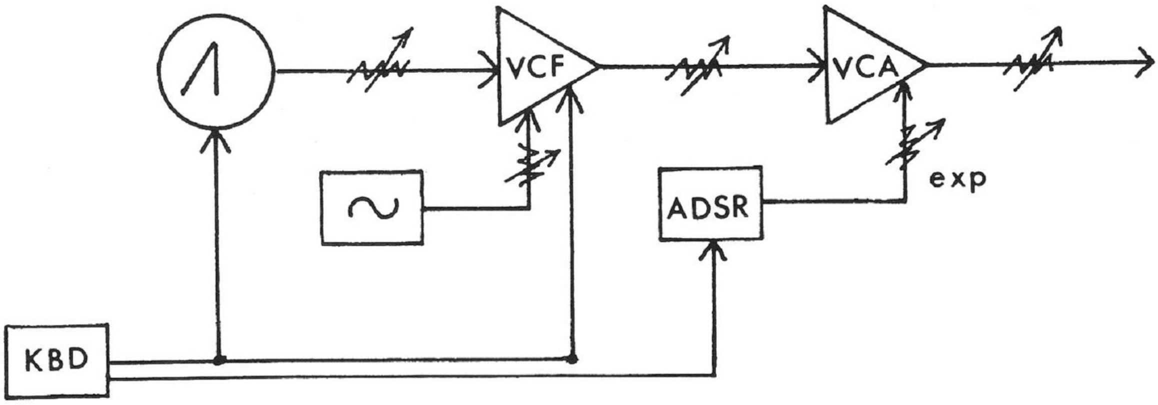
38. CLOSE THE ATTENUATOR OVER THE INPUT TO THE MIXER. REMOVE THE END OF THE PATCHCORD THAT IS IN THE INPUT TO THE MIXER AND PUT THAT PLUG INTO AN AUDIO INPUT TO THE VCA. THE FILTER OUTPUT IS NOW GOING INTO AN AUDIO INPUT OF THE VCA.



39. THE FOUR ATTENUATORS ON THE ADSR ENVELOPE GENERATOR HAVE TO BE SET IN SOME WAY TO SIGNAL THE VCA HOW TO OPEN AND CLOSE. HERE ARE SOME ARBITRARY SETTINGS: SET THE *ATTACK* ATTENUATOR HALFWAY TO MAXIMUM; SET THE *DECAY* ATTENUATOR A LITTLE HIGHER THAN THE ATTACK ATTENUATOR; SET THE *SUSTAIN* ATTENUATOR A LITTLE HIGHER THAN THE DECAY ATTENUATOR; AND SET THE *RELEASE* ATTENUATOR AT ABOUT THE SAME POSITION AS THE ATTACK ATTENUATOR. THE SETTING IS EXPRESSED NUMERICALLY AS 2,2½,3,2. (SEE PARAGRAPH 15.)

40. PATCH THE OUTPUT OF THE ADSR ENVELOPE GENERATOR TO THE EXPONENTIAL CONTROL INPUT TO THE VCA. OPEN BOTH ATTENUATORS, THE ONE ASSOCIATED WITH THE AUDIO INPUT TO THE VCA AND THE ONE ASSOCIATED WITH THE CONTROL INPUT TO THE VCA.

41. PATCH THE VCA OUT, AND OPEN THE LAST ATTENUATOR TO A LEVEL OF 3.



42. *NOW* PLAY THE KEYBOARD.

A VCA will not pass a signal unless there is *both* an audio signal and some control voltage coming through it. The audio signal from the VCF is always going into the VCA, but the control voltage doesn't go into the VCA until a key is pressed. This triggers the envelope generator (ADSR) and tells it to send a control voltage to the VCA. How that voltage will cause the VCA to open and close is determined by the ADSR attenuator settings. When the VCA detects a control voltage from the ADSR envelope generator, it opens up and the signal is allowed to pass. When the key is released, no more control voltage is sent, so the VCA closes and the signal from the VCF can't get through.

43. CHANGE THE SETTINGS OF ALL THE ATTENUATORS ON THE ADSR AND YOU WILL BEGIN TO GET A FEEL FOR WHAT AN ENVELOPE GENERATOR DOES. TAKE YOUR TIME WITH THIS. THERE ARE, AS YOU WILL SEE, LITERALLY HUNDREDS OF DIFFERENT POSSIBLE ADSR SETTING COMBINATIONS. IF YOU RAISE THE ATTACK ATTENUATOR ALL THE WAY UP, YOU WILL HAVE TO KEEP A KEY DEPRESSED QUITE A WHILE TO GET ANY SOUND.

44. WHILE PLAYING THE KEYBOARD, YOU MIGHT ALSO WANT TO EXPERIMENT WITH CHANGING THE FIVE PARAMETERS MENTIONED IN PARAGRAPH 25. CHANGING THEM TOGETHER WITH CHANGING THE ADSR SETTINGS CREATES A WIDE VARIETY OF POSSIBLE SOUNDS.

45. Hard-wired synthesizers have patches that have been installed at the factory; you may or may not use the patch, but you may *not* create a new patch that has not been provided for by the manufacturer. "Quasi-modular" synthesizers (Chapter 7, section I-C) have *pre-patches,* which are indicated by symbols like this one:

That symbol means that the manufacturer has hard-wired a patch from VCO-2's sine wave output to the module. You would not need to use a patchcord to get VCO-2's sine wave output at the module input; all you need do is open the appropriate attenuator and the signal will appear, *just as if a patchcord had been used.*

Pre-patches on patchable synthesizers are items of convenience: manufacturers generally create them where they are most used. For example, an LF sine wave is typically used to control the cutoff frequency of a VCF, so pre-patches of sine waves to VCFs are common. Don't think of the pre-patches as the "correct" patch; doing so will seriously limit your proper use of the synthesizer. A pre-patch is only one possibility.

A pre-patch is defeated (negated) any time a patchcord is inserted in the jack associated with the pre-patch. For example, if you patched a sawtooth wave from a different VCO into the jack above the symbol indicating that a sine wave from VCO-2 was pre-patched, that pre-patch would no longer apply. It would be defeated and you would hear the effect of the sawtooth wave rather than the sine wave. See Chapter 3, section I-B for more about hard-wired and pre-patches.

chapter two

Concepts Necessary To Understand Synthesizers

I. ROUTING SIGNALS AND BLOCK DIAGRAMS

A. Basic Block Diagrams

Oscillators (sometimes called "function generators") generate waves which, when in the audio range, need to get to your ears in order to be perceived. It is helpful, in the context of routing, to think of the wave as a quantum, a distinct entity, that goes from its origin to its destination. The easiest way to get that wave from an oscillator to your ear would be to send the signal directly from an oscillator output to the input of an amplifier, from the amplifier to a speaker, then through the air to your ear (Figure 2-1).

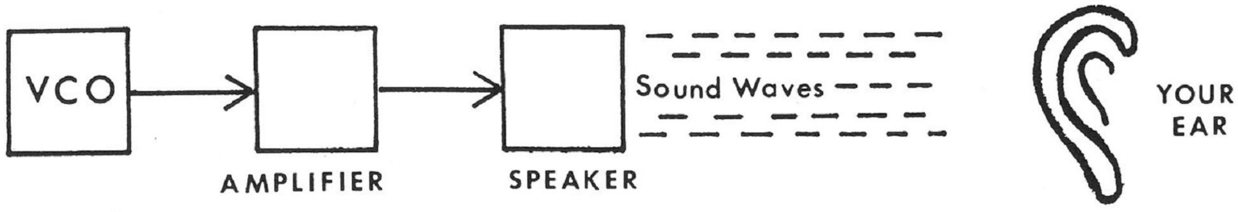
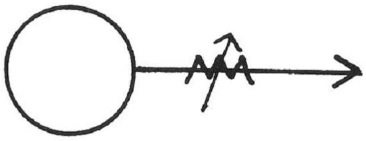


FIG. 2-1

EXPERIMENT #1: An unmodified wave

Patch any wave of a VCO in its audio frequency out, open the indicated attenuator, and you will hear that wave. To hear an audio frequency well, set the oscillator's coarse frequency attenuator so that the pitch you hear is about middle-C.



There are essentially three types of modules on all synthesizers: signal generators, modifiers, and controllers. Oscillators (and a noise generator) are the primary signal generators on all synthesizers. Filters, VCAs and mixers are typical signal modifiers. Keyboards, envelope generators, and LFOs are typical controllers. However, a module that generally serves one of these functions may in fact function very differently in a given patch (e.g., Experiment 28, where a VCF—typically a signal modifier—functions as a controller). *You must look to where a signal goes—not where it comes from—to know if it is functioning as a generator, modifier, or controller.*

The "map" of the route a signal takes is called a *block diagram*. In order to simplify understanding of block diagrams the following convention[1] will be used throughout this book:

1. Signal generators will be represented by a circle. These will typically be VCOs. If there is no drawing of a wave in the circle, then any AF wave may be used; if there is a drawing of a specific wave, then that wave is to be used. Audio-frequency waves should generally start in the octave between middle-C and high-C to be heard easily. If the circle has an N in it, a noise generator is the signal source.
2. Signal modifiers will be represented by an isosceles triangle whose apex is at the right.[2]
3. Controllers will be represented by a rectangle with the type of controller designated within the rectangle. If the controller is an LF wave, the waveshape will be in the rectangle. If it is an envelope generator, that will be indicated. If it is a keyboard, sample and hold, or whatever, that will be indicated within the rectangle.

   Note that, unless otherwise stated, a waveshape in a circle implies an AF wave; a waveshape in a rectangle implies an LF wave, unless otherwise stated.
4. Audio signals are diagrammed as leaving a module on its right and entering a module on the left.
5. Control signals are diagrammed as leaving a module on its right and entering a module at the bottom.
6. Keyboard control of a VCO or a VCF indicates the control voltage signal output by the keyboard; keyboard control of an envelope generator indicates the timing signals (gate and trigger) output by the keyboard.
7. An attenuator (a signal modifier that can cut the level of an input signal) in the signal path is represented by the electrical symbol for a variable resistor:

[1] From an article by Bugg, "Patch Notation: A Business Approach," *Synapse*, V. 2, no. 5 (March/April 1978), 37.

[2] This is the electronic representation of an operational amplifier (op amp), from which many modifiers may be derived.

Typically a wave is generated by an oscillator, sent through a filter (VCF) and VCA and then out to the world. Figure 2-2 is a block diagram of that signal routing.

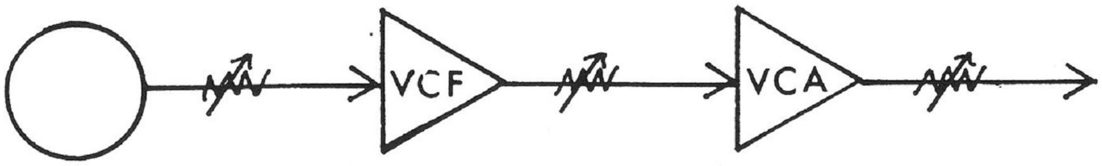


FIG. 2-2

In many cases (as you learned from the introductory chapter) a synthesizer has built-in hard-wired patches (Chapter 7, section I-B); in such cases patchcords are unnecessary, although the block diagram is the same. Examples of hard-wired synthesizers include the Minimoog, Arp Solus and Sequential Circuits Prophet 5. Some synthesizers have pre-patches (Chapter 7, section I-C); these are hard-wired patches that can be defeated by insertion of a patchcord at the appropriate jack. Examples of pre-patched synthesizers include the ARP 2600, Electrocomp 101 and Korg MS-20.

We could read Figure 2-2 as follows: "An audio signal originates in a VCO, goes out the VCO into the VCP, is (probably) modified in some way there and then goes out the VCF into the VCA, is (probably) modified in some way there and then goes out into the world."

The waves that oscillators generate can be used in two ways: as audio signals that will follow an audio path (as above), or as control signals, more commonly called *control voltages.* We will hear much more about control voltages, because voltage control is the most important distinguishing feature of contemporary *analog synthesizers.* In Chapter 1 (paragraph 22) you heard an example of a wave being used to control another wave when a low-frequency sine wave was used to control the opening and closing of a voltage-controlled filter.

Just as there are audio paths (as in Figure 2-2), so there are control paths. A block diagram of a low-frequency sine wave controlling a filter would look like Figure 2-3.

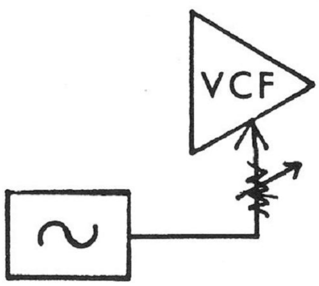


FIG. 2-3

A total block diagram may have both audio and control paths. As an example, let's combine the two we have already seen. The result is as in Figure 2-4.

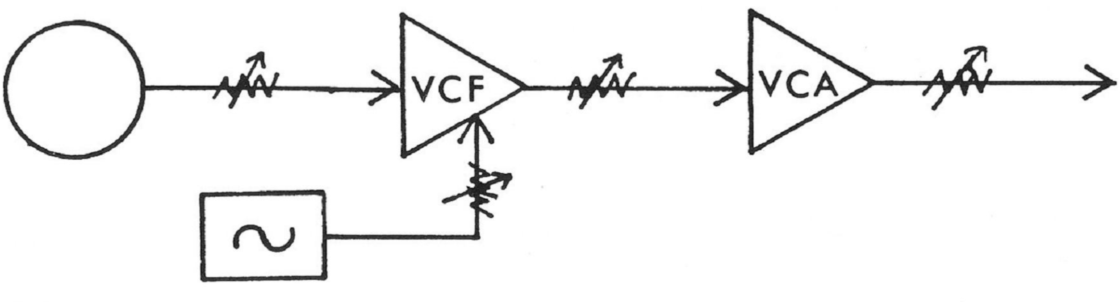


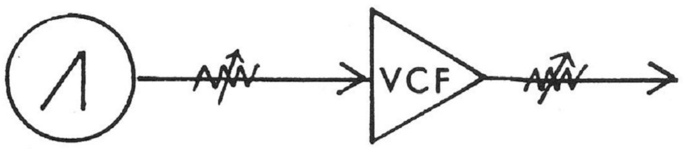
FIG. 2-4

B. Audio and Control Signals

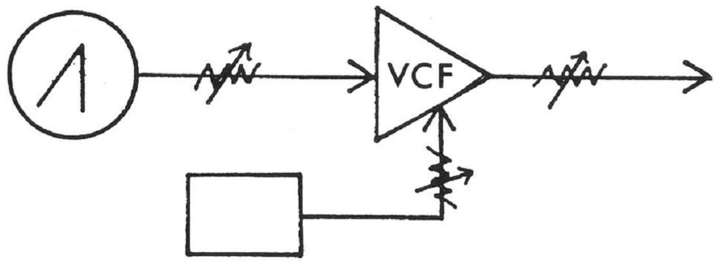
It is extremely important to understand the difference between an audio signal and a control signal. A voltage may be either, and there's no way of telling which it is by seeing where it comes from; you must see where it goes. *If it goes into an audio input to a module, then it is an audio signal and the module modifies it; if it goes into a control input, then it is a control signal and it controls the module output.*

EXPERIMENT #2: Audio vs. control signals

Patch an AF sawtooth wave from a VCO to any audio input to a filter. Open the filter wide (move its coarse filter cutoff frequency attenuator all the way to the right or up) and patch the filter out. You hear the sawtooth wave, essentially unfiltered. Now open and close the coarse filter frequency attenuator back and forth, and you hear the effect the filter has on the signal. The module (the filter) is modifying the signal.



Patch an LF wave into an attenuated control input to the filter. Put the LFO's frequency attenuator to minimum. Close the filter (put the coarse filter frequency attenuator all the way to the left or down). Now the LF wave will very *slowly open the filter. Give it time. Experiment with both attenuators in different positions to see what a wide variety of effects are possible. The LF wave is controlling the filter, opening and closing it at whatever frequency the LFO is oscillating. Become familiar with the different sound qualities of the filter modifying a wave, and of a wave (functioning as a control voltage) controlling the filter.*



VCOs do not have audio inputs; all oscillator inputs are control inputs. VCFs and VCAs have both audio and control inputs. If you intend to modify an audio signal by either the filter or the VCA, patch that signal into one of their *audio* inputs. If you intend to control either the VCF or the VCA by a signal of some sort, patch the signal into a *control* input of the VCF or VCA. In general, AF waves will go into audio inputs and LF waves into control inputs. Thus Figure 2-4 indicates that an output from a VCO is patched into an audio input to the filter, that the output from the VCF is patched into an audio input to the VCA, that a low-frequency sine wave is patched into an attenuated *control* input to the filter, and that the entire product is then output from the VCA through an attenuator to the world. *Whenever there is an indication that a signal goes "out," it means out through an input to the output mixer or the VCA, unless otherwise indicated.*

C. Information in Block Diagrams

A typical block diagram might look like this:

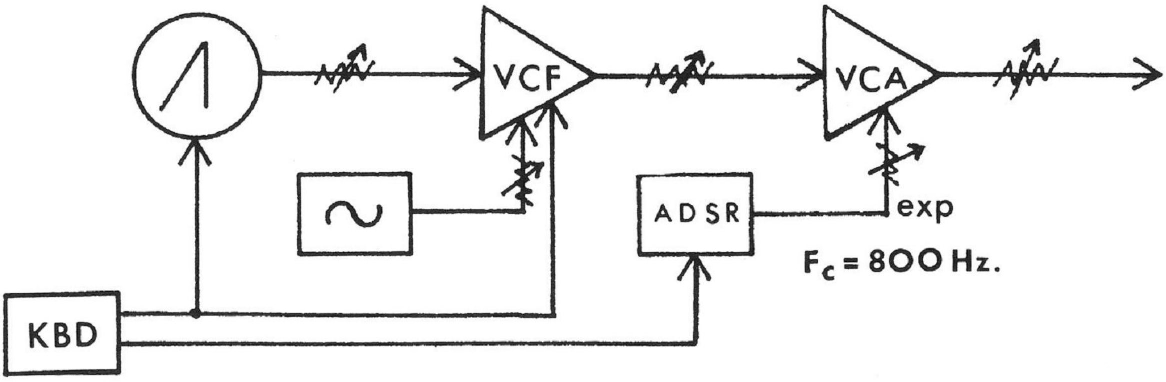
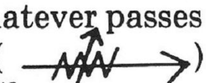


FIG. 2-5

Figure 2-5 is the same as the last block diagram in Chapter 1. This is the information given in Figure 2-5:

An audio-frequency sawtooth wave is going to be filtered, sent through the VCA, and then sent out. (This illustrates one good idea about reading block diagrams: always find the audio path first and get the general idea of what is happening. Although it is not always so easy to find that path on complicated block diagrams, in general, if you follow the horizontal lines you will find the audio path.) Since the VCO is controlled by the keyboard, what frequency (pitch) you hear will be determined by what note you play on the keyboard. (The pitch is initially determined by the coarse oscillator frequency attenuator associated with the VCO, and then the keyboard control voltage determines it from the initial setting. On many synthesizers the keyboard control voltage is hard-wired to an unattenuated control input to the VCO so no patchcord is necessary to get that control voltage to the oscillator.) The cutoff frequency of the VCF is also controlled by the keyboard (frequently via hard-wired patch). When a key is pressed, the keyboard simultaneously outputs a control voltage which determines oscillator frequency (pitch) and VCF tracking, and timing signals (a gate and a trigger) to the envelope generator (ADSR).

The trigger will tell the ADSR to fire,[3] that is, to start; the gate will tell the ADSR how long to go before stopping. The gate will be high as long as the key is pressed. When the ADSR receives this information, it will control the VCA, beginning to open it the instant it receives the trigger signal and allowing it to remain open as long as it receives the gate signal. This VCA opening will allow the sawtooth signal, as modified by the filter, to pass through and out. How the VCA opens will be determined by the settings of the sliders on the ADSR. Finally, the VCF, with an initial filter cutoff frequency setting of about 800 Hz., will be controlled by a low-frequency sine wave so that whatever passes through the VCF will have filter modulation to a depth of 2. The symbol () means that there is an attenuator in the path from the sine wave to the filter.

If you don't understand this now, don't worry. Envelope generators, unattenuated inputs, and other, related concepts will be gone into later. For now, just understand that a great deal of information can be conveyed in a block diagram. (See section I in Appendix A.)

II. WAVE PARAMETERS

Since waves and their modification are so important to audio synthesis you need to understand them. Waves are either *periodic* or *aperiodic* (not periodic). Periodic waves

[3] Exception: Moog ADSR, see Chapter 3, section IV-E.

recur again and again (see, for example wave 1 in the graph on page 18), and can be defined by their frequency, amplitude, harmonics, phase and voltage relationships. Aperiodic waves are discussed in section II-B of this chapter.

A. Periodic Waves

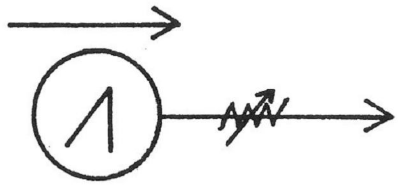
1. Frequency

Frequency refers to how often a periodic wave recurs in a given time period, usually one second. If a periodic wave repeats itself regularly 1000 times each second, it has a frequency of 1000 cycles per second. The phrase "cycles per second" has recently been called *Hertz,* after the German physicist Heinrich Hertz, an early investigator into the nature of electromagnetic waves. Saying 1000 Hertz or 1 Kilohertz is exactly the same as saying 1000 cycles per second. Hertz is abbreviated Hz.; Kilohertz is abbreviated KHz.

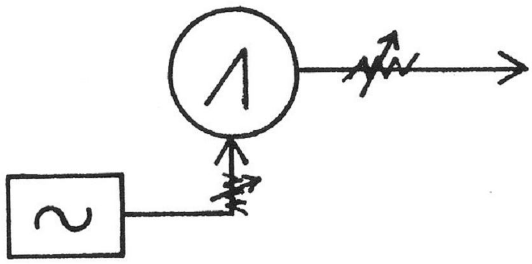
If a periodic wave has a frequency of between 20 Hz. and 20 KHz., it is the audio range and is thus an *audio wave.* It has a *pitch;* you can hear it. If it is below 20 Hz., you can perceive its *effect* on other waves but cannot hear it.[4] Such a *low-frequency wave* is typically used to control an audio wave.

EXPERIMENT #3: LF sine wave controlling a VCO

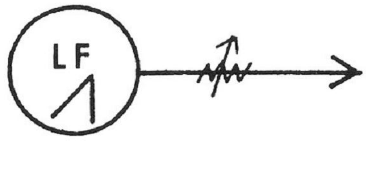
Turn the synthesizer on and make sure it is in neutral. *Patch an AF sawtooth wave out. Slowly open the coarse frequency attenuator associated with the VCO. This raises the frequency, which is perceived as raising the* pitch *of the wave.*



Now set the coarse frequency attenuator to high-C, about 500 Hz. Patch an LF sine wave to an attenuated control input of the AF VCO. Set the frequency of the LFO somewhat less than 3 Hz. Open the control attenuator into the VCO. You hear the pitch of the VCO rise and come back down to its starting point, and then fall and come back up to its starting point. You are hearing the effect of the low-frequency sine wave as it voltage-controls *the AF VCO. The frequency attenuator associated with the LFO will control the* rate of change *in the VCO's pitch. The control attenuator will control the* depth of change *(whether the deviation is up and down a semitone, 4 octaves, etc.).*



[4] If the low-frequency wave has an instantaneous voltage rise or fall, you *can* hear this as a discrete "click." E.g.:

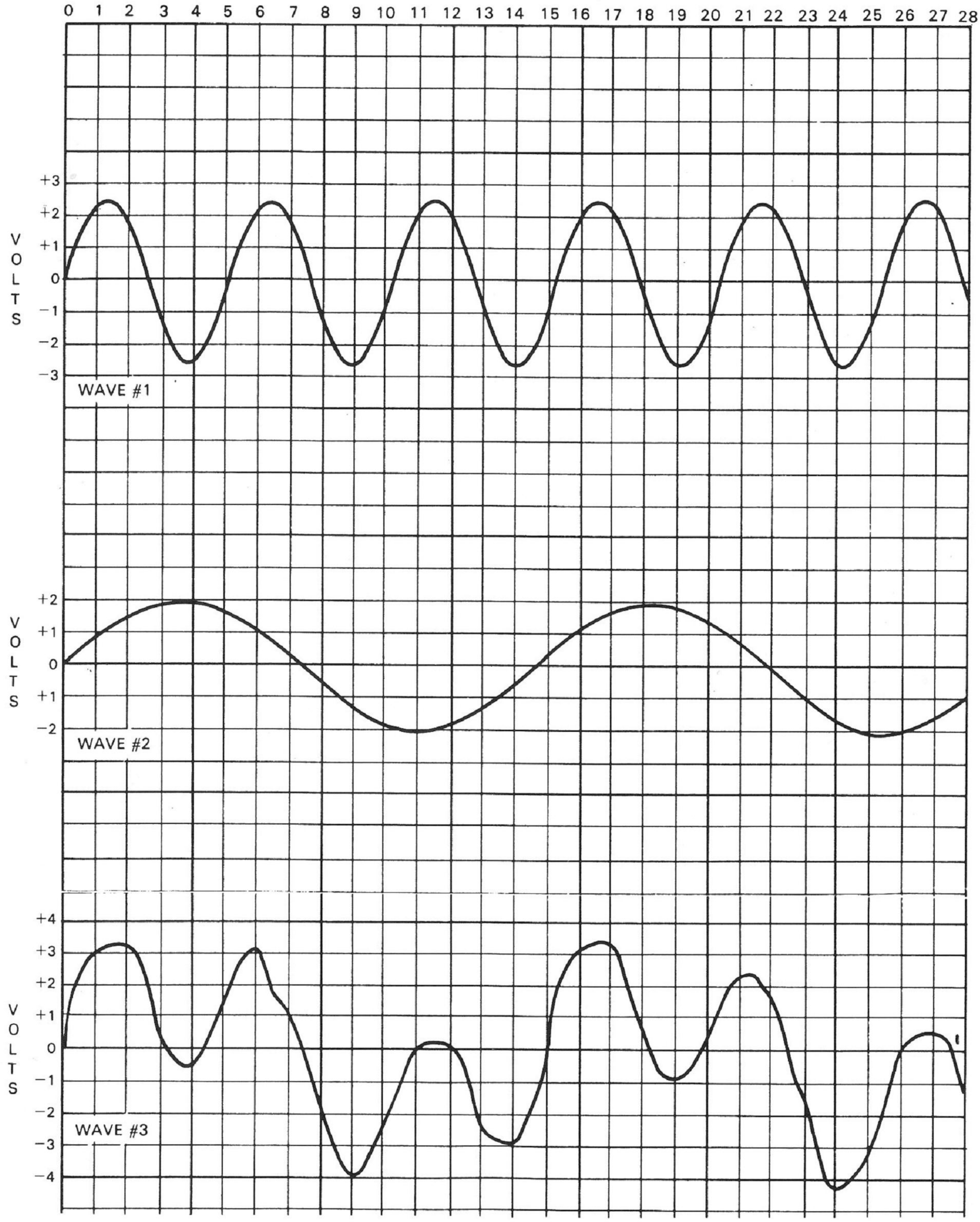


If you have them available, patch other LF waves to control the VCO. As you listen, see if you can define what is happening.

Frequency can be shown on a graph, as wave 1 is shown on this page. You can see that wave 1 recurs over and over again, so it is periodic. Assume that each square on the graph going horizontally represents one second. Can you see that the frequency of the wave is about once every 5 seconds? It goes through its entire cycle and returns to its starting point once every 5 seconds.

What is the frequency of wave 2 on the graph? Before reading on, look at the graph and come up with an answer.

If you said about once in 7 seconds, you made a common mistake. Seven seconds is how long it took for the wave to return to its original position, but it has gone through only half a cycle at that point. It must complete the full cycle, so its frequency is once every 14½ seconds. Both wave 1 and wave 2 are very low-frequency waves.



Look at wave 1 again. If the horizontal axis from numbers 1 to 28 represented just one second, then the wave would complete 5½ cycles in one second and its frequency would be 5½ cycles per second. If that same space on the horizontal axis represented 1/1000 of a second, then the wave would complete 5½ cycles in 1/1000 of a second, which is equivalent to a frequency of 5500 cycles per second, or 5.5 KHz.

2. Amplitude

Amplitude refers to the strength of a signal, either audio or control.

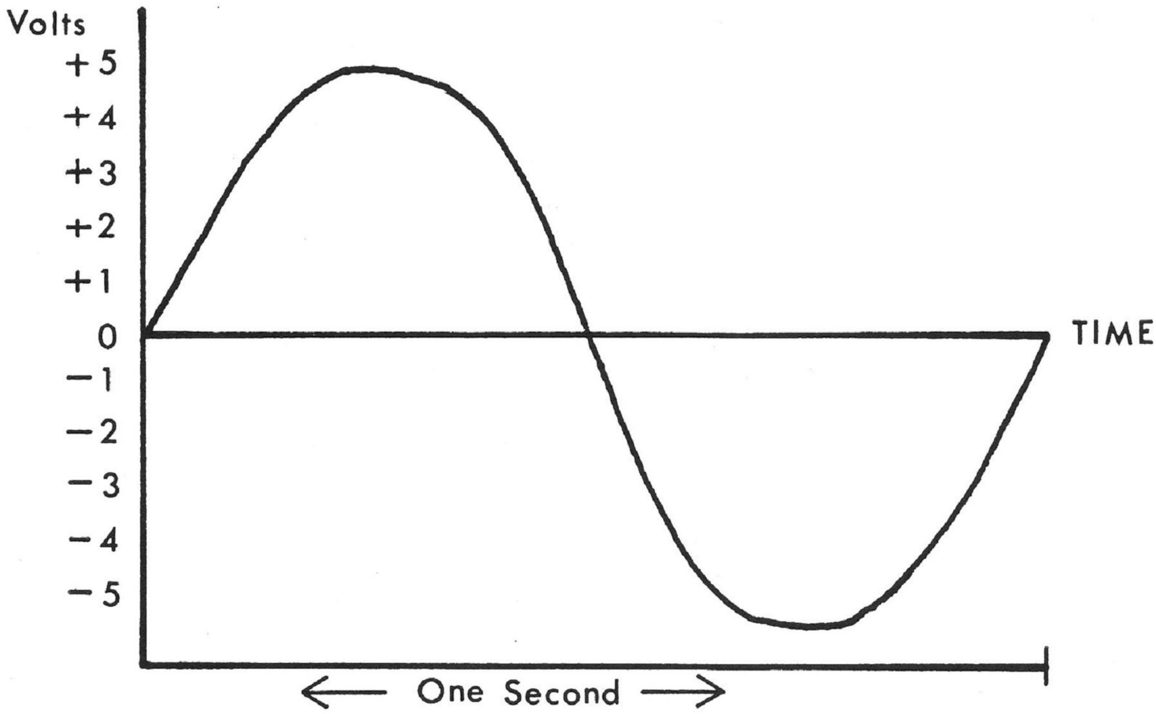
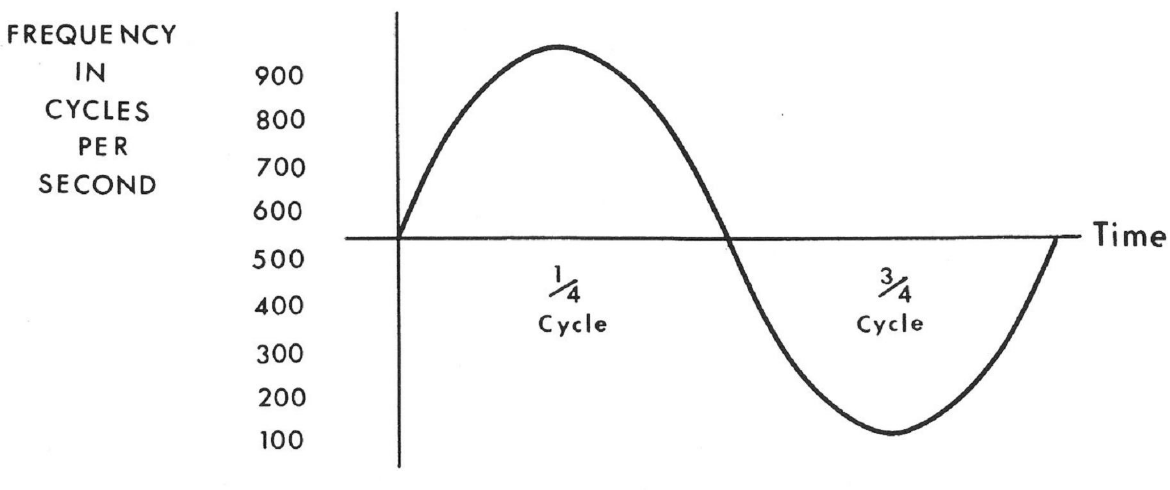


FIG. 2-6

Figure 2-6 shows one complete cycle of a sine wave; its frequency is 1 cycle per second. The wave rises, and its strength increases for the first quarter of a second, then it drops to its starting point—its strength decreases—for the next quarter-second. In the third quarter-second the wave drops to its maximum negative point, and finally it rises back to its original strength again, to commence another cycle. *The concept of negative value here simply means equal and opposite.* You heard the effect of such a sine wave when a low frequency sine wave controlled an AF VCO in Experiment #3. We could show the effect of the sine wave in that experiment by plotting the hypothetical frequency of the VCO against time (Figure 2-7).

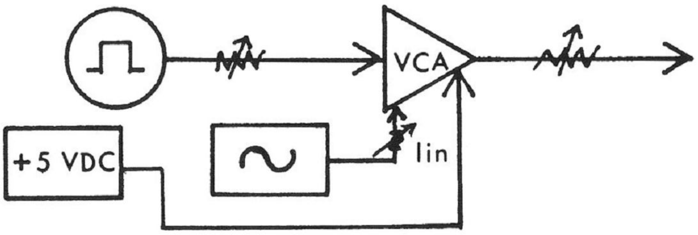
FIG. 2-7



The VCO's frequency rose for the first quarter of the cycle and dropped to its original position in the second quarter; in the third quarter the frequency (pitch) of the VCO dropped by an amount equal to the amount it rose in the first quarter of the cycle, and finally it returned once again to its starting point.

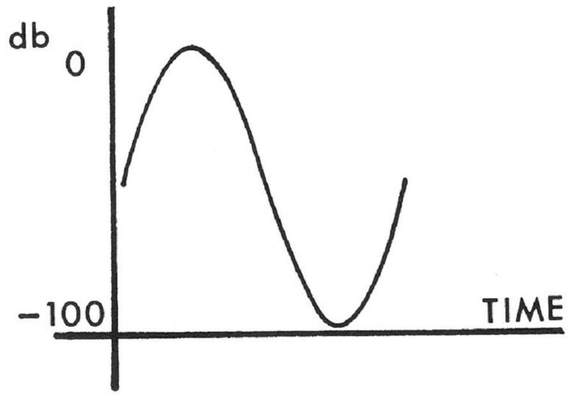
In Chapter 1 you heard the effect of a low-frequency sine wave controlling a VCF; earlier in this chapter you heard its effect on a VCO; now hear what happens when it controls a VCA.

EXPERIMENT #4: Biased LF sine wave controlling a VCA (volume)



Patch an audio frequency square wave to an audio input to the VCA. (Did you remember to put the synthesizer in neutral first? Did you set the coarse frequency attenuator to some audio frequency you will easily hear?)

Now patch an LF sine wave to the linear control input to the VCA. For reasons that will be explained in Chapter 3, section III, open the "gain" control associated with the VCA about halfway. (This is "biasing the sine wave up 5 volts"; you will learn more about this in Experiment #74.) Raise the appropriate attenuators.



You should hear the audio signal from the VCO rising and falling in volume, as its amplitude is controlled by the biased LF sine wave.

This is another example of voltage control. By varying the frequency of the LFO you change the rate at which the VCA opens and closes. You might be able to raise and lower the volume of a signal once or twice a second, but you couldn't do it evenly six or ten or more times a second. That's the beauty of a voltage-controlled amplifier.

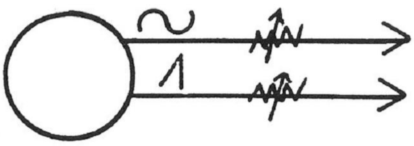
3. Harmonics

EXPERIMENT #5: Comparison of harmonics of different waves

Set the frequency of a VCO which has both sine[5] and sawtooth wave outputs to about middle C. Patch a sine wave from the VCO out. Patch a sawtooth wave out.

[5] If your VCO has no sine wave output, use a triangle. If you have neither sine nor triangle, use a square wave.

Alternately open and close the two attenuators into the mixer. Spend some time comparing and contrasting the different characteristic sounds of a sine and a sawtooth wave.



In subjective terms, did you notice that the sine wave sounded relatively pure compared with the sawtooth? That the sawtooth sounded rich and full compared with the sine? The reason for this is that the sine wave *is* relatively pure,[6] and what it is pure of— what it lacks— is *harmonics,* or overtones.

The sine wave is *the* basic wave in acoustics. All other periodic waves, no matter how complex, can be created as the superimposition of sine waves of differing frequencies and amplitudes upon one another.[7] As an example, look back at wave 1 and wave 2 in the graph on page 18. Here the amplitude of these two waves is plotted against time. You can see by the numbers at the top and bottom of the page that the total time represented is 28 seconds. Wave 1 has a frequency of 1 cycle every 5 seconds, an amplitude of +2½ volts, and a *peak-to-peak amplitude* (the total amplitude between the maximum positive and maximum negative values) of 5 volts. Wave 2 has a frequency of 1 cycle every 14½ seconds, an amplitude of +2 volts, and a peak-to-peak amplitude of 4 volts. If these waves were audio waves generated by two different oscillators and mixed (Figure 2-8), they would have a different sound quality than either would separately.

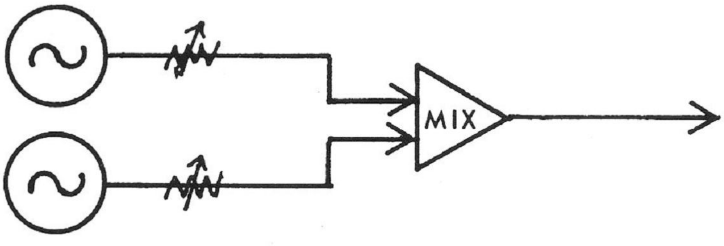


FIG. 2-8

In our example on the graph, the composite waveform created by the mixing (addition) of waves 1 and 2 is shown by wave 3. The way you can confirm this for yourself is as follows: Look what happens to the amplitude of each wave at second #0: both waves are at 0 amplitude, so on a third graph (the graph for wave 3) put a mark at 0 amplitude and 0 time. At second #1, wave 1 has an amplitude of +2¼ volts, and wave 2 has an amplitude of +¾ volt, a *total* of +3 volts. *When two or more waves are mixed, their total amplitude is the algebraic sum of the amplitudes of each wave at each instant.* Put a mark at 3 volts on your third graph for second #1. At second #2, wave 1 has dropped back to +1¾ volts amplitude, but wave 2 has risen to +1½ volts. Since 1¾ and 1½ equals 3¼ volts, the composite wave will be at 3¼ volts for second #2. Put a mark on the third graph at that point. Continue adding wave 1 and wave 2 for each succeeding second, and then connect all the marks. This will give you a picture of the wave that will be created by mixing wave 1 and wave 2. Since *a change in wave shape is generally equivalent to a perceived change in timbre,*[8] wave 3 will sound different from either 1 or 2.

[6] It is *relatively* pure because no synthesizer can generate a *completely* pure sine wave.

[7] In fact, the superimposition of sine waves, called *additive synthesis,* was the primary way in which electronic music composers created their compositions prior to the invention of voltage control.

[8] Excepting phase changes, any change in wave shape will imply the addition or subtraction of harmonics. This will almost always alter, however subtly, the way the wave sounds.

Just as you can add waves 1 and 2 to get wave 3, you could say that wave 3 can be defined as the addition of two sine waves, in this case 1 and 2. Similarly, *all periodic complex waves can be shown to be equivalent to the algebraic addition of sine waves of varying amplitude and frequency.* A wave that looks like the one in Figure 2-9 might be defined as the addition of 31 sine waves of differing amplitude and frequency but, with the right mathematical knowledge, we could state specifically the frequency and amplitude of all the necessary sine waves to create the wave in Figure 2-9. (The mathematical method used to do this is called Fourier analysis and is beyond the scope of this book.)[9]

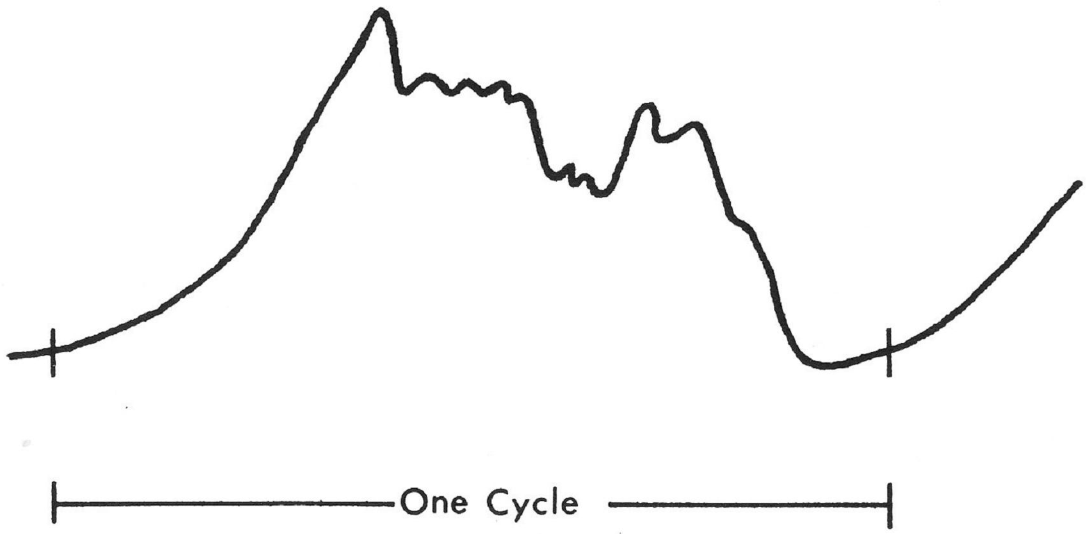


FIG. 2-9.

Like all other complex periodic waves, a *sawtooth wave* may be defined as equivalent to a series of sine waves of varying amplitude and frequency superimposed upon one another. For generating a sawtooth of, say, 440 Hz., the frequency and amplitude relationships of the harmonics to the fundamental are as follows: Wave 1 would be the fundamental, a sine of 440 Hz.; wave 2 would be a sine of double the frequency and half the amplitude of the fundamental. If we arbitrarily say that the amplitude of the fundamental is 10 volts, wave 2 would be a sine wave of 880 Hz. and an amplitude of 5 volts. Wave 1 is called the first harmonic, wave 2 the second harmonic, and so on. To the sum of waves 1 and 2 we add a third sine wave of three times the frequency and one-third the amplitude of the fundamental. To the sum of these three waves we add a sine wave of four times the frequency and one-fourth the amplitude of the fundamental. If we keep going until about the 16th harmonic, a sine wave of 16 times the frequency and one-sixteenth the amplitude of the fundamental (about 7040 Hz. and .6 volt), we will then have a composite wave form that looks approximately like Fig. 2-10. As you might guess, it sounds like a sawtooth.

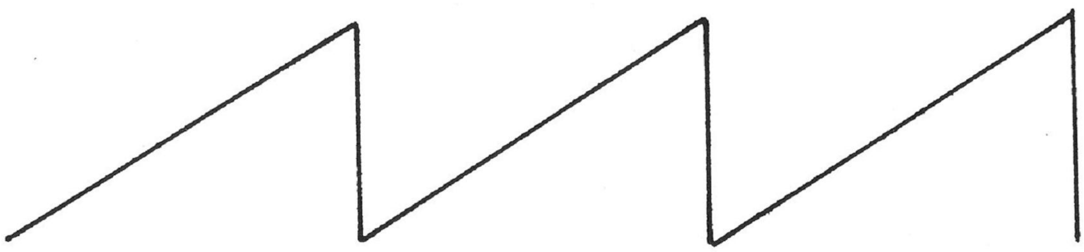


FIG. 2-10

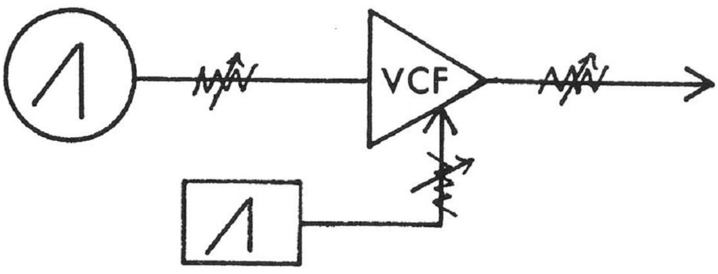
[9] See Wells (7) in the Bibliography.

A sawtooth wave has all the harmonics in the *harmonic series.*[10] These harmonics are not just mathematical abstractions but are the actual higher frequency components of the original wave. Any frequency that exactly doubles another frequency sounds exactly one octave higher; in fact, that is the definition of an octave. In our example the fundamental was 440 Hz.; the second harmonic, 880 Hz., is exactly one octave higher and, in a sawtooth, half the strength (amplitude). The third harmonic, at 1320 Hz., is an octave and a fifth above the fundamental, and the fourth harmonic, at 1760 Hz., is exactly two octaves above the fundamental.

EXPERIMENT #6: Harmonics of a sawtooth wave

Patch an AF sawtooth wave into an audio input to the filter. Patch the filter out and open the appropriate attenuators. Open the filter until you hear some substantial degree of sound (the cutoff frequency attenuator will probably be about 1 KHz.). Now open the resonance attenuator to the right until you hear a uniquely different sound. This is the sound of the filter beginning to oscillate,[11] something you'll learn more about in Chapter 3, section II. Once you hear that tone, close the resonance attenuator just until the tone disappears. You now hear the sound you first heard, and the filter is in a state of maximum stable resonance, the point just beyond which the filter will oscillate. Now put the F_C attenuator to about the same frequency as that of the VCO.

Patch a very low-frequency sawtooth wave into an attenuated control input to the filter. Open the attenuator over the control input to the filter. The sound you will hear will be that of the LF sawtooth opening the filter in such a way that you successively hear the different harmonics of the AF sawtooth from the VCO.



VCF : MAXIMUM STABLE Q

Note that "Q" is the symbol meaning "resonance." The effect of that highly resonant filter is to cause each harmonic of the input wave to be emphasized, one after the other, as the LF sawtooth wave opens the filter (see resonance, Chapter 3, section II-E).

Needed: negative DC voltage

To hear this even better keep everything as it is and patch some negative DC voltage to a control input to the LFO. This will have the effect of slowing down the controlling LF sawtooth wave; thus the filter will open more slowly, giving you more time to listen to the harmonics.

[10] The mathematical series $\frac{1}{2} + \frac{1}{3} + \frac{1}{4} + \ldots \frac{1}{n}$ is called the harmonic series. Since a sawtooth wave has frequency components of twice the fundamental, three times the fundamental, four times the fundamental . . . *n* times the fundamental, it contains all the harmonics in the harmonic series. As a practical matter one would rarely be concerned with frequency components higher than the sixteenth harmonic.

Harmonics are sometimes called *partials,* although there may be inharmonic partials if the higher frequency components are not exact integer multiples of the fundamental. Inharmonic partials are typically created by AM and FM techniques; see Chapter 6.

[11] Some VCFs (e.g., Oberheim, KORG) do not oscillate.

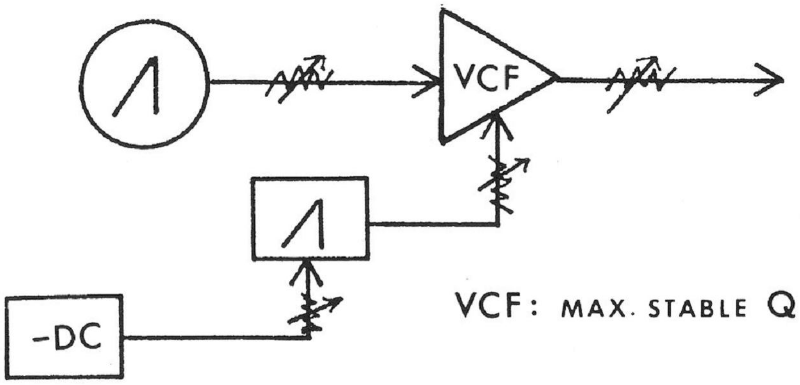


Table 2-1 summarizes the frequencies and amplitudes of sine waves necessary to create a sawtooth *or* a triangle wave, and it gives the position of the harmonic in terms of musical pitch as well. F represents the fundamental of the wave, 2F the second harmonic, 3F the third harmonic, and so on. A represents the amplitude of the wave; thus A/2 indicates that the amplitude is one-half as strong as the original amplitude, A/3 that the amplitude is one-third as strong, and so on. You need not memorize this but you should be familiar with it. Note that a triangle wave has no even harmonics and that the amplitudes of the harmonics it does have decrease *exponentially,* much more rapidly than with a sawtooth. Thus a triangle wave will sound more like a sine because it has less harmonic energy than a sawtooth. In fact, almost 90% of all the energy in a triangle wave is in the fundamental.

TABLE 2-1

	Sawtooth	Triangle	Harmonic
F	A	A	Fundamental
2F	A/2		F + 1 octave
3F	A/3	A/9	F + 1 octave + 5th
4F	A/4		F + 2 octaves
5F	A/5	A/25	F + 2 octaves + M 3rd
6F	A/6		F + 2 octaves + 5th
7F	A/7	A/49	F + 2 octaves + 5th
8F	A/8		F + 3 octaves
9F	A/9	A/81	F + 3 octaves + M 2d
10F	A/10		F + 3 octaves + M 3d

EXPERIMENT #7: Harmonics of a triangle wave

Repeat the last experiment, only this time use a triangle wave from a VCO instead of a sawtooth. You will hear those odd harmonics.

By definition the frequencies and amplitudes of the harmonics of a sawtooth and a triangle wave are in a specific mathematical relationship to the fundamental (shown in Table 2-1). Most synthesizer oscillators also generate a *pulse wave* whose shape can be varied by either manual or voltage control and (excluding phase changes) *if the shape varies, the harmonics change and thus the timbre will change.*

EXPERIMENT #8: Harmonics of a pulse wave

Patch an AF pulse wave out. Now manually vary the pulse width attenuator associated with the pulse wave of the VCO. Take your time and listen to the effect of harmonic changes within the wave.

If you patched a pulse wave output directly into an *oscilloscope,* you would see that the wave looks like Figure 2-11.

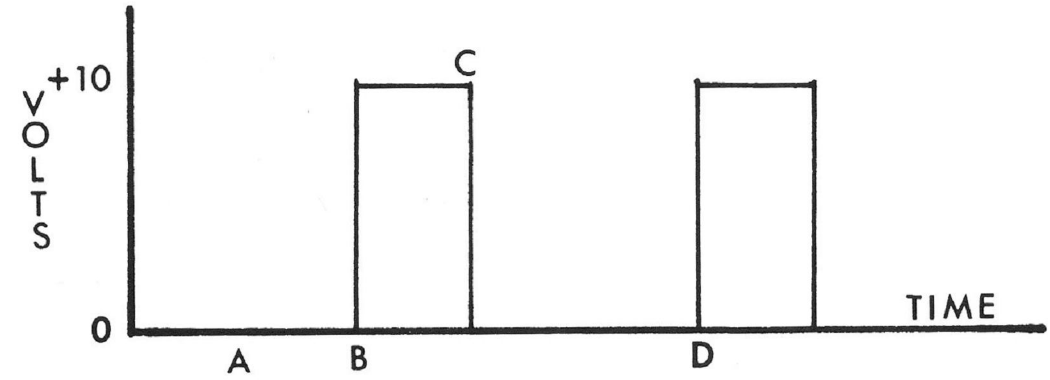


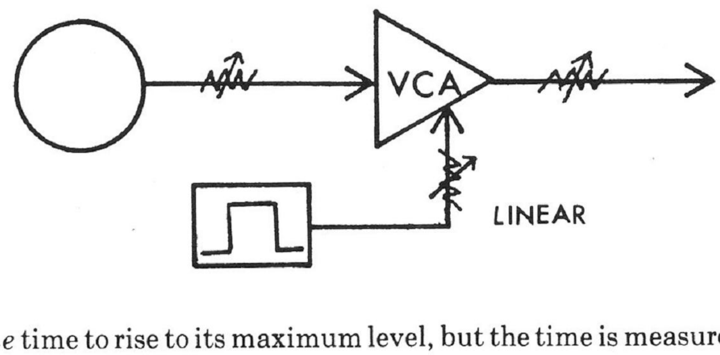
FIG. 2-11

Figure 2-11 shows that the voltage of a pulse wave is instantaneously either high or low. Thus at point A the pulse is low, at point B it *instantaneously*[12] rises to its maximum level, between points B and C it stays at that level, at C it instantaneously drops to its low level and remains there until the cycle begins again at point D. This pattern can be demonstrated by hearing the effect of a low-frequency pulse wave as it voltage-controls a VCA.

EXPERIMENT #9: LF pulse wave controlling a VCA

Patch any AF wave into an audio input to the VCA. Patch an LF pulse wave into the linear control input to the VCA. Open the attenuators over these inputs. Set the frequency of the LFO at about 1 Hz. and the pulse width attenuator to about 50%. Now patch the VCA out and open the appropriate attenuators.

What is happening is that the LF pulse wave is controlling the output of the VCA. When the pulse is low, no voltage gets to the VCA control input, so no sound comes out. When the pulse goes instantaneously high, maximum voltage gets to the VCA control input, so the VCA opens and we hear the audio signal being input. The LF pulse is functioning like an on-off switch, telling the VCA when to be open and when to be closed.



[12] Of course, it takes *some* time to rise to its maximum level, but the time is measured in *millionths* of a second, which is, for practical purposes, instantaneous.

Although the LF pulse itself always goes from zero to +10 volts, the amount of voltage controlling the VCA can be regulated by the attenuator over the jack into which the pulse wave is input. If the attenuator is all the way open, all the voltage gets through and the VCA opens up to its maximum capability, letting maximum audio signal through. If the attenuator is only halfway open (level 2), then only 5 of those 10 control volts get through, and the VCA opens up only half as much (Experiment #35).

It is important to understand that the LF pulse is still varying from 0 to +10 volts; it is just that the attenuator's being halfway open lets only half of that control voltage through.

The pulse width attenuator determines the percentage of each cycle that the pulse is high and low. The total of both high and low for one cycle must equal 100%. If the attenuator is set at 10%, then the pulse is high for 10% of its cycle and low for 90%. The proportion of time of a pulse spent high to low is called the *duty cycle.* If the pulse width is 25% it has a 25% duty cycle, which means it is high for one-fourth of the entire cycle. If the duty cycle of the pulse controlling the VCA in the previous example had been 25%, you would have heard sound from the VCA followed by silence which was three times as long as the sound (the ratio of 25:75 being the same as 1:3). Since a pulse wave is periodic, this would have recurred again and again. Try it and hear.

What happens harmonically when the pulse width (duty cycle) varies is complicated, but here is a general rule of thumb: If you convert the duty cycle to a fraction, then the denominator of that fraction, together with all integer multiples, will be missing in the wave.[13] An example: A given pulse wave has a duty cycle of 25%, which converts to one-fourth. Thus the 4th harmonic, together with each integer multiple (the 8th, 12th, 16th, 20th, etc.) will be missing from the wave. If the duty cycle were 20% (one-fifth), the 5th, 10th, 15th, 20th, etc., harmonics would be missing.

EXPERIMENT #10: Creation of a pulse wave with 25% duty cycle

Needed: negative DC voltage

Patch an AF pulse wave whose duty cycle is 10% into an audio input to the filter. Open the attenuator over the jack into which you have patched the pulse wave. Set the filter to a point of maximum stable resonance, just as you did in Experiment #6.

Patch a very low-frequency sawtooth wave to an attenuated control input to the filter. Open the attenuator.

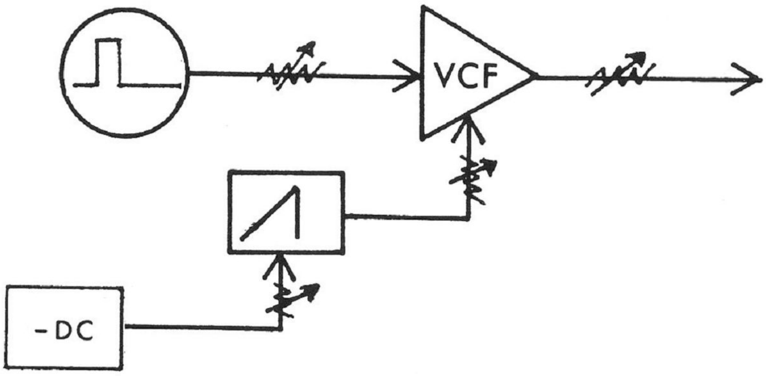
At this point the LF sawtooth opens the filter in such a way that you hear the harmonics of the pulse wave. Since the pulse width is 10% you will hear almost all harmonics (only the 10th, 20th, etc., will be missing).

Patch some negative DC voltage to control the LFO.

[13] This is really only true when the numerator of the fraction is 1. The equation for determining the amplitude of *any* harmonic in a pulse wave is

$$a_n = 1/n[\sin(180 \times n \times d)],$$

where a = amplitude, n = the harmonic number, and d = the duty cycle of the pulse wave.



The purpose of this experiment is to create a 25% pulse wave. Since 25% = ¼, a 25% pulse wave would have every 4th harmonic missing. Conversely, a pulse wave which has every 4th harmonic missing is a 25% pulse wave. Therefore to create a 25% pulse wave you need to isolate and then eliminate the 4th harmonic.

Listen as the LF sawtooth sweeps the harmonics of the pulse wave. Just before it gets to the 4th harmonic (which is a pitch 2 octaves above the fundamental), open the attenuator over the negative DC voltage into the LFO. This will slow the LF sawtooth and give you time to isolate and eliminate the harmonic.

Once you hear the 4th harmonic, move the pulse width attenuator of the VCO from its original position slowly one way and the other. Soon the 4th harmonic will get softer and then disappear. What has happened is that you have experimentally found the position at which the duty cycle of the pulse wave is 25%. How do you know? Because the 4th harmonic has been eliminated and you can verify for yourself that every 4th harmonic is gone. Thus by definition you have a 25% pulse wave.

Try this with a 20% pulse wave (eliminate the 5th harmonic) and other pulse waves as well.

There is a special case when the duty cycle is 50%. This converts to one-half, so the 2nd, 4th, 6th—in fact every even harmonic—is missing. This particular kind of pulse wave is called a *square wave,* and it's easy to see why (Figure 2-12). The pulse is high 50% of the time and low 50% of the time, so it looks like a square (or series of squares for many pulses). If you set the pulse width of any pulse wave at 50% you will have a square wave. Table 2-2 gives

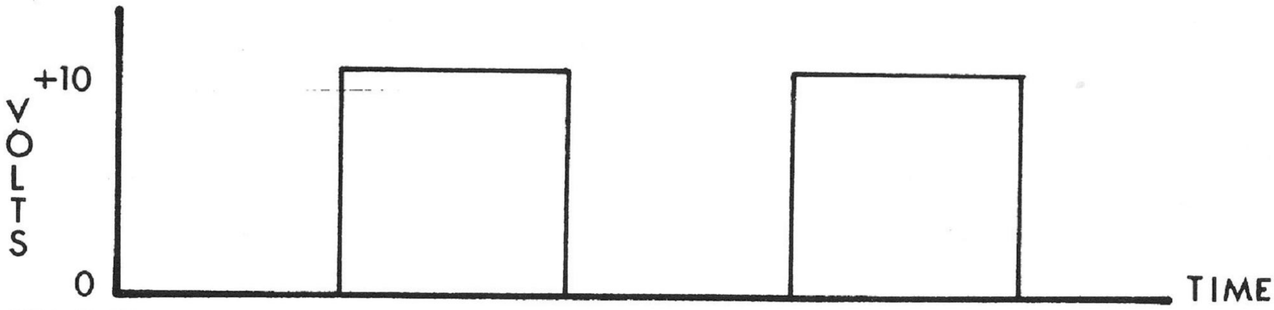


FIG. 2-12

relationships of harmonics of sine, triangle, sawtooth, and two varieties of pulse wave for you to compare.

What you should remember is that the fewer harmonics a wave has, the purer it will sound; the more it has, the brighter it will sound. A sine wave has no harmonics; a square

TABLE 2-2

	Sine	Triangle	Sawtooth	33% Pulse	Square	Example of Frequency (Hz)	Corresponding Pitch
F	A	A	A	A	A	440	A
2F			A/2	A/2		880	A (octave)
3F		A/9	A/3		A/3	1230	E
4F			A/4	A/4		1760	A (2 octaves)
5F		A/25	A/5	A/5	A/5	2200	C#
6F			A/6			2640	E
7F		A/49	A/7	A/7	A/7	3080	G
8F			A/8	A/8		3520	A (3 octaves)
9F		A/81	A/9		A/9	3960	B
10F			A/10	A/10		4400	C#
11F		A/121	A/11	A/11	A/11	4840	D 1/4 sharp
12F			A/12			5280	E
13F		A/169	A/13	A/13	A/13	5720	F 1/4 sharp
14F			A/14	A/14		6160	G
15F		A/225	A/15		A/15	6600	G#
16F			A/16	A/16		7040	A (4 octaves)

and triangle wave have only odd harmonics (but those of the triangle have less amplitude than those of the square wave and so the triangle sounds purer than the square); a pulse has varying harmonics, and a sawtooth has all the harmonics.

Having all the harmonics, a sawtooth gives a full, bright sound. If you wanted to synthesize strings or brass, you might begin by patching sawtooth and rich harmonic content pulse waves into the filter. Pulse waves with mid-range duty cycles (e.g., 30–50%) are the basic building blocks of double-reed instruments, while those with shorter duty cycles (10–20%) sound richer. A square wave, with its missing even harmonics, sounds somewhat hollow, like a clarinet. Most of a flute's sound can be approximated by a sine wave.

4. Phase

So far we have defined a periodic wave by its frequency, amplitude, and harmonics. The final way in which we define such a wave is by its phase. Phase generally has relevance when used to describe a relationship to another wave.

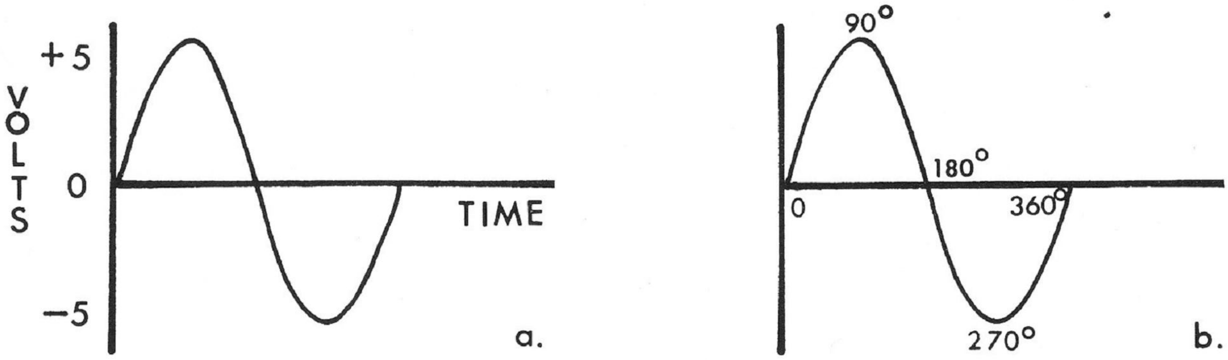


FIG. 2-13

Figure 2-13a shows the way the amplitude of a sine wave varies over time. Its phase can be shown as in Figure 2-13b. The point that is defined as the beginning of a wave's cycle is the point of 0° phase. A periodic wave is said to "travel through" 360° of phase each cycle. To the extent that a wave has traveled less than 360° it has traveled less than a full cycle. At its point of maximum positive amplitude the sine wave in Figure 2-13b has a phase of 90°; when it is at its maximum negative amplitude its phase is 270°. The point of 360° phase is the same as the point of 0° phase (that is, the end of one cycle is the beginning of the next cycle).

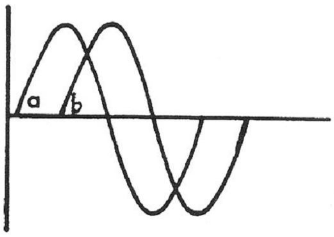


FIG. 2-14

Figure 2-14 shows two sine waves, one whose cycle commences at some point in time after the other's. Wave B commences when wave A has a phase of 90°; therefore wave B is said to be 90° out of phase with wave A.

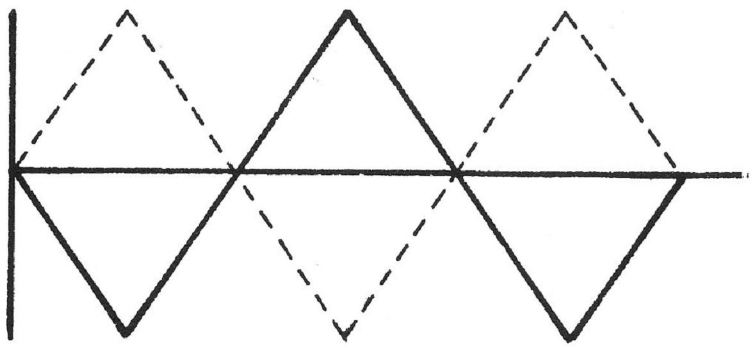
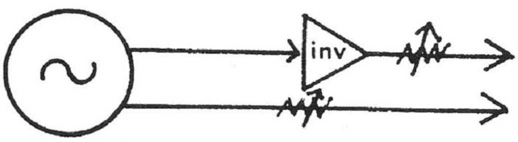


FIG. 2-15

In Figure 2-15 two triangle waves are 180° out of phase with each other; that is, one wave can be said to begin 180° after the other wave. To the extent that waves are *completely* (i.e., 180°) out of phase with one another, those harmonics that they have in common will completely cancel each other.

EXPERIMENT #11A: Mixing a wave 180° out of phase with itself

Needed: inverter, multiple



Patch an AF sine wave into a multiple. Patch one multiple output directly out. Patch another multiple output into the inverter. One of the effects an inverter has is to shift a sine wave through 180° (Chapter 5, section II-C). Patch the inverter out. Listen first to one wave alone, then to the other wave alone. Note that they sound the same. Now open first one attenuator and, leaving that open, slowly open the other attenuator. By so doing you are mixing two waves identical in every respect except that one is totally out of phase with the other. At some point as you open the second attenuator the sound will cease altogether, as the waves completely cancel each other's voltage.

Did you notice that the inverted wave *sounded* the same as the original? The phase of any one wave is generally aurally unimportant; just by listening to one AF wave you can discern nothing of its phase. Rather, phase is important to synthesists for three reasons:

1. When two waves that have frequency components out of phase are mixed the components will "beat" against one another, causing a specific "whirring" sound which may or may not be musically desirable (Experiment #12).
2. Since difference in phase really means a difference in the *time* that two waves pass a given point, phase difference is a primary cue in determining *location*. If you perceive a sound as coming from the right it is the difference in time (i.e., phase) that the sound takes to reach both your ears that tells you where the sound is located.
3. Waves mixed out of phase will produce different effects as control voltages than if they were in phase. Figure 2-16a shows a fundamental and third harmonic in phase; note the shape of the new wave created by mixing these two waves (Figure 2-16b). Figure 2-16c is the same two waves as in 2-16a, but the third harmonic is 180° out of phase with the fundamental. The resultant wave (Figure 2-16d) is significantly different from that in Figure 2-16b. Aurally they would sound the same, but as control voltages (i.e., at low frequencies) the two waves would have significantly different effects because of their different shapes.

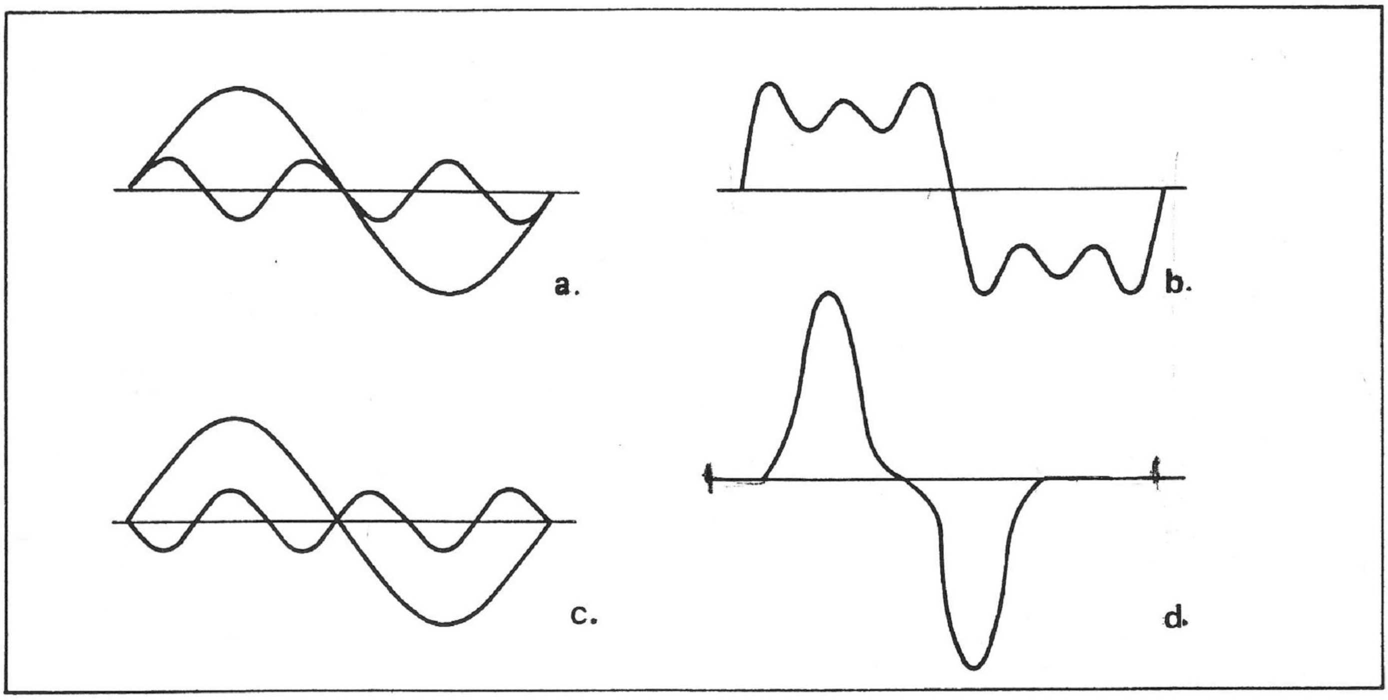


FIG. 2-16

The only way to shift the phase of a wave on a standard synthesizer (without a specific module called a "phase shifter," a special kind of filter) is to use an inverter with a sine, triangle, or A-C square wave; because they are asymmetrical one cannot phase shift a pulse or sawtooth wave with an inverter (this would give reversed polarity but no time difference; graph it and see for yourself).

The relevance of phase is generally in the relationship of two or more waves. Experiment 11A showed that if two mixed waves have important phase differences, you may get a very different aural effect than you would otherwise expect. The waves generated by any given VCO have specific phase relationships. The manual which came with your synthesizer should show you the phase relationships of waves generated by the VCOs. For example, Figure 2-17 shows phase relationships of the waves generated by

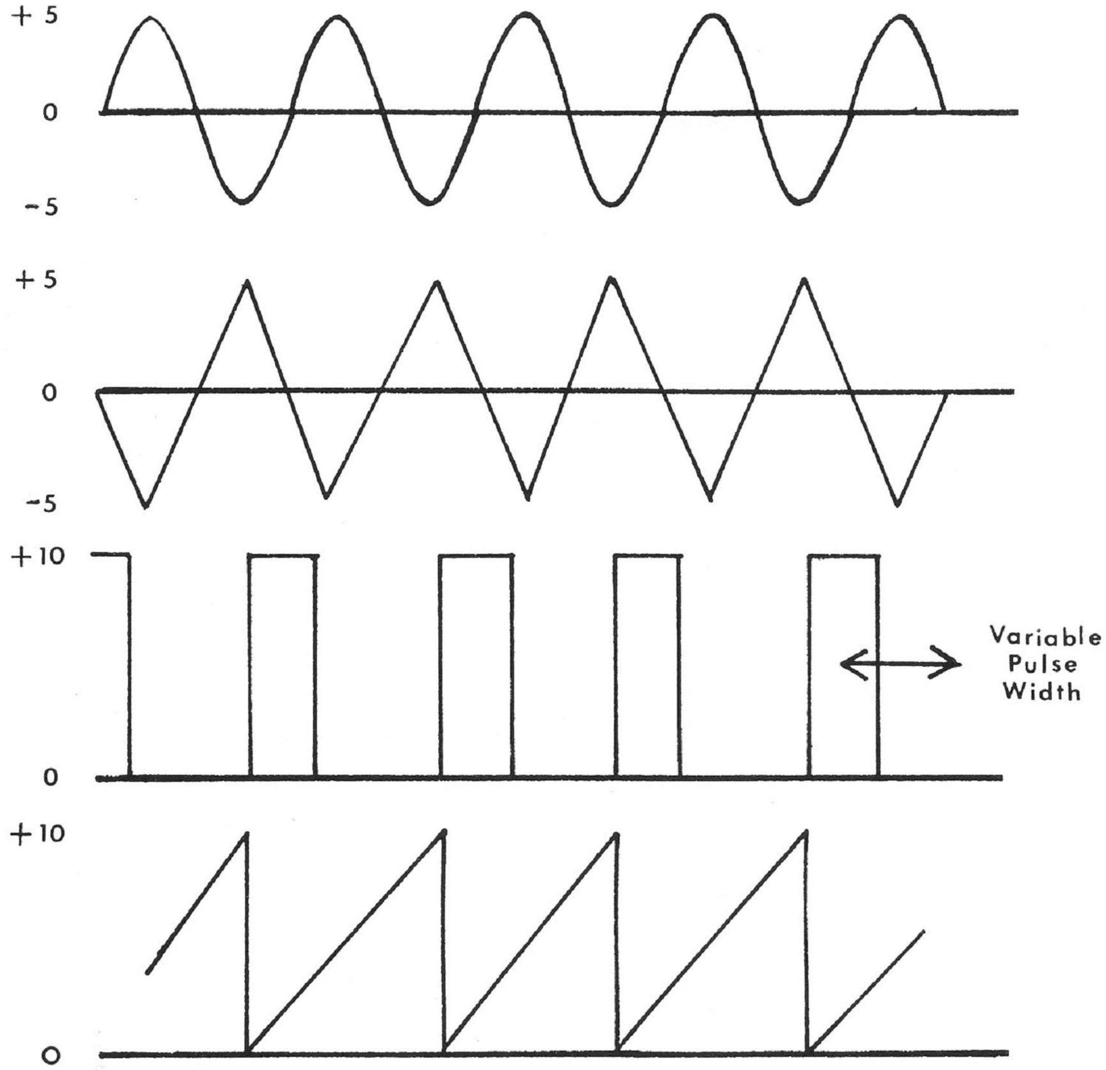
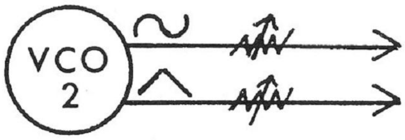


FIG. 2-17

VCO-2 of the ARP 2600 (as shown by that synthesizer's manual). Note that the sine and triangle waves are 180° out of phase with each other, as are the pulse and sawtooth waves (the pulse goes high the instant the sawtooth drops to 0 volts). Therefore, mixing VCO-2's sine and triangle waves would give you an output of only the harmonics of the triangle wave. This is because all their harmonics in common will completely cancel; the only harmonic in common is the fundamental, and the only frequency components left are the harmonics of the triangle wave.

EXPERIMENT #11B: Mixing different waves 180° out of phase



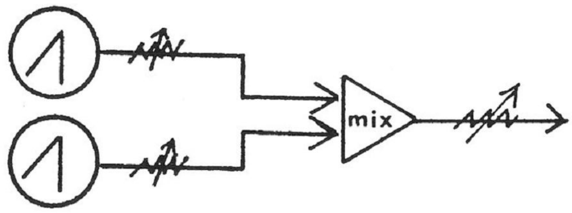
If the manual that came with your synthesizer indicates that two particular waves from one VCO are out of phase, patch each one out. Open the attenuator over the first wave and then slowly raise the attenuator over the other. You will hear the timbre change as the harmonics that the waves have in common cancel each other and drop out.

The above diagram shows the patch for the ARP 2600, since the sine and triangle waves on VCO-2 of that synthesizer are 180° out of phase with each other.

Another example of a different aural effect than you might generally expect is the "phasing" sound heard when two oscillators are tuned to unison. This is caused by the different harmonics of the generated waves "beating," or going in and out of phase with one another.

EXPERIMENT #12: Harmonics going in and out of phase

Patch an AF sawtooth wave from a VCO out. Now patch another AF sawtooth wave from another VCO out and adjust the tuning of the second VCO so that the two waves are in unison (if you have trouble doing this, see Chapter 3, section A-4, Tuning and Beat Frequencies). The two VCOs are generating sawtooth waves of almost identical frequency and amplitude. However, VCO's cannot be tuned absolutely *identically unless they are locked in synchronization (Chapter 3, section I-E), and so the waves are not in phase with one another.*[14] *If you listen carefully to the two sawtooth waves you will hear a "whirring" sound something like the sound created by a phase shifter.*



These are the harmonics of the two waves beating against each other as they go in and out of phase with one another.

Most synthesizers have oscillators that generate waves with a p-p (peak-to-peak) amplitude of 10 volts. Typically the sine and triangle waves fluctuate from +5 to −5 volts, while the pulse and positive-going sawtooth go from 0 to +10 volts: the pulse goes instantaneously high and low while the positive-going sawtooth goes gradually high and then instantaneously low. A negative-going sawtooth wave (sometimes called a ramp wave) begins its cycle at +10 volts and gradually goes to 0 volts. *Thus, in their low frequencies a sine or triangle wave can drive a voltage-controlled device either up or down, but a positive-going sawtooth or pulse wave can only drive a voltage-controlled device up.*[15]

You have learned that each wave has certain characteristic harmonics (or lack of harmonics). You have also learned that each wave has a particular pattern of voltage fluctuation. Harmonics are important to remember when you are dealing with waves in their audio frequencies; voltage fluctuation is important to remember when you are dealing with waves in their low frequencies. Thus, although it is true that *for each cycle* of a 1 KHz. pulse wave the wave is very rapidly varying from 0 to some specific voltage (typically 0 to +10 volts), you will not often have occasion to be concerned about the nature of the voltage fluctuation of a pulse wave in its audio frequency, because that wave is an audio wave—it is not *controlling* anything. Similarly, it is true that a 3 Hz. sawtooth wave has harmonics of 6 Hz., 9 Hz., 12 Hz., etc., but you won't often have to be concerned about these harmonics when dealing with waves in their low frequencies, because that wave is a control

[14] The only way two waves can be in phase is if they are synchronized in some way. For example, a square wave from VCO-2 of the ARP 2600 is in phase with a triangle wave (Figure 2-17). Many other synthesizer VCOs (e.g., Moog Prodigy, Oberheim SEM, Sequential Circuits Prophet 5) have a "synch" function that causes one VCO to follow and be "slaved" to the phase of another. See Chapter 3, section I-E.

[15] There are certain processing methods, known as offsetting and inverting, that can change this general rule. These are discussed in Chapter 5.

wave, and you are not listening to it but rather to its effect on another wave. *If the wave is at an audio frequency, think harmonics; if it is low frequency and being used as a controller, think shape of voltage fluctuation.*

B. Aperiodic Waves

In general, aperiodicity implies something that happens just once, or perhaps several times randomly (like a baby banging a cup on a table), or something that happens forever unpredictably. If the aperiodicity is of the first variety, it is called an *event*. There can be many events with no periodicity to them—random events—but as soon as they recur *with regularity,* they become periodic, subject to the parameters discussed earlier in this chapter. Shoes hitting the floor are aperiodic events; heartbeats are periodic events.

One of the primary aperiodic signals dealt with in synthesis is called *noise* and is created, appropriately enough, by a noise generator. Noise is usually an audio signal although, particularly when filtered, it can be a source of random control voltage. The noise generator is treated in depth in Chapter 5. Let us say at this point that white noise occurs when at any given instant there is an equal probability of any given frequency being present with respect to the probability of any other frequency being present. Practically, it sounds like all frequencies are always present. To hear white noise, open the noise generator attenuator(s), patch the noise generator out, and open the appropriate attenuator. Whenever an experiment calls for you to use the noise generator, you should use white noise unless otherwise stated.

III. BASIC VOLTAGE CONTROL[16]

A. Voltage Control of a VCO

You have probably heard that human beings have an approximate hearing range of 20 Hz. to 20 KHz. If you recall that an octave is defined as a frequency exactly double that of another frequency, you will understand how it is that we have a hearing range of about ten octaves.

Octave	Frequency (in Hz.)
0	20
1	40
2	80
3	160
4	320
5	640
6	1280
7	2560
8	5120
9	10240
10	20480

For now you should know that, with regard to exponential control voltage inputs to the

[16] This section will be more understandable when you have completed Chapter 3.

VCOs and the VCF, most synthesizers track frequencies at the rate of *one octave per control* volt:

EXAMPLE:

If a VCO has an initial oscillator frequency of 500 Hz. and 1 control volt is input to the VCO, the frequency will rise one octave. If 3½ control volts are input the frequency will rise 3½ octaves. Remember that this does not *mean the frequency will be 3½ × 500, or 1750, Hz. Rather it means that since the oscillator's initial frequency is 500 Hz., the first octave is 1000 Hz., the second is 2000 Hz., the third is 4000 Hz. and, since the fourth is 8000 Hz., the frequency will be about 6000 Hz. The relationship between frequency and octaves is* exponential.

EXAMPLE:

If a VCO has an initial frequency of 1000 Hz. and −2 control volts are input to the VCO, the frequency of the VCO will be lowered two octaves, to 250 Hz.

How control voltages are input to VCOs will be discussed in detail in Chapter 3, section I. There are several possible voltage controllers. One way to input control voltages to an oscillator (or any other voltage-controlled module) is through a keyboard. A typical keyboard might have a span of four octaves,[17] which means it has a range of 4 control volts. Think of a synthesizer keyboard as a source of control voltage rather than as a piano-type keyboard. The lowest key is assigned a value of 0 volts, the highest, a value of +4 volts.[18] Middle C is assigned a value of +2 volts. If there is no other control voltage input, the lowest key will cause a VCO to sound the same pitch as the VCO's initial frequency. Thus, depressing the highest note on the keyboard will raise the frequency of a VCO controlled by the keyboard by four octaves, because you have input +4 control volts to the oscillator.

Since there are twelve *semitones* in an octave, each semitone is 1/12 volt more or less than an adjacent semitone. Playing the G just above the lowest C will add a control voltage of +7/12 volt to the controlled oscillator (G is seven semitones higher than C).

B. Voltage Control of a VCF

A VCF allows or inhibits the passage of harmonics, which are higher frequency components of a particular input audio signal. It too is generally controlled at the rate of 1 volt per octave.

EXAMPLE:

A sawtooth wave whose fundamental is 500 Hz. is input into an audio input to the filter. That sawtooth has harmonics of 1000, 1500, 2000, 2500 . . . Hz. Assume that the initial cutoff frequency is also 500 Hz. Since the initial F_C setting is the point at which harmonic attenuation begins,[19] then attenuation (diminution) of harmonics

[17] Not including a transpose switch or pitch bend mechanism, covered in greater detail in Chapter 5, section III-B.

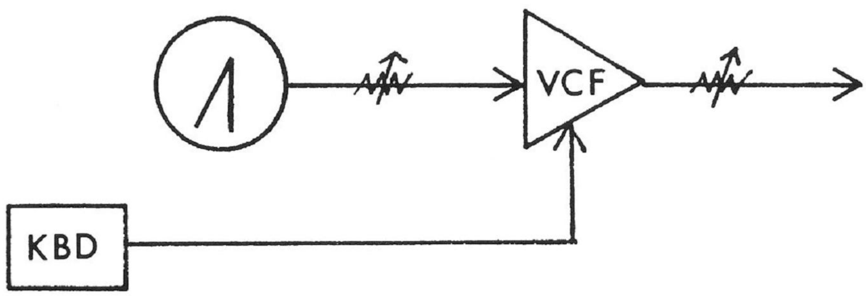
[18] On some synthesizer keyboards the middle key is assigned a value of 0 volts. The lowest key then has a negative voltage value.

[19] See Chapter 3, section II.

will begin immediately. If +1 control volt is put into a control input to the filter, the F_c will be raised to 1 KHz. Attenuation of harmonics would then begin with the second harmonic (at 1 KHz.) rather than the fundamental.

If −1 control volt had been input to the VCF, the F_c would be lowered to 250 Hz. Since the audio signal was 500 Hz., even the fundamental would be attenuated. You wouldn't hear much.

EXPERIMENT #13: Voltage control of a low-pass VCF at the rate of 1 volt/octave



Set up the above patch by inputing an AF sawtooth wave into an audio input to the filter. The coarse frequency attenuator associated with the filter should be set to about 1 KHz. The block diagram indicates keyboard control of the VCF but not of the VCO; therefore a dummy plug *must be inserted into the keyboard control voltage input of the VCO you are using (or the appropriate switch thrown if there is a pre-patch which would otherwise automatically provide for keyboard control of the VCO).*

Play the lowest C on the keyboard. You hear the sawtooth with harmonics severely attenuated. Now play the next highest C. The filter allows more harmonics to pass. Playing continually higher notes will open the filter more and more, allowing more and more of the sawtooth's harmonics to pass. By using a "transpose" switch to the left of the keyboard (if your synthesizer has one), you can open or close the filter even more.

C. Voltage Control of a VCA

The range of loudness that humans hear is astonishing. The proportion of the loudest sound we can hear before reaching the threshold of pain to the least sound we can hear is more than a trillion to one. Changes in loudness are measured in a quantity which increases or decreases logarithmically. That quantity is called a *decibel, db* for short.

Decibels are important to understand because we just do not grasp immense numbers very well. Saying that one sound is 90 db above another is more comprehensible than saying it is a billion times more intense, although that is what it means.

The relationship of decibels to loudness is shown in Table 2-3.

To say that one signal is 30 db greater than another is to say that it is 1,000 times as loud as the first. An example of that difference would be a comparison of ordinary conversation with an ordinary whisper (the difference between 40 db and 10 db). We normally converse at a volume 1,000 times greater than that at which we whisper!

There are two methods by which voltage can typically control the opening and closing of a VCA. The first is linear: The VCA opens in direct proportion to the control voltage.

TABLE 2-3

db	× Amplification	Typical Example
0	10^0 or 1	threshold of hearing
10	10^1 or 10	normal whisper at 10 feet
20	10^2 or 100	
30	10^3 or 1,000	residential street traffic
40	10^4 or 10,000	normal conversation at 10 feet
50	10^5 or 100,000	
60	10^6 or 1,000,000	loud orchestra string section
70	10^7 or 10,000,000	
80	10^8 or 100,000,000	
90	10^9 or 1,000,000,000	noisy traffic
100	10^{10} or 10,000,000,000	
110	10^{11} or 100,000,000,000	
120	10^{12} or 1,000,000,000,000	threshold of pain

The second is exponential: Control voltage generally increases the rate of the VCA opening at 10 db per control volt:

EXAMPLE:

Recall that an LF square wave instantaneously rises to its maximum level, stays there, and instantaneously falls to its minimum level, stays there for the same amount of time, and then rises again. If the controls are set such that an LF square wave patched into the exponential control input of a VCA has a minimum control voltage of 0 volts and a maximum of +5 volts, then it will instantaneously rise by 5 control volts, raising the VCA output by 10 db per control volt, or by 50 db. When the controlling square wave is high, the VCA output will be 100,000 times the output when the square wave is low. Although the numbers are large, this is not at all uncommon.

Voltage control thus typically permits you to modulate frequency, timbre, and amplitude in ways far more varied and precise than you could manually. You will experiment with voltage control in much greater depth throughout this book and particularly in Chapter 3.[20] For now remember these important points:

1. Voltage-controlling an oscillator will affect frequency; voltage-controlling a filter will affect timbre (tone color); voltage-controlling a VCA will affect amplitude (generally volume).
2. As control voltages, AC sine and triangle waves fluctuate from positive to negative voltage; positive-going pulse and square waves go instantaneously high and low; positive going sawtooth waves rise gradually and then instantly fall to their starting point; negative going sawtooth waves rise instantly and then gradually fall to their starting point.
3. Voltage control of a VCO or a VCF will generally be at the rate of 1 volt/octave; of the VCA either in direct proportion (linear input) or at the rate of 10 db/volt (exponential input).

[20] You have already heard some examples of voltage control when you used a sine wave as control voltage to modulate a VCO (Experiment #3), a VCF (Chapter 1, paragraph 22) and a VCA (Experiment #4).

chapter three

The Basic Synthesizer Modules

I. THE VOLTAGE-CONTROLLED OSCILLATOR

If it is true that each of the modules discussed in this chapter is equally important to a synthesizer—indeed if any one of them were missing you wouldn't have a real synthesizer—it is also true that the VCO is first among equals. It is the heart of the synthesizer, analogous to the vibrating string of an acoustic instrument. You need not filter a particular sound, nor gate it with an envelope generator, nor use a VCA in a patch at all; but (excluding a noise source) without an oscillator there will not generally be any sound in the first place.

There are two aspects of a VCO (sometimes called a "function generator") with which you are familiar but which will be mentioned here because they are so basic. The first is that an oscillator has the ability to generate different waveshapes and make them available at specific outputs. The second is that within almost all VCOs is an exponential converter which allows tracking at the rate of one volt per octave: each time you input +1 volt the frequency of the desired waveshape will double.[1]

A. The Control Voltage (CV) Mixer

As you know, whenever control voltage is input into a VCO whose control rate is exponential, the VCO's frequency will change at a rate of 1 volt per octave. If +1/6 volt is

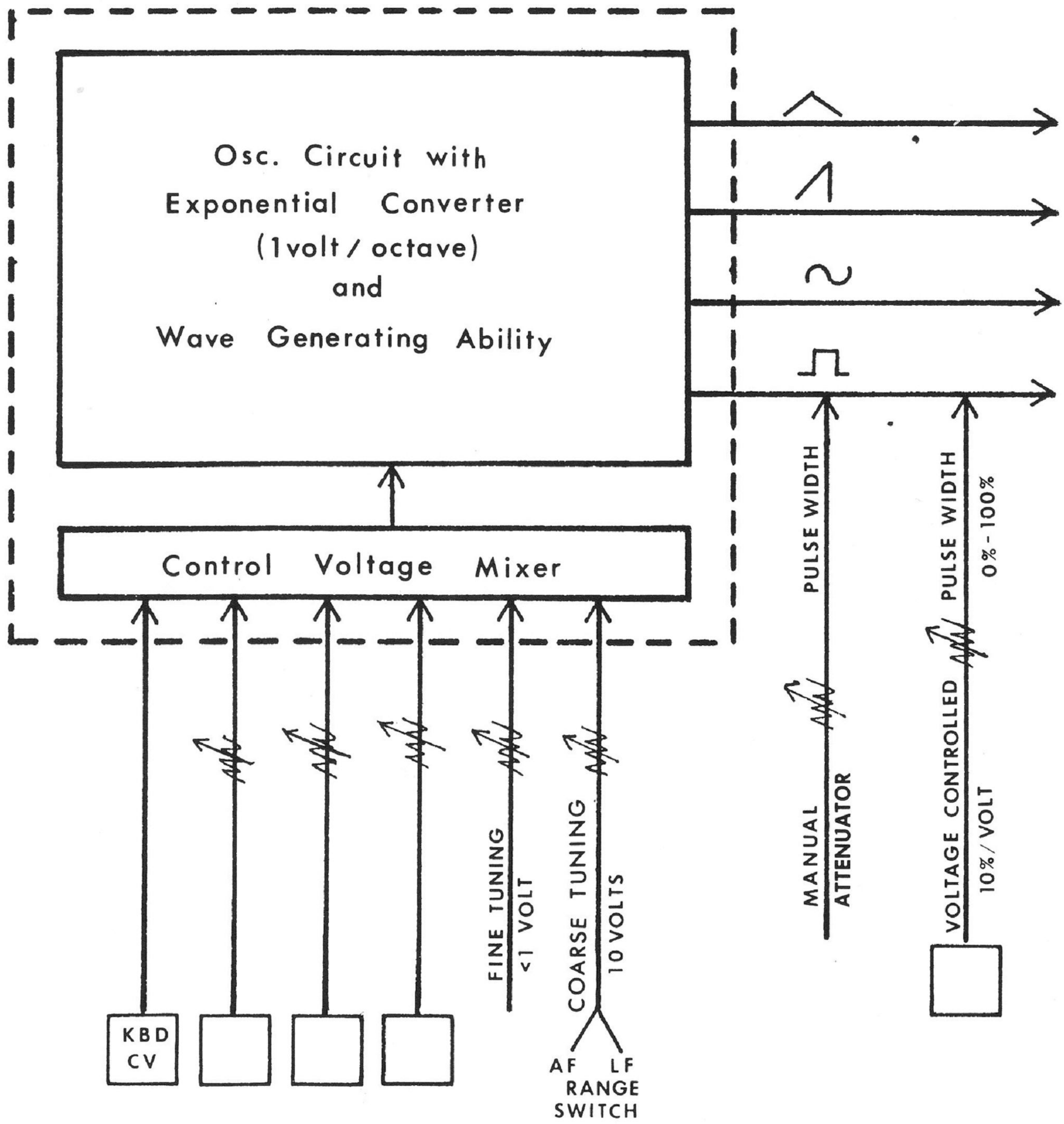
[1] Some modular synthesizer manufacturers (e.g., Serge-Modular, Moog, Aries, E-mu) offer VCOs that have linear as well as exponential control inputs. Voltage input into a linear control input will cause a VCO's frequency to vary in direct proportion (rather than exponentially) to the input control voltage. This is important in creation of harmonic timbres using FM techniques. See Chapter 6, footnote 2.

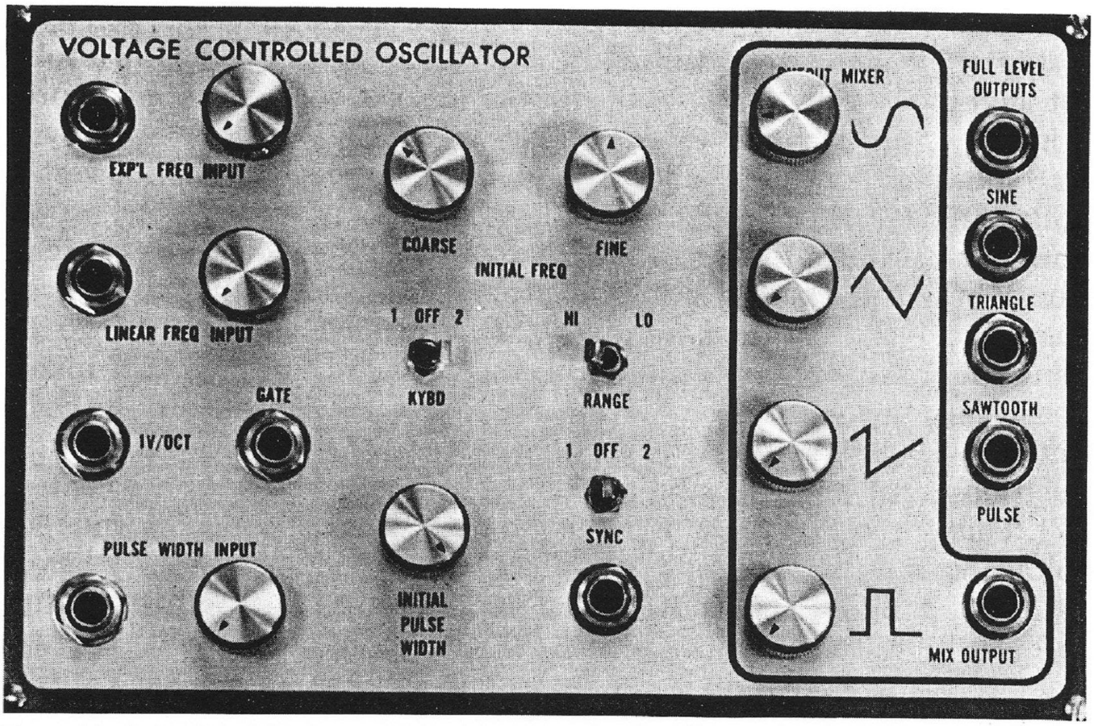
input the frequency will rise by an amount equal to 1/6 volt at a 1 volt/octave rate; at that rate 1/6 volt equals one whole tone. (There are six whole tones in an octave.) Conversely, if the pitch of a VCO rises one whole tone you know that +1/6 control volt as been input to it.

The control voltage may come from a keyboard, but that is only one of several sources of control voltage. Envelope generators, LFOs, sample and hold, indeed any voltage may serve as a source of control voltage to affect the frequency of a VCO. The coarse frequency attenuator associated with a VCO is also a source of control voltage(it must be, since moving it affects the frequency of the VCO). Typically such an attenuator has a range of 0 to + 10 volts. That translates into a range of 10 octaves and, if the VCO is in its AF range, the entire audible frequency range can be heard if you slowly sweep that attenuator from one side to the other. Try it.

A "fine tuning" attenuator typically has a range of slightly less than one volt (i.e., one octave). It's a good idea to put the fine tuning attenuator in its middle position *before* do-

FIG. 3-1



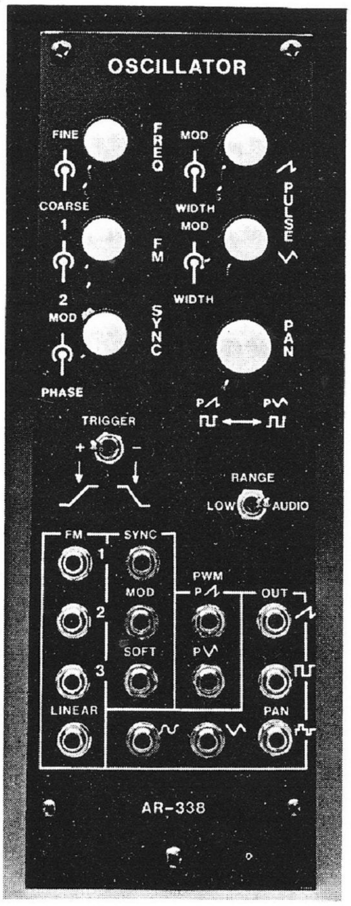


E-mu Module 2200 - Voltage-Controlled Oscillator

ing coarse tuning. Then you can either raise or lower the voltage when you have approximated the frequency you want with the coarse tuning attenuator. If you do the coarse tuning first and the fine tuning attenuator is all the way to the left you will only be able to add voltage (i.e., increase frequency) and, for tuning purposes, you may want to decrease voltage as well.

The block diagram of a typical VCO (Figure 3-1) shows that all control voltages are summed in a control voltage (CV) mixer before reaching the oscillator itself. (Remember that mixing voltages—including control voltages—means algebraically adding them.) The sum of every instantaneous control voltage in the CV mixer will determine the frequency which the VCO outputs at that instant.

Here is an example to see how frequency might be determined: Say you had set the coarse frequency attenuator at about 500 Hz. Since the lowest frequency available when a range switch[2] is in its "audio frequency" position typically is 10 Hz., 500 Hz. is the equivalent of putting about +5½ volts into the control voltage mixer.[3] Let's further say that you want the initial frequency of the oscillator to be high C, 523 Hz. If you had a tuning reference you would carefully adjust the fine tuning attenuator until the pitch you heard from the oscillator exactly matched the one you heard from the tuning reference.[4] If the voltage you added from the fine tuning attenuator to get 523 Hz. was 1/12 volt, the total voltage in the control voltage



Aries AR-338 VCO*

[2] VCOs of hard-wired synthesizers generally have no range switch; they generate only audio frequencies of about 10 Hz. to 10 KHz.

[3] If 10 Hz. is defined (for the purpose of conceptualization) as 0 volts, then 20 Hz. is 1 volt, 40 Hz. is 2 volts, 80 Hz. is 3 volts, 160 Hz. is 4 volts, 320 Hz. is 5 volts and 640 Hz. is 6 volts. Thus 500 Hz. is roughly 5½ volts.

[4] See discussion of beat frequencies in section D.

*All Aries modules are distributed exclusively by Rivera Music Services. See appendix b, page 191, for their address.

mixer would be 5-7/12 volts. If you then played a note on the keyboard, you would either add to or subtract from the total voltage in the CV mixer, anywhere from 0 volt (playing the lowest key) to +4 volts (playing the highest key if your keyboard is four octaves), resulting in a pitch change of up to four octaves higher than high C. (Some synthesizers have their 0 volt setting in the middle of the keyboard rather than at the lowest note. The control voltage range of such a keyboard, if it were four octaves, would be −2 volts to +2 volts.) The CV mixer reacts *instantaneously* to all algebraic additions (which includes subtractions)[5] of voltage, raising or lowering the VCO output at the rate of 1 volt per octave. Thus to the fixed control voltage of 5-7/12 volts you can rapidly change the control voltage (and thus the pitch you hear) by playing different keys. The CV mixer will instantaneously feed that information to the VCO, and the VCO will output the frequency you expect. (See Experiment #A-1 in Appendix A).

The leftmost control voltage indicated in Figure 3-1 is the keyboard control voltage (KBD-CV). This is frequently an unattenuated hard-wired or pre-patch and refers to the control voltage coming from the keyboard. This *unattenuated input* allows the voltage appearing there to go fully into the control voltage mixer, with no opportunity for attenuation.[6]

B. Hard-Wired Patches and Pre-Patches to CV Inputs

Many hard-wired synthesizers have switches that allow one of two or more control voltages to control a module. For example, the ARP Odyssey has a switch underneath its first VCO which, when up, allows the sample and hold voltage output to control the VCO; if it is down, the ADSR output controls the VCO.[7] Another example: The LFO on the Multimoog produces both a square and a triangle wave; the user flips a switch to determine which will be in use.

"Quasi-modular" synthesizers (see Chapter 7, section I-C) have hard-wired prepatches that may be defeated by insertion of a patchcord. For example, ARP has prepatched various outputs from other modules into the control voltage inputs of various ARP 2600 modules. The square wave output from VCO-1 is pre-patched into a CV input of VCO-2. If you do not patch anything else into that particular CV input and you raise the attenuator, the square wave voltage from VCO-1 will be mixed into VCO-2's CV mixer. However, you should not think of those jacks as outputs from various other modules; rather *they are all control voltage inputs to VCOs. One* of the uses of each input is the prepatched configuration, but it is only one of many possibilities. Do not limit yourself to the pre-patches; they make live performance very convenient, but a patchable synthesizer is much more flexible.

Switching a VCO into its LF range[8] sometimes negates a keyboard control pre-patch. However, if you want the speed of the LFO to vary as you play the keyboard, you can reinstate voltage control of the LFO by the keyboard by patching the keyboard control voltage output to a control voltage input to the VCO. If the input into which you patch the KBD CV is unattenuated (Figure 3-2a) the frequency of the LFO will vary at a rate of 1 volt/oc-

[5] E.g., +5 + (−2) = +3.

[6] There is a way to attenuate the keyboard control voltage so that microtones are created. Follow the same procedure as that shown in Experiment #32, but use a VCO instead of a VCF as your primary module.

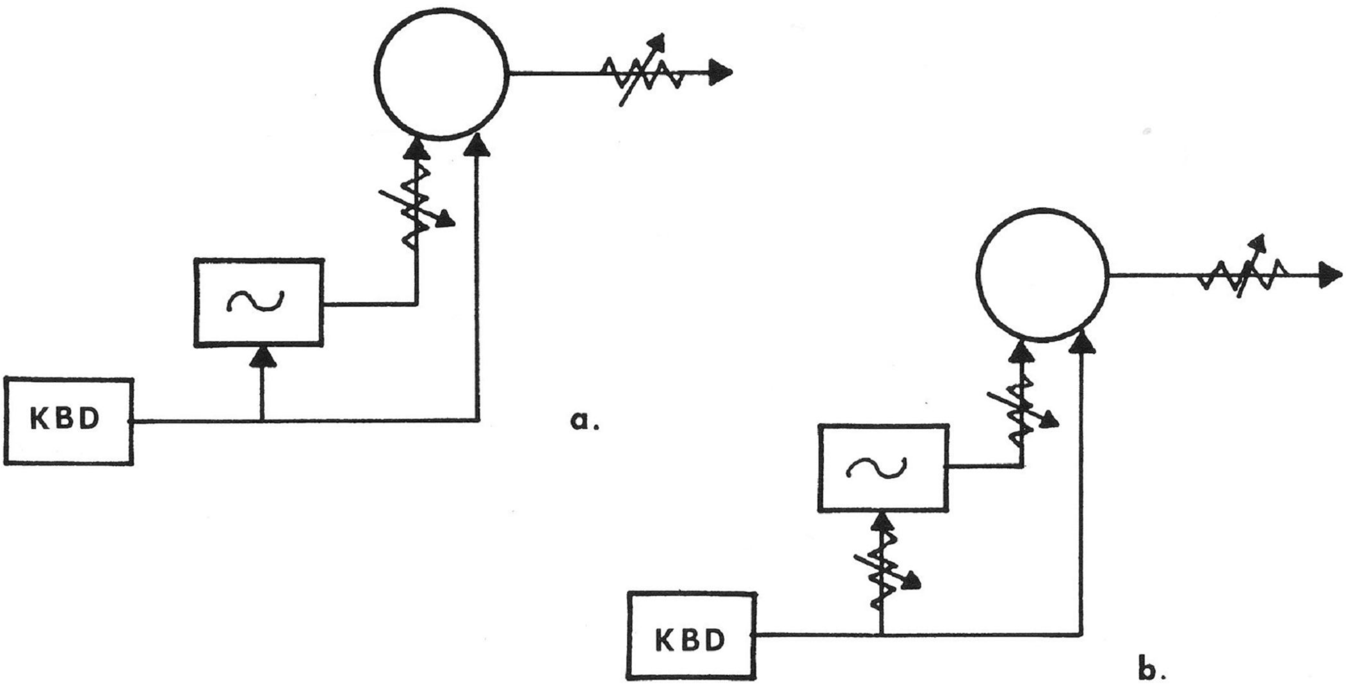
[7] Of course, the input attenuator must be raised before either the S & H or the ADSR can control the VCO.

[8] Some synthesizer VCOs (e.g., Serge Modular) provide for continual frequency changes of more than 20 octaves, without use of a range switch.

Polyfusion Model 2004 VCO

tave. For example, if the speed of vibrato were 4 Hz. and the keyboardist then played the octave higher, the vibrato speed would double to 8 Hz. If the input has an associated attenuator (Figure 3-2b), the speed of the LFO can vary at a rate of anything up to 1 volt/octave, depending on the attenuator setting.

FIG. 3-2



C. Voltage-Controlled Pulse Width

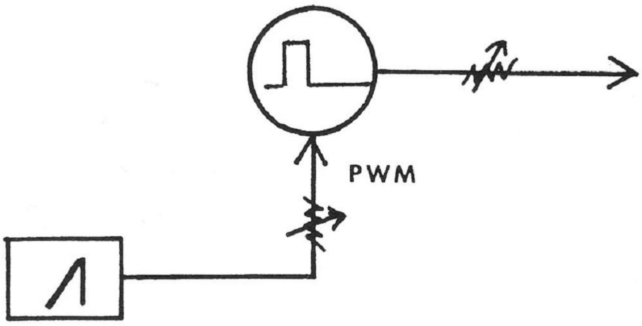
Most VCOs have a pulse wave output and an associated attenuator that allows the user to vary the duty cycle of the pulse, causing harmonic (timbral) changes in the AF pulse wave.[9] Although you experimented with this manually in Chapter 2 (Experiment #8), some VCOs allow for voltage-controllable as well as manually controllable pulse width.

Voltage-controllable pulse width generally varies at a rate of 10% per volt. If the pulse width is originally at 40% and you input +1 control volt to the pulse width control input, the duty cycle will be raised to 50%. At that point if you input −3 control volts the duty cycle will be lowered to 20%, with corresponding harmonic changes.

One way you can input control voltage is with low-frequency waveforms.

EXPERIMENT #14A: Voltage control of pulse width by an LF wave

Patch an AF pulse wave out. Put the pulse width attenuator at 10%. Patch a very slow LF sawtooth wave into the control voltage input to pulse width modulation (PWM). Open the attenuator associated with that jack.



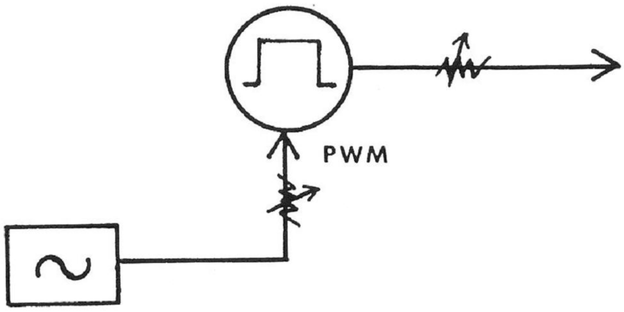
Since a positive-going low-frequency sawtooth wave gradually rises in voltage and then instantly drops, the effect here will be to sweep the duty cycle of the audio-frequency pulse wave and then instantly drop back to its starting point, a pulse width of 10%. The attenuator associated with the pulse width control input will determine how much of the pulse width is swept by the LF sawtooth, and the LFO's frequency slider will determine the rate of sweep.

You can lower the duty cycle to 0% (e.g., input −1 control volt when the duty cycle is at 10%; at the rate of a 10% change per control volt the duty cycle will drop to 0) or raise it to 100% (e.g., input +1 control volt when the duty cycle is at 90%). However, there will be no sound from the pulse wave output if the duty cycle is at or less than 0% or at or more than 100%.[10] Normally you would want to be careful not to voltage-control the pulse width all the way down to 0% or up to 100% (although there's nothing wrong with doing that if you need DC).

EXPERIMENT #14B: LF sine wave controlling pulse width

Patch an AF pulse wave out. Set the pulse width attenuator at 90%. Patch a very slow LF sine wave into the CV input to pulse width.

[9] If the pulse wave were LF, its effect as a control voltage would change with pulse width modulation.
[10] At 0% or 100% the pulse becomes DC.



You will hear the duty cycle vary through some percentages and then you will hear nothing for awhile; then the process will repeat itself. When the sine wave is negative it goes from 0 to −5 volts, which means the duty cycle will vary from 90% to 40%. However, when the sine wave goes positive, anything over +1 volt will drive the pulse width up to and over 100%, and you will hear nothing until the sine wave drops back to +1 volt or lower.

As a general rule, if you want to sweep a pulse width duty cycle with a voltage that alternates positive and negative (typically a sine or triangle wave), start with the duty cycle at 50% and you will never go lower than 0 or higher than 100%. If you want to sweep the duty cycle with a wave that is a positive-going sawtooth or pulse, start at 10% and make sure the attenuator setting doesn't let so much control voltage through that the duty cycle would exceed 100%.

Any voltage may control pulse width. Envelope generators and AF waves as control voltages of pulse width are discussed in greater detail in section IV of this chapter, and in Chapter 6.

D. Tuning and Beat Frequencies

Tuning two or more oscillators to unison or some harmonic interval is a two- or three-step process.

Step one is an approximate tuning using just your ears and the coarse frequency attenuators of both oscillators. You must develop a musical sense such that you can tell when the pitches generated by two VCOs are in the same octave and are relatively near each other.

The second step is to use the fine tuning attenuator of one VCO to bring the two as close as possible. At some point (when the pitches get within about 12 Hz. of each other) you will begin to hear the phenomenon known as "beating," an oscillation at a frequency that equals the difference in frequencies of the two VCOs. For example, if the frequencies of the two VCOs were 300 Hz. and 306 Hz., the beat frequency would be 6 Hz. You want to get that "beating"—called the beat frequency—as slow as possible. The more in tune the VCOs, the less the difference in frequencies and the slower the beat frequency.

The third step is to route the two VCO outputs

Moog Model 921 VCO

through a balanced modulator (if you have access to one) and open all the appropriate attenuators (Figure 3-3). You will hear further beating, which can be made slower by further attenuation of the fine tuning slider of one VCO.

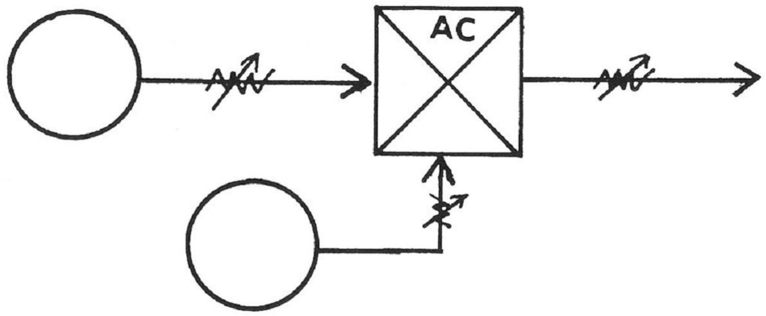


FIG. 3-3

Although a beat frequency of zero is theoretically impossible (unless you have a "synch" function on your VCOs that "slaves" the phase of one to the other), you can get the beat frequency so slow (one cycle every 5 to 10 seconds) that for practical purposes the VCO's are tuned to unison. If this is a bit unclear, listening to two oscillators being tuned to unison should be quite helpful. Two oscillators *slightly* detuned give a "fat" sound to a lead synthesizer line.

Beat frequencies also occur when two oscillators are tuned to harmonic intervals. Try tuning two VCOs to unison, a perfect fifth, a perfect fourth, and an octave.

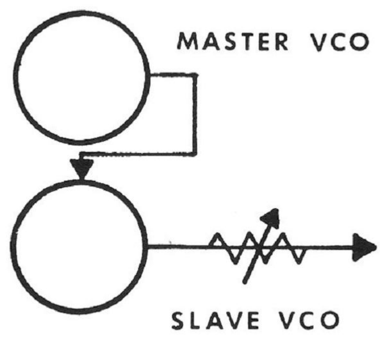
E. Oscillator Synchronization

Many synthesizers that have two or more oscillators have a synchronization (*synch,* sometimes called *reset*) function that allows one oscillator (called the *master*) to "control" another (the *slave*).[11] It does this by causing the output wave of the slave to reset to the beginning of its cycle at a time determined by when the master resets to the beginning of *its* cycle. At audio frequencies this creates extremely interesting and useful timbral possibilities; at low frequencies it creates the possibility of complex rhythmic structures.

In Figure 3-4 the AF sawtooth wave (a) is the slave and the square wave (b), which has a higher frequency, is the master.

[11] Synch capability is typically presented in one of two manners. For those hard-wired synthesizers that offer it, if a switch is off, two oscillators are independent; if on, one becomes master and the other slave. Examples: Oberheim SEM, OBX-a, Sequential Circuits Prophet 5, Octave CAT SRM, Moog Prodigy. All modular synthesizers offer a special synch input on particular oscillators. Of particular interest is the "phase-modulated synch" offered by Aries, a method of oscillator synchronization that offers much greater musical flexibility than typical oscillator synchronization. See Perrin, "Using Sophisticated Modifications in Creative Synthesis," *Polyphony,* 5, no. 3 (September -October 1979), 27.

Synch is noted in block diagrams thus:



Of course, the master VCO might be patched elsewhere if desired, as well as to the synch input of the slave VCO.

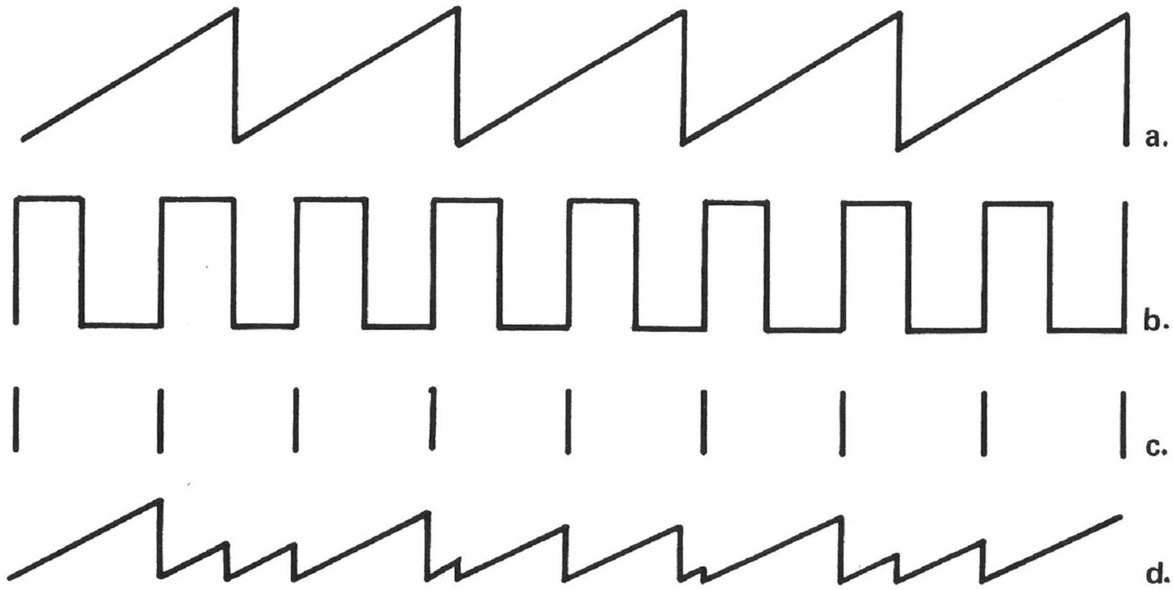


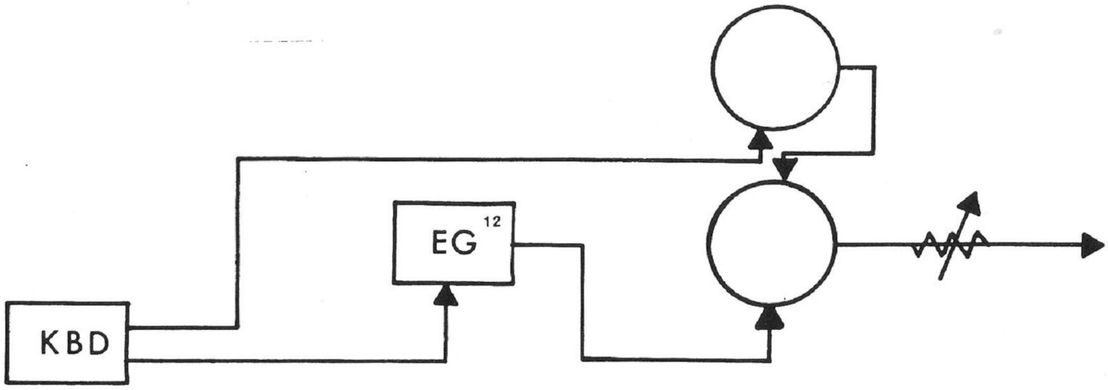
FIG. 3-4

The synch circuitry of the slave derives a trigger (Chapter 3, section IV-E) from the rising edge of each wave of the master (Figure 3-4c). It uses this trigger to tell itself when to reset. Note that the resultant wave (Figure 3-4d) has voltage drops both when the trigger rises *and* when it would have without the trigger. It is no longer a simple sawtooth; it is an unnamed periodic wave whose frequency is the same as that of the master. The "synch" circuitry forces the slave to assume the same frequency as the master. To do this it changes the shape of the waveform of the slave. At audio frequencies this will cause a unique timbre.

Figure 3-5 shows the typical patch in hard-wired synthesizers with a "synch" function on the VCOs.

The keyboard control voltage controls the frequency of the master VCO which, by definition, controls the frequency of the slave. The control voltage from the envelope generator would normally change the pitch of the slave, but here it cannot. The synch input overrides normal control input ability to change frequency. The envelope generator thus changes the slave's waveshape and timbre.

Note that synch allows the achievement of timbres completely different from those achievable by either filtering (next section) or modulation synthesis (Chapter 6). As such, it is a unique and important method of timbral construction.

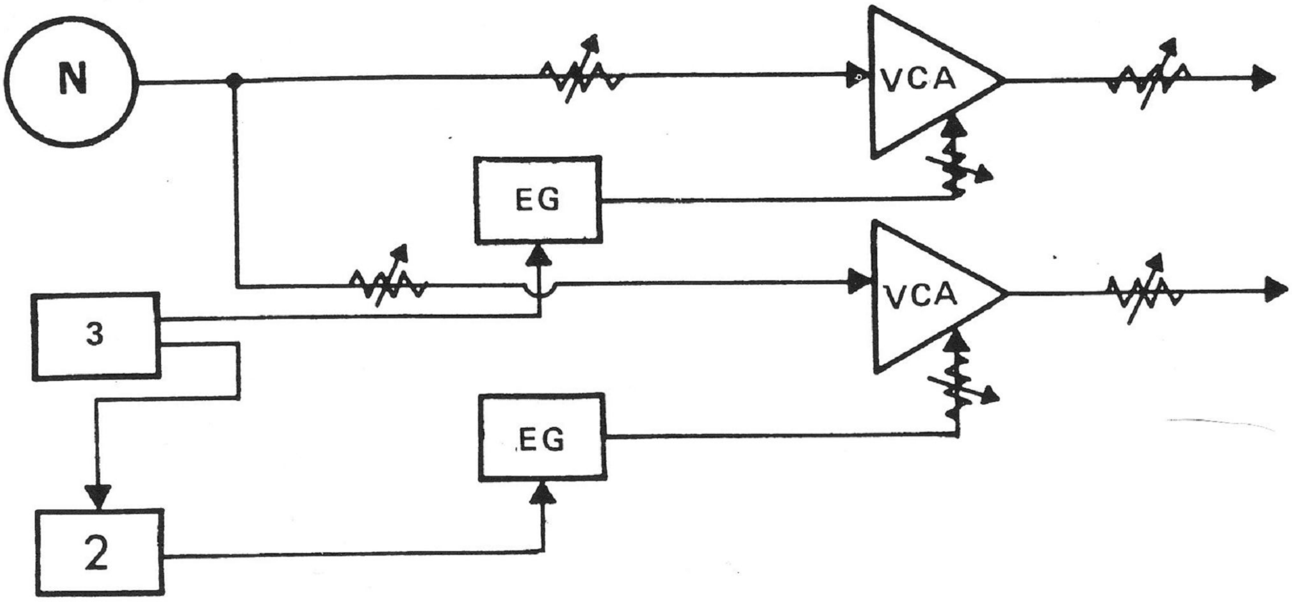


[12] We are a little ahead of ourselves here, having not yet discussed envelope generators. Experiments 15A and 15B will best be understood once you have completed section IV of this chapter.

FIG. 3-5

EXPERIMENT #15: Oscillator synch with LF waves

This experiment requires two VCOs with range switches that allow the VCO to become an LFO.



Tune two VCOs a fifth apart (e.g., tune one to C and the other to the next higher G). Then put both VCOs in their LF position, using the range switch.

The interval of a fifth is equivalent to a rhythmic ratio of 3:2.[13] *The LFO that had been generating the higher frequency (as a VCO) will "click" three times for every two "clicks" of the other LFO. This will cause the envelope generators to fire in a "three against two" rhythm, which will be perceived as a complex meter.*

If the LFOs had been tuned to other ratios, different complex meters would have resulted.

F. Miscellaneous Oscillators

There are several miscellaneous oscillators of which you should be aware:

1. As you discovered in Chapter 1, paragraph 24, at a certain point many VCFs will begin to oscillate and generate a sine wave of frequency F_C (which is defined on page 51).
2. There are frequently specific low-frequency oscillators (LFOs) that are used as control voltages. They may generate a specific LF wave (such as the LF sine wave on the Oberheim SEM, or the "internal clock" [which is just an LF square wave] on the ARP 2600), or there may be various LF waves available at different outputs of one LFO (such as the square and triangle waves available from the Multimoog's LFO). LFOs are sometimes called "modulation generators," since they generate a voltage that modulates a module.
3. The noise generator (NG) is not an oscillator although, like a VCO, it is a signal generator. For practical purposes, though, it can be thought of as an oscillator which generates all frequencies simultaneously.
4. Hard-wired synthesizers typically have an "octave" switch with four or five positions. Each position of the switch raises or lowers all VCO frequencies by exactly one octave.

[13] To learn more about the fascinating equivalence of pitch and duration see Duesenberry, "Rhythmic Control of Analog Sequencers," *Polyphony,* V. 4, no. 2 (September–October 1978), p. 26.

The switch positions may be called 4′, 8′, 16′, etc., a convention borrowed from organ terminology.

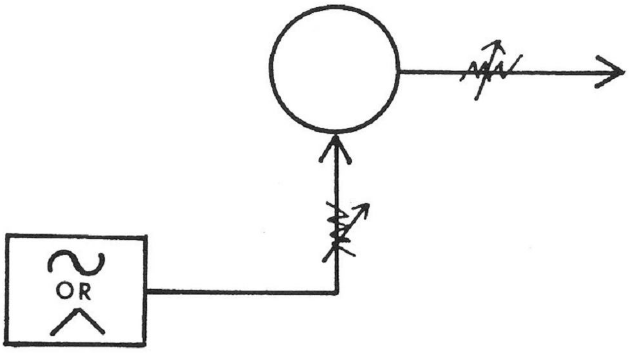
The VCF in oscillation is discussed in the next section of this chapter. The other miscellaneous oscillators are discussed in Chapter 5.

G. Other VCO Experiments

EXPERIMENT #16: Vibrato

Slight modulation of an AF wave by an LF sine or triangle wave within a frequency range of 4–8 Hz. is called vibrato. *It is a technique which acoustic instrumentalists have used for hundreds of years; when a violinist rapidly moves her/his index finger of the left hand back and forth over a string to change the pitch, vibrato is produced. The rapidly moving finger is the equivalent of the LFO, the violin string of the VCO.*

Since both a sine and a triangle wave typically fluctuate from positive to negative voltage, the pitch of the VCO will rise and fall as the LFO controls it. The depth, or intensity, of the vibrato is controlled by the attenuator between the LFO and the VCO.

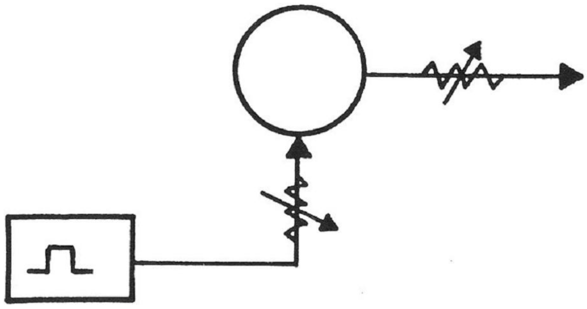


Set up the patch shown in the diagram. Once you are familiar with vibrato, vary the frequency of the LFO and the amount of modulation as you play the keyboard. Take your time. Discover nuances. What does a 4 Hz. vibrato do that is different from a 7 hz. vibrato? What is the effect of changing vibrato rate in real time (you can do this manually by varying the LFO's frequency attenuator, or by voltage control [on patchable synthesizers] by patching the KBD CV output, if necessary, to the KBD CV input of the LFO). Is there a difference between using a sine or triangle for your control waveform?

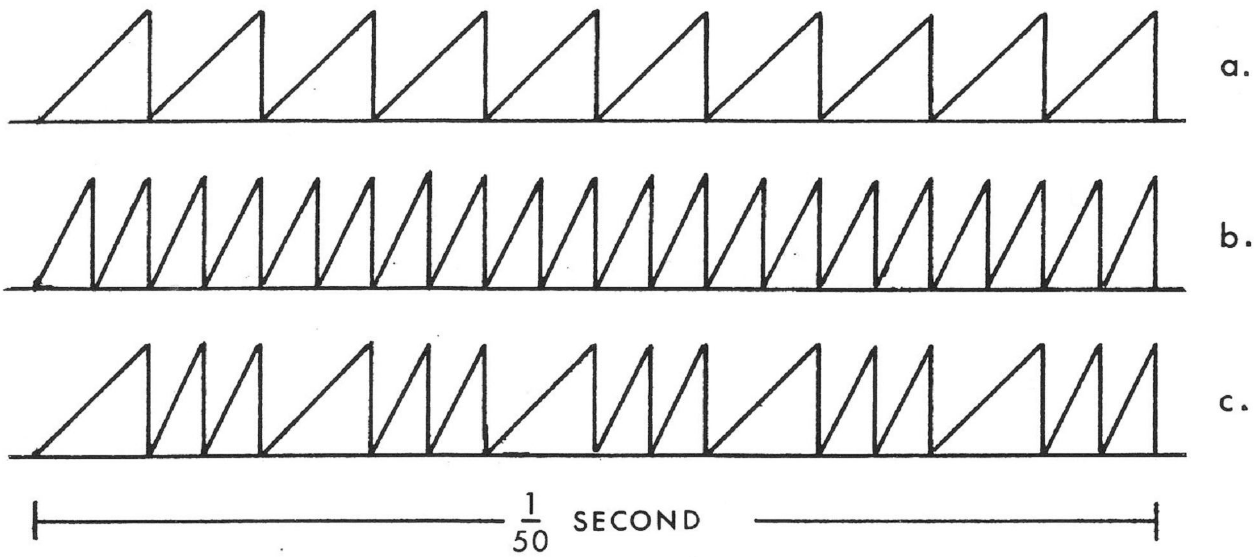
EXPERIMENT #17: Trill

A trill, a quick alternation of notes a tone or a semitone apart, can be produced by having the LFO control the VCO with a square wave. Since an LF square wave rises and falls instantly it will likewise cause the pitch of the VCO to instantaneously rise and fall.

In this experiment the VCO is being frequency-modulated by the LFO; that is, the LF square wave is causing a change in (modulating) the pitch (frequency) of the VCO. We can show frequency modulation graphically. Assume that the VCO is generating a sawtooth wave of 500 Hz. and that the control input attenuator into the VCO (which governs the amount by which the LFO can modulate the VCO) is set so that +1 control volt is allowed to pass to the VCO. Thus the VCO will



generate a frequency of 500 Hz. when the square wave is low (0 volts) and 1 KHz. when the LF square wave is high (and +1 control volt is input to the VCO at a rate of 1 volt/octave). Line a of the following diagram shows ten cycles of a wave whose frequency is 500 cycles per second; therefore the time shown is



1/50 second. Line b of this diagram would thus be twenty cycles of a wave whose frequency is 1000 cycles per second. Line a is a graph of the VCO's frequency when the LF square wave is low; Line b of the diagram is the VCO's frequency when the LF square wave is high. Line c shows these two frequencies alternating evenly, and so is a graph of the VCO being frequency-modulated, since it shows that for a given time period the frequency is 500 Hz., then for an equal time period it is 1 KHz., and then back and forth again and again.

EXPERIMENT #18A: Chords on a one-voice synthesizer

Listening to beat frequencies, tune three VCOs (if you have them) to a major triad, a minor triad, and other chords which come to mind. Don't be limited by traditional chords. Play the keyboard using alternately one or a combination of these VCOs.

EXPERIMENT #18B: Chords on a one-voice synthesizer

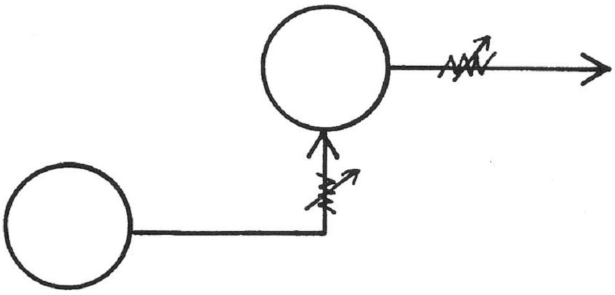
As a variation on Experiment 18A, tune the three VCOs to a major triad. Then use an LF square wave to create a minor third of the high note. For example, if one VCO is C and a second VCO is E, tune a third VCO to G and then let the LF square wave control the third VCO such that the pitch alternation is G to B flat. This will give you the effect of a dominant seventh chord: C–E–G–B flat.

EXPERIMENT #19: Drone

Insert a dummy plug (one end of a patchcord) into the keyboard control voltage input to a VCO (i.e., dummy out the keyboard control voltage). Now tune that VCO and another VCO to unison. If you play the keyboard, the first VCO will be a drone, remaining at the same pitch, while the second VCO is controlled by the keyboard control voltage.

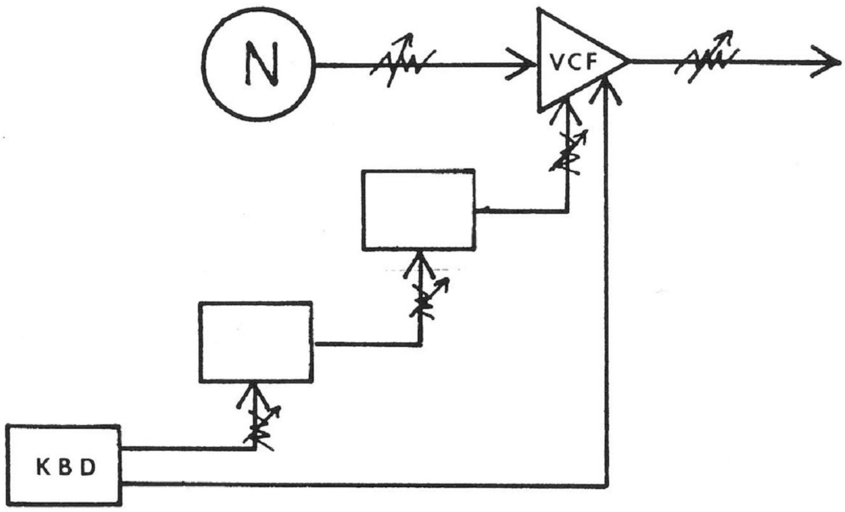
EXPERIMENT #20: FM

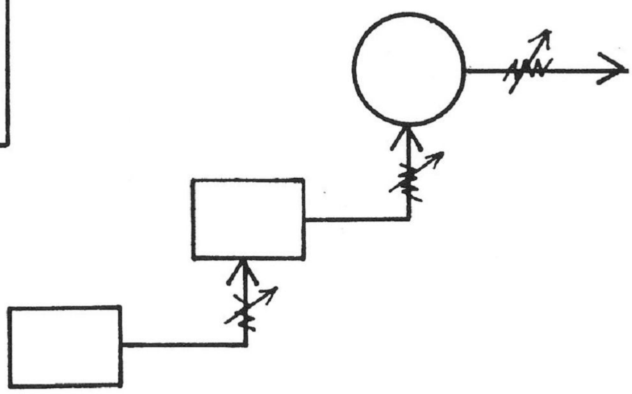
Although frequency modulation will be discussed in Chapter 6, you should become familiar with the effect of an AF VCO controlling (modulating) a second AF VCO (see the diagram).



EXPERIMENT #21: Rhythm

A very important use of control voltages is to control other control voltages. Experiment with the many varieties of the patch shown in the first diagram. You might also try the alternative shown in the second diagram; as you play the keyboard, you will be changing the frequency of one of the LFOs. This will mean you will have a different rhythm each time you play a different key.





II. THE VOLTAGE-CONTROLLED FILTER

The fact that each musical instrument has its own unique sound is largely due to the accentuation of particular harmonics over time. Most instruments have *formants,* specific bands of higher harmonics that are emphasized because of the shape and material of the instrument itself. Moreover, the character of a sound differs in its higher and lower registers. VCFs allow one to emphasize certain frequency components of a complex wave and deemphasize others, providing a wide array of timbral possibilities.

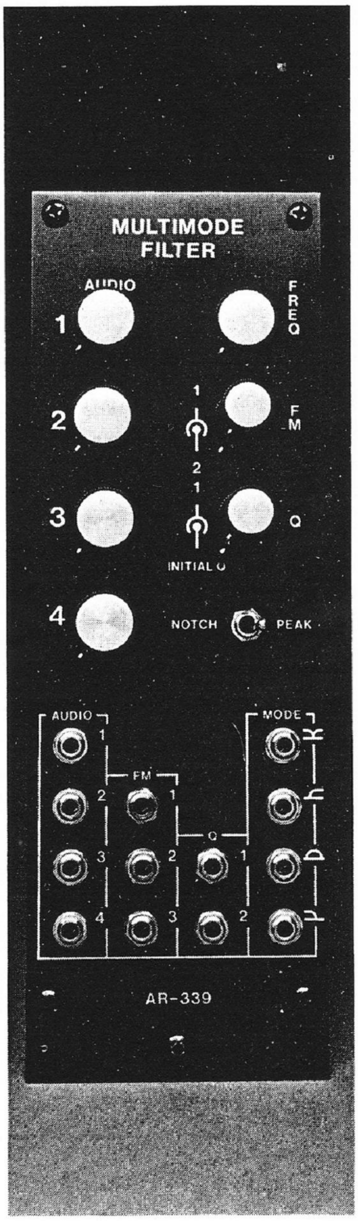
The purpose of any kind of filter is to allow the passage of some components of an entity and inhibit the passage of other components. A coffee filter allows the passage of liquid and inhibits the passage of grounds. A synthesizer filter allows the passage of some wave harmonics and attenuates (inhibits) the passage of others. For this reason filtering is sometimes called "subtractive synthesis." You have learned that a change in presence of harmonics will be perceived as a timbral change—a change in tone color. Similarly, a substantial change in the amplitudes of harmonics (which is what happens when harmonics are attenuated) will be perceived as a timbral change. Filters cause these kinds of changes. Having many filters available gives a musician important control over timbral construction.

There are several types of synthesizer filters and, in order to adequately describe any given filter, there are certain specific questions that must be answered:

1. In what manner does the filter attenuate harmonics? If it attenuates some but not all harmonics, which does it attenuate and which does it allow to pass?
2. What is the rate of attenuation? Are all attenuated harmonics totally gone or are there remnants left to further color the tone?
3. At what frequency does harmonic attenuation begin? You wouldn't hear much if there were complete harmonic attenuation of all frequencies over 100 Hz. Conversely, attenuation of all harmonics over 10 KHz. wouldn't have much of an effect either. (Why not?)
4. Does the filter have an associated resonance circuit and, if so, can the filter oscillate?

A. Cutoff Frequency (F_C)

The most common synthesizer filter is a voltage-controlled low-pass filter;[14] this filter will pass all frequencies up to a certain point; frequencies higher than that point are attenuated, and the point is determined by the amount of control voltage input to the filter. Recall that, except for sine waves, all waves have higher harmonics associated with them. These vary in number and amplitude depending on the type of wave.



Aries AR-339 Multimode Filter

[14] Other synthesizer filters are discussed in part F of this section.

The unique timbre of a given wave is a function of the number, frequencies, and amplitudes of these harmonics (higher-frequency components). A low-pass filter passes lower frequency components but attenuates the higher ones, and the point at which harmonic attenuation begins is called the *cutoff frequency*[15] and is written as F_C. Thus a low-pass VCF is a frequency selective attenuator, removing frequency components of the signal above the F_C.

The F_C is initially determined by the setting of a coarse tuning frequency attenuator, also called the cutoff frequency attenuator, associated with the VCF. A VCF fine tuning frequency attenuator allows one to be quite specific about the F_C. Since the range of the coarse frequency attenuator typically is 10 Hz. to 10 KHz., the initial F_C can be anything from subsonic to almost supersonic.

B. Harmonic Attenuation

The *rate* at which harmonic attenuation occurs is measured in decibels per octave (db/octave), and the specific rate of attenuation of many low-pass VCFs is −24 db/octave. A filter that attenuates frequencies at a rate of −24 db/octave is called a *4-pole filter.* Recall from Chapter 2, section III-C, that 20 db is equivalent to a power multiple of 100 (10^2) and that 30 db is equivalent to 1000 (10^3). Thus 24 db is equivalent to a multiple of about 250 times, so a signal of −24 db is about 1/250 as strong as one of 0 db. The signal level before any harmonic attenuation begins is referred to as 0 db. One octave higher than the F_C, the signal is "down 24 db." All frequencies below the F_C are passed unattenuated.

EXAMPLE:

Say you input a 500 Hz. sawtooth wave into a low-pass VCF whose rate of attenuation is −24 db/octave and that you have set the F_C at 1500 Hz. Remember that a 500 Hz. sawtooth wave has harmonics at 1000, 1500, 2000, 2500 . . . Hz. Since the F_C is 1500 Hz., all harmonics up to that point will pass unattenuated. Thus the fundamental (the first harmonic), and the second and third harmonics will be fully passed. All harmonics over 1500 Hz. will be attenuated at a rate of −24 db/octave. Since one octave over 1500 Hz. is 3 KHz., the harmonics at that point will be −24 db with respect to the unattenuated signal; that is, they will be 1/250 what they would otherwise have been. In between 1500 Hz. and 3 KHz. harmonic attenuation will be taking place, but those harmonics (in this example the fourth and fifth harmonics) will be less attenuated than the sixth harmonic. Similarly, harmonics higher than 3 KHz. will be even more attenuated.

Figure 3-6 is a graphic representation of the effect of a low-pass filter whose *slope* (rate of attenuation) is −24 db/octave. Note first that all frequencies below the F_C are fully passed. Harmonic attenuation begins at (actually slightly before) the F_C, and the rate of attenuation is −24 db/octave.[16]

[15] To be completely accurate, harmonic attenuation begins slightly before F_C, since F_C is defined as already being −3 db. For the purpose of this introduction to audio synthesis, however, assume that F_C is where harmonic attenuation begins.

[16] Some synthesizer filters have slopes of −12 db/octave (e.g., Oberheim SEM, E-mu's low-pass filter on their universal filter module); they are *2-pole filters.* Lag processors or the color attenuator of a noise generator are really one-pole filters which have a slope of −6 db/octave.

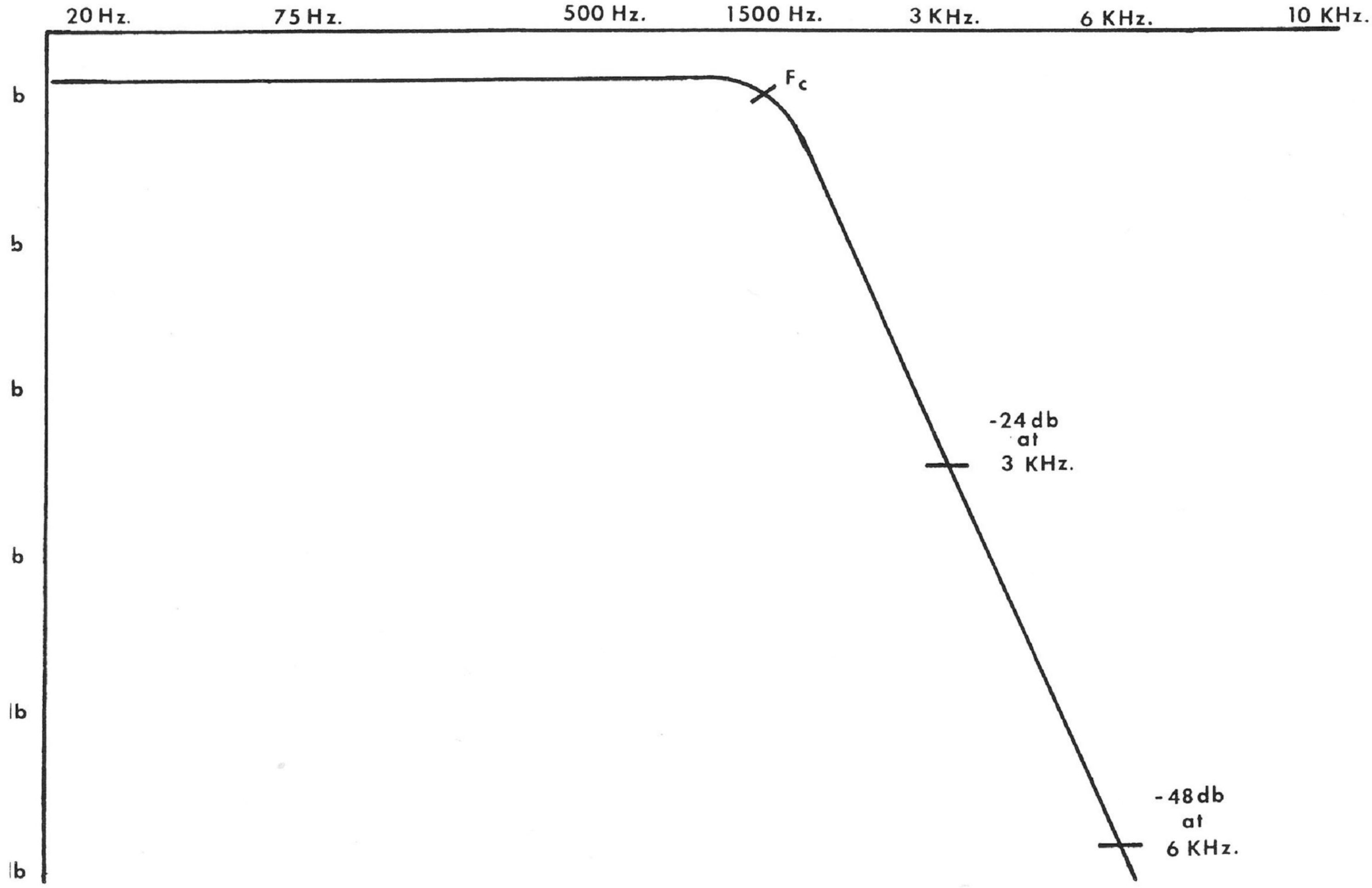


FIG. 3-6

At two octaves above the F_c the harmonics have been attenuated −48 db, which means their power is about 1/100,000 what it would normally be if there were no harmonic attenuation. Remember, however, that even with no attenuation the sixth harmonic of a sawtooth has an amplitude only 1/6 that of the fundamental. Thus at −48 db this harmonic has 1/100,000 of 1/6 the amplitude of the fundamental. For practical purposes it's gone.[17]

If you know the frequency and type of waveform input to a voltage-controlled filter, as well as the F_c and the slope of the filter, you can predict technically what the filtering effect will be.

EXAMPLE:

You input a 250 Hz. pulse wave with a duty cycle of 25% into a low-pass VCF whose slope is −24 db/octave, and you have set the F_c at 1 KHz. Since the duty cycle is equivalent to a fraction of one-fourth, you already know that the 4th, 8th, 12th, 16th, etc. harmonics will be completely missing. The harmonics that will be present are at 250 (the fundamental), 500, 750, 1250, 1500, 1750, 2250 . . . Hz. Remember that an octave is double the frequency of another frequency. Since the F_c is 1 KHz., the harmonic at 2 KHz. will be −24 db. In this case that is the 8th harmonic and it is missing, but you can make a graph nevertheless as though *the 8th harmonic were there. The graph in Figure 3-7 is an accurate representation of harmonic attenuation in this example.*

[17] By attenuating certain harmonics, a filter also shifts the phase of the composite waveform. See Wells, (7) in bibliography.

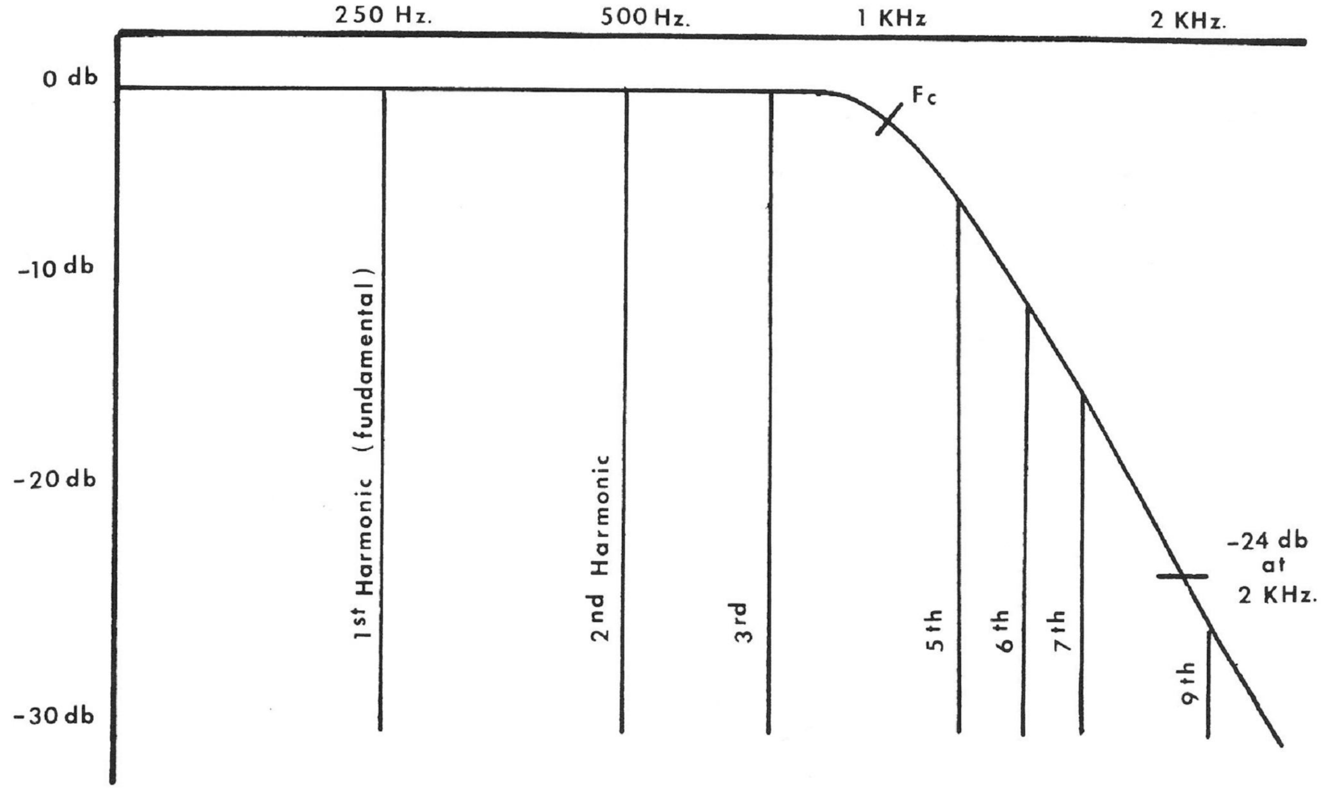


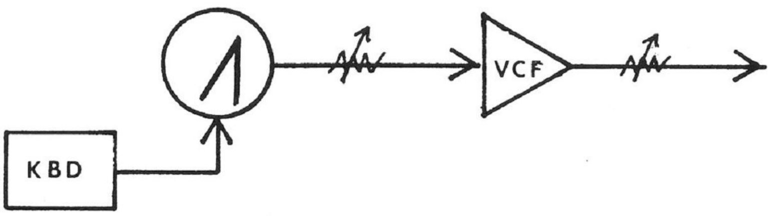
FIG. 3-7

C. Filter Tracking (1 volt/octave)

At this point an interesting question should arise. If all harmonics above the F_C are attenuated at the rate of −24 db/octave, which is a steep attenuation, how is it that when you set the F_C at a relatively low setting you can still hear high frequencies when you play a keyboard? Why haven't those higher-frequency components been completely attenuated? The answer is that most synthesizer filters are hard-wired or pre-patched to track a keyboard exactly as a VCO does. *In order for a filter to seem to maintain an even timbre, the F_C must rise or fall as the pitch rises and falls.* To put it another way, if a VCO tracks the keyboard at 1 volt/octave, then in order for the VCF to attenuate the same harmonics regardless of pitch, it too must track the keyboard at a rate of 1 volt/octave.

EXPERIMENT #22: Nontracking VCF

Input a 500 Hz. sawtooth wave into the VCF and set the F_C at 500 Hz. Put a dummy plug in the filter keyboard control voltage pre-patch (if there is one). In this on-figuration the F_C will not track at all; it will remain constant. Play a scale up the



keyboard, until you can no longer hear any pitch. The fundamental and its harmonics are all so much higher than the F_C that they are all completely attenuated.

Now patch the KBD CV to an unattenuated control input to the filter. The VCF will immediately track the keyboard; the F_C will be raised by exactly the correct *voltage needed to make the pitch being played on the keyboard a new fundamental with its own associated harmonics.*

Here is another example to make this more understandable:

Normally as pitch of a complex wave rises, your ears perceive harmonics of higher frequency. This makes sense. A sawtooth fundamental of 200 Hz. will have its tenth harmonic at 2 KHz. The amplitude of any tenth harmonic is quite low, not easily perceived. On the other hand, a sawtooth fundamental of 2 KHz. has 10 KHz. as its fifth harmonic. If each of these waves were input into a VCF whose F_C were 1 KHz. and did not track, harmonic attenuation of the 200 Hz. signal would begin at the fifth harmonic, but the fundamental of the 2 KHz. wave would already be −24 db. To keep timbre even throughout a wide pitch range, the F_C must vary with pitch (frequency). That is why the VCF typically tracks at 1 volt/octave.

Like the initial oscillator frequency of a VCO, the initial filter cutoff frequency tracks the keyboard at 1 volt/octave[18] and the voltage varies depending on all control voltages entering the VCF's control voltage mixer.

Figure 3-8 is a block diagram of a typical low-pass VCF.

D. Control and Audio Mixers

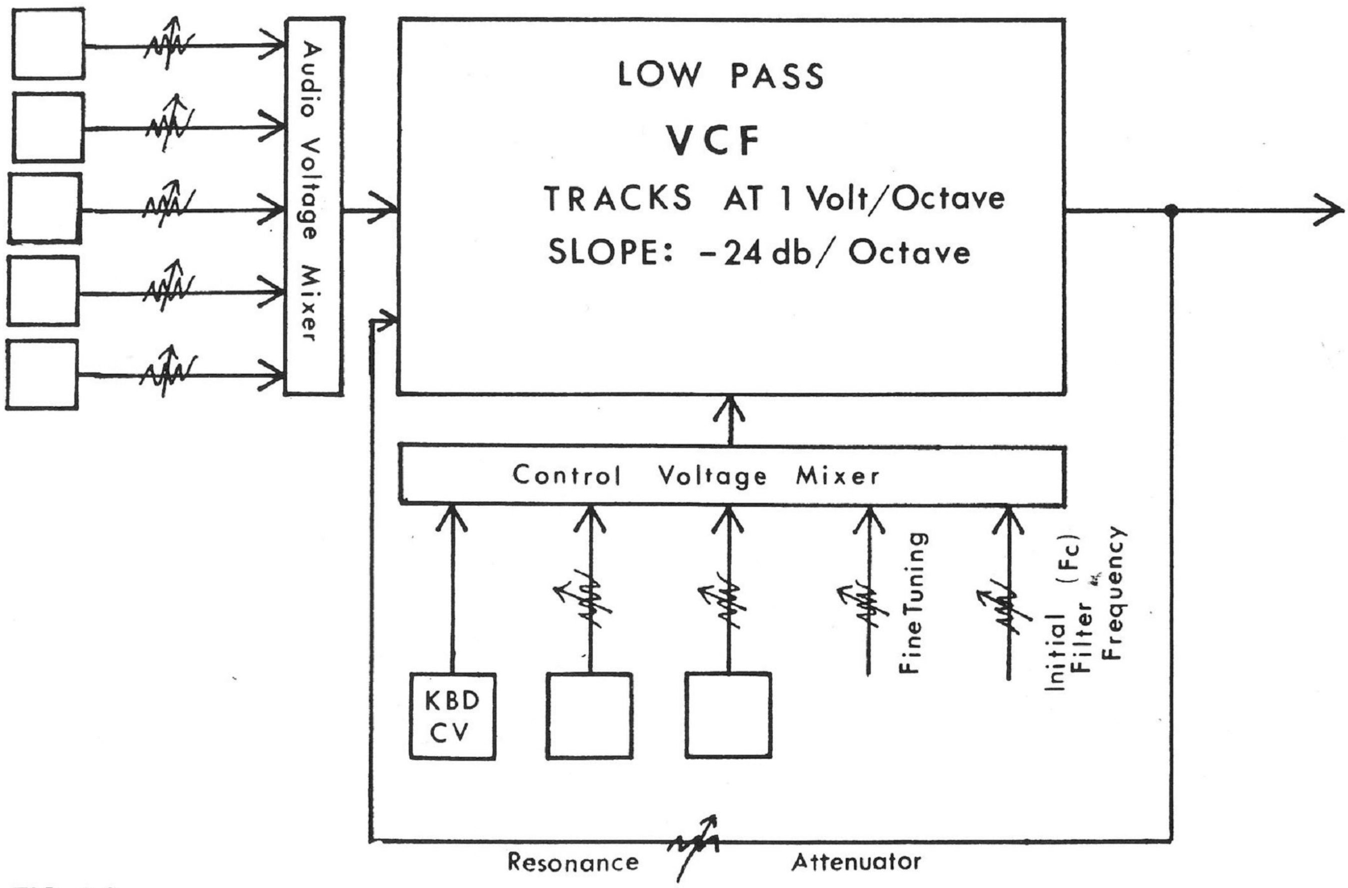


FIG. 3-8

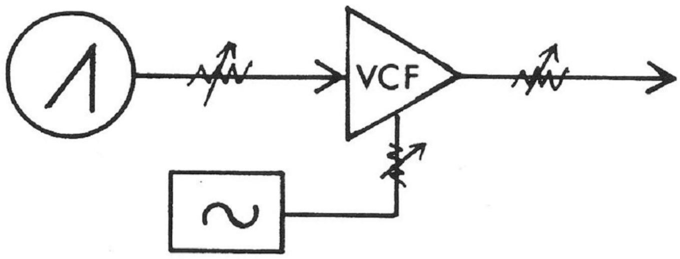
The control voltage mixer begins with the initial voltage input by the setting of the initial cutoff frequency attenuator. If that and the fine tuning attenuator together yield an F_C of 1 KHz., then about $+6\frac{1}{2}$ volts appear initially in the CV mixer (if you don't know why, see footnote 2 of this chapter). Any pre-patched or hard-wired unattenuated

[18] Some synthesizer low-pass VCFs do not automatically track a keyboard—e.g., Oberheim SEM, Korg-MS-20, modular systems without a pre-patch.

keyboard control voltage input allows the F_C to track as the keyboard is played at the rate of 1 volt/octave. Thus timbre will remain constant throughout the entire audio spectrum.

When controlling a low-pass VCF you must be aware of the nature of the controlling signal in conjunction with the initial F_C. If the filter is wide open (F_C = 10 KHz.) and you control it with an LF sine wave, the positive portion of that sine wave will have little effect; inputting +5 control volts into a low-pass VCF whose F_C = 10 KHz. would theoretically raise the F_C to 320 KHz., which well exceeds the filter's (and your ear's) upper limit of effectiveness. Of course, the amount of control voltage into the filter can be attenuated, but usually you will want to be sure that the total control voltage won't cause the F_C to exceed 10 KHz.

EXPERIMENT #23: Introduction to a low-pass VCF



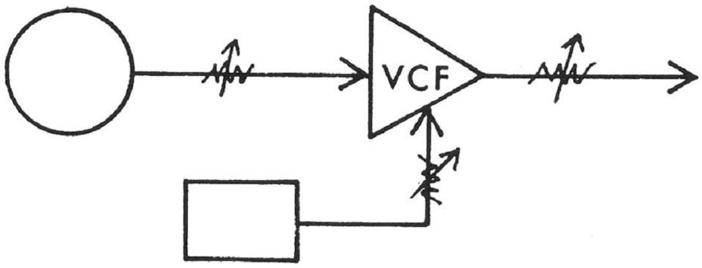
Input an AF sawtooth wave into the VCF. Set the filter's initial F_C at 500 Hz. Input an LF sine wave into an attenuated control input to the filter and open the attenuator slowly. Experiment with different F_C's and amounts of control voltage allowed by the attenuator.

What happens when the initial F_C = 10 Hz. and the CV attenuator from the LFO into the VCF is fully raised? Can you explain why this is happening?

All LF control voltages will open and close a low-pass VCF in proportion to their own voltage characteristics: A positive-going LF sawtooth wave will gradually open the filter and then instantaneously drop back to the original F_C. Therefore if you control the filter with that LF sawtooth, either your initial F_C would have to be 10 Hz. (in order to fully appreciate the filter sweep), or you would have to attenuate the sawtooth control voltage input (which would yield a different effect, because instead of the ten-octave filter range, only a few octaves would be swept), or the filter would open so wide that for some time of the sawtooth's cycle there would be no perceived filtering.

An LF pulse wave functioning as a control voltage causes the filter instantaneously to go from one F_C to a higher one and back again. The duration of the high or low F_C is a function of the duty cycle of the pulse wave. How far the F_C travels depends on the control voltage attenuator setting. An LF triangle control voltage would be similar to an LF sine, as Experiment #24 will show.

EXPERIMENT #24: LF waves controlling a low-pass VCF



Input each wave successively, in its LF, as a control voltage to the VCF. Make sure there is some audio signal to the VCF as well.

If an envelope generator (EG) controls a VCF, the F_C will vary in direct proportion to the control voltage allowed by the EG attenuator. An EG typically has the potential to output 10 control volts. Remember when mixing all those control voltages (the initial voltage, the keyboard control voltage, and any other control voltage) that as a rule the more control voltage input to the CV mixer, the lower the initial F_C should be. This combination will make for a more striking filter movement. Envelope generator control of the VCF is discussed in section IV of this chapter.

VCFs typically have attenuated audio inputs, all of which are mixed in the VCF's audio mixer. Any pre-patches should be considered items of convenience only. In the exercises at the end of this section you can explore other possibilities.

E. Resonance

Most synthesizer filters have an associated resonance circuit. (The resonance attenuator is sometimes labeled "Emphasis," since its function is to emphasize certain harmonics.) There is in such a VCF a connection that feeds the filter output signal back to the input (Figure 3-9). The amount of feedback is controlled by the resonance attenuator. It's like a PA system that begins to howl. In that case the feedback has its input at the microphones, the signal circulates through the amplifier and out acoustically and then back into the microphones again. If the loop becomes closed, the PA system will howl at its resonant frequency.[19] The nooks and crannies of many acoustic instruments (e.g., violin, saxophone, bassoon) make up a complex resonant filter. If a given note has harmonics whose frequencies lie within those resonant areas, the harmonics will be emphasized. This is one of the reasons that acoustic instruments have such complex timbres.

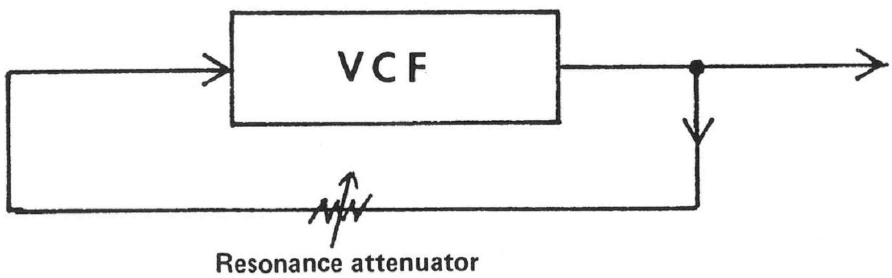
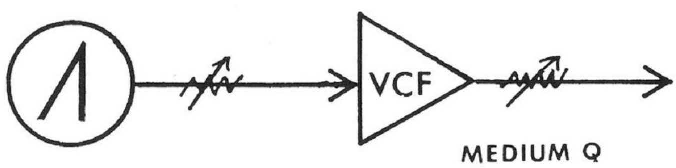


FIG. 3-9

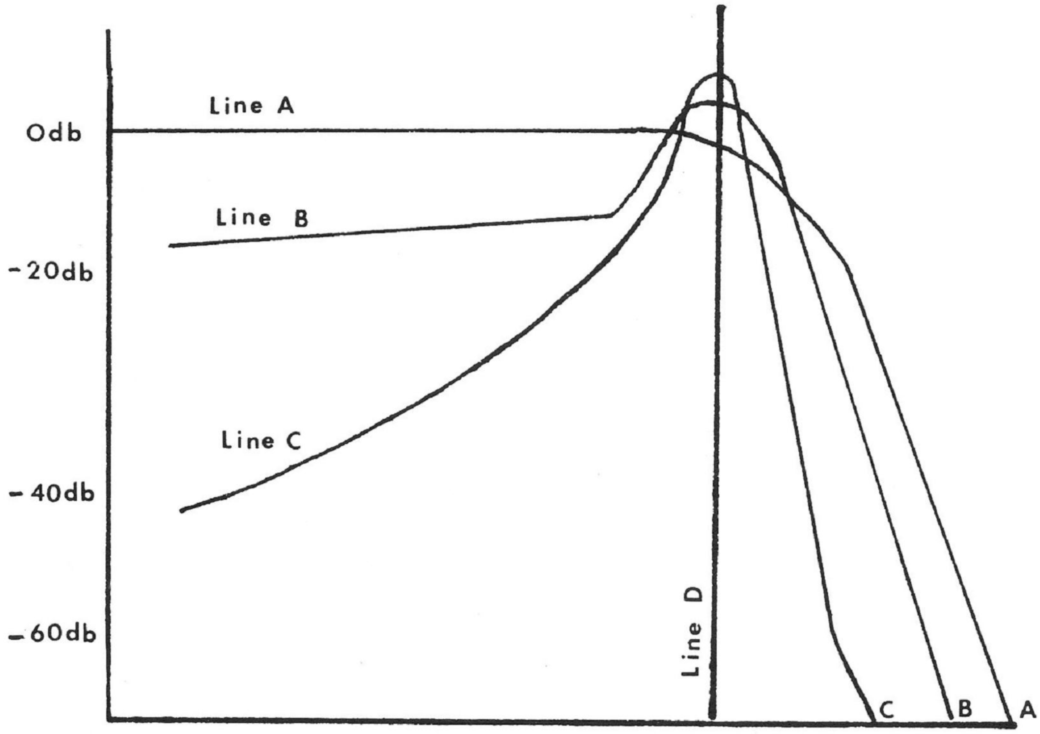
EXPERIMENT #25: Resonance (Q)



Patch an AF sawtooth wave into an audio input to the low-pass filter. Set the initial F_C at about 1 KHz. Open the resonance attenuator halfway (medium Q). Leave the synthesizer like this.

[19] Every physical object has at least one frequency at which it will vibrate, its resonant frequency. This is why a bookshelf rattles when a certain bass note is sounded. On the TV show *Star Trek* a phaser worked by finding the resonant frequency of a human being and starting her/his molecules vibrating.

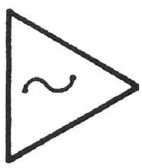
The first thing you probably noticed was that it seemed that the volume decreased as you opened the resonance attenuator. The following graph shows what happens to the filter output as you open the resonance attenuator more and more. Line A shows a typical low-pass filter response curve. As you increase the resonance by opening the attenuator, the signal around the F_C is boosted slightly and the rest of the signal is otherwise attenuated (notice that most of line B is down some db with respect to line A). The total energy available remains the same, but



less and less goes into frequencies other than the F_C and more and more goes into the narrow band of frequencies around the F_C. As you continue moving the resonance attenuator to the right, you hear even fewer frequencies, but the F_C becomes quite pronounced (line C).

At this point close the attenuator over the AF wave that you have input to the VCF. Now there is no audio signal being input to the filter. Continue opening the resonance attenuator. Finally at some point all frequencies except the F_C are completely attenuated; the filter emits a pure tone of frequency F_C. Since there is no audio input, the filter itself must have become a sine wave oscillator (line D).[20]

The symbol for resonance is Q. If Q is not mentioned in a patch, then there is no Q. If the resonance attenuator is used, then the amount will be called low Q, medium Q, high Q, or maximum stable Q. If the filter oscillates, it is represented thus:



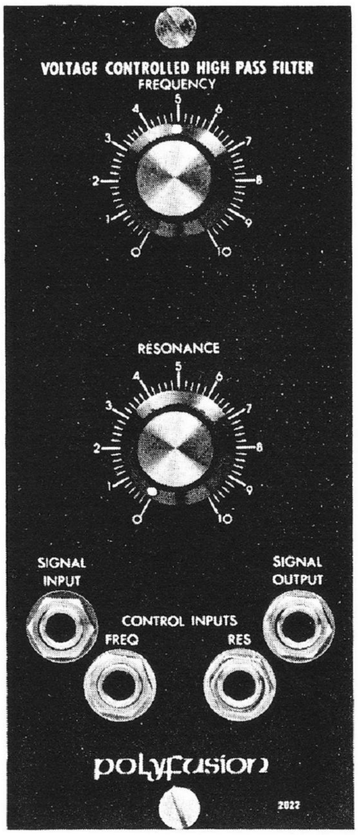
Once the filter is oscillating, you can vary the frequency of the output sine wave by changing the voltage in the filter's CV mixer; you can move the coarse frequency at-

[20] Some synthesizer VCFs have a resonance circuit but will not go into oscillation, e.g., Oberheim SEM and OBX, KORG MS-20. These filters could oscillate, but the controls on the front panel will not go far enough to allow oscillation; they stop at about the point of maximum stable resonance (see Experiment #33). This is typical of synthesizers with 2-pole filters.

tenuator or input keyboard or any other control voltage and the pitch will change at a rate of 1 volt/octave.

F. Synthesizer Filters Other Than a Low-pass Filter

There are several types of filters other than low-pass that synthesists typically use. A high-pass filter is just the opposite of a low-pass filter; it attenuates all frequency components *below* the F_C (generally at a rate of 12 or 24 db/octave) and allows the higher-frequency components to pass (Figure 3-10a). A low-pass and high-pass filter *in series* create a band-pass filter, which allows only a particular band of frequencies to pass, attenuating frequency components on either side of the band (Figure 3-10b). A low-pass and high-pass filter *in parallel* create a band-reject filter, which is the opposite of a band-pass filter; it allows all frequency components to pass except for a particular band (Figure 3-10c). If the slope of the band rejected is very steep (typically −40 db/octave or more) the band-reject filter is called a notch filter. A typical application of a notch filter is to eliminate 60-cycle hum from an otherwise useful circuit.[21]



Polyfusion Model 2022 VCF

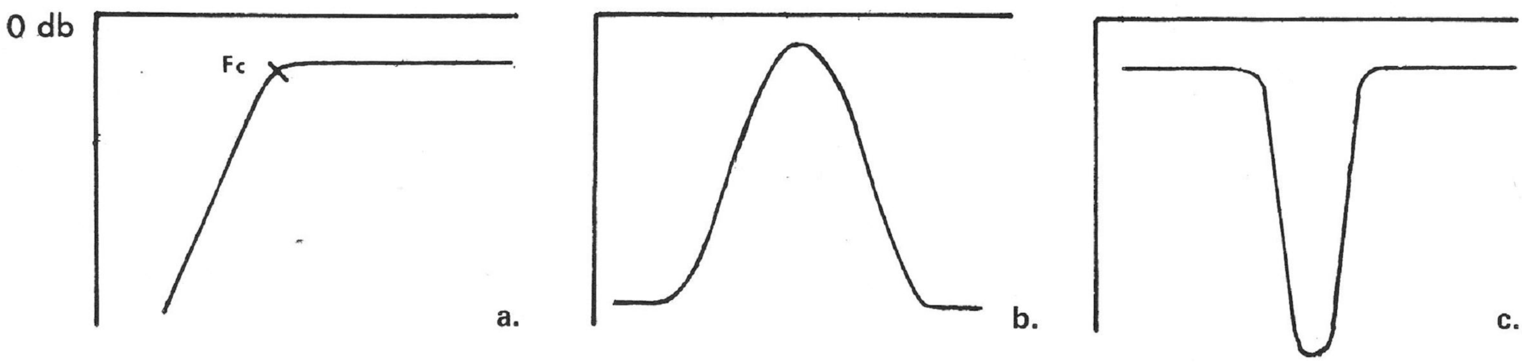
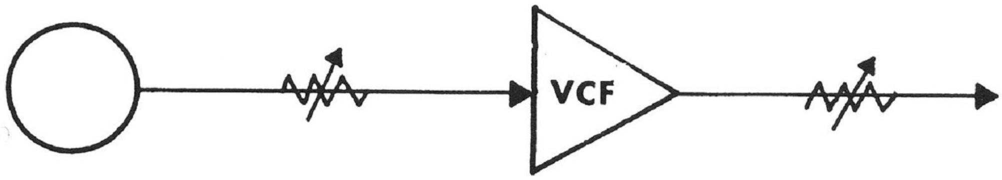


FIG. 3-10

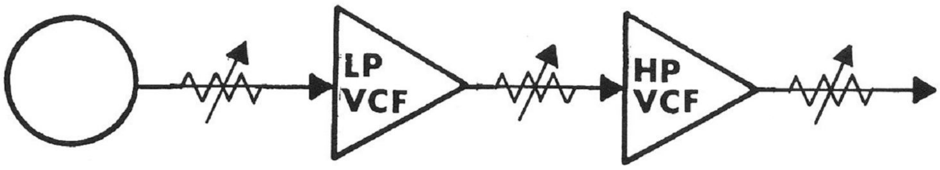
EXPERIMENT #26: Different types of filters

Experiment with all the filters you have available; begin to be able to hear the effect of high-pass (listen as the low-frequency components of the input wave drop out), band-pass, and band-reject filtering.

[21] VCFs are presented in many different configurations. The ARP Odyssey and KORG MS-20 include a high-pass and low-pass filter in series (creating a band-pass filter if desired). Serge-Modular, E-mu, and Aries offer multimode filters with simultaneous outputs for low-pass, high-pass, band-pass, and band-reject applications. The Oberheim SEM offers all four outputs although only one is available at any given moment. Serge-Modular also offers other filters with voltage-controllable slope (6 db/octave through high resonance) and band width. Although Moog offers only high- and low-pass filters, their "filter coupler" allows for band-pass and band-reject applications also. Moog also offers a non-voltage-controlled fixed filter bank of ten band-pass filters on its modular systems. Polyfusion's variable formant filter, also not voltage-controllable, allows one signal to be processed by three separate bandpass filters, each of which has independently variable center frequency, bandwidth, and amplitude.



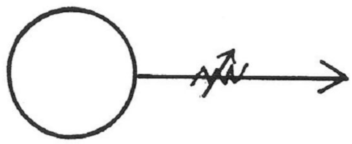
If you have different filters, connect them in different ways. For example, this patch yields a band-pass filter.



G. Other VCF Experiments

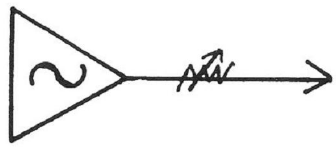
On various occasions in this section mention has been made of having the VCF begin to attenuate harmonics at a specific cutoff frequency: For example, if a VCO is tuned to any given frequency, having harmonic attenuation (that is, the F_C) begin two octaves higher. If you've wondered how to tune the filter so that harmonic attenuation begins exactly two octaves higher, Experiment #27 shows you two procedures.

EXPERIMENT #27: Tuning a low-pass VCF



1. *Patch a 200 Hz. wave from any VCO out. Open the attenuator and listen to the pitch. Close the attenuator.*

   Make the filter oscillate by opening its coarse frequency attenuator about 80% of the way, opening the resonance attenuator all the way, patching the VCF out, and then slowly closing the coarse frequency attenuator. At some point you will hear the filter oscillate as it emits a sine wave of frequency F_C. At first the pitch will be quite high.

   Once again open the other attenuator and compare the two pitches. Close the VCF coarse frequency attenuator until you can tell, by use of beat frequencies and your ear, that the two pitches are the same.

   Once the pitches are the same, open the VCF coarse frequency attenuator. The pitch of the sine wave from the oscillating filter will become higher; soon it will be one octave higher than the VCO, and then two octaves. At that point the VCF is tuned two octaves higher than the VCO.

   Close the resonance attenuator. You will no longer hear the filter oscillate but that doesn't matter. You know that the filter is tuned two octaves higher than the VCO and thus harmonic attenuation will begin at that point. Set up whatever patch you want.

2. *Once you are familiar with VCFs there is another, and much easier way, to tune a filter. Simply input a wave (let's say a sawtooth) to an audio input to the filter with the F_C at 10 Hz. and the resonance attenuator far to the right, at the point of maximum stable resonance. As you now open the VCF coarse frequency attenuator, you will hear the harmonics of the input wave in succession. If you want to emphasize the fourth harmonic, just stop moving the attenuator when you hear it.*

EXPERIMENT #28: Sine waves from a VCF

How can you get an AF sine wave from a synthesizer? There are three easy ways:

1. *Take it directly from a sine wave output, if one is available.*
2. *Put a VCF in oscillation as shown in Experiment #27, which gives you an AF sine wave.*
3. *Filter any wave down to its fundamental; you will get a sine wave of the same frequency as the wave being originally output.*

Output any complex wave into an audio input to the filter with the filter's coarse frequency attenuator all the way closed. Slowly open that attenuator and the first strong harmonic you hear will be the fundamental, a sine wave.

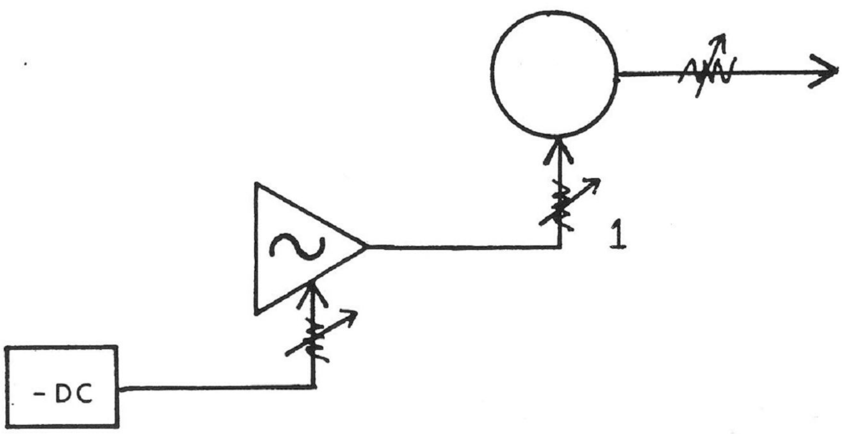
Needed: negative DC voltage

Just as it is possible to get an AF sine wave from a VCF, it is also possible to get an LF sine wave from a VCF. Here's how:

Put the synthesizer in neutral. Now make the VCF oscillate (generate a sine wave) as shown in Experiment #27.

Inputting negative DC voltage into a control input to the VCF will have the effect of reducing the frequency of the sine wave at a rate of one octave per control volt. As you open the control attenuator into the filter, the frequency of the sine wave becomes lower and lower, until it becomes a true LF sine wave.

To demonstrate this, patch any AF wave from a VCO directly out. Patch the output of the VCF to an attenuated control input to the VCO. Raise the control attenuator into the VCO to a level of 1.



You are now using the LF sine wave being generated by the VCF to impart vibrato to the VCO.

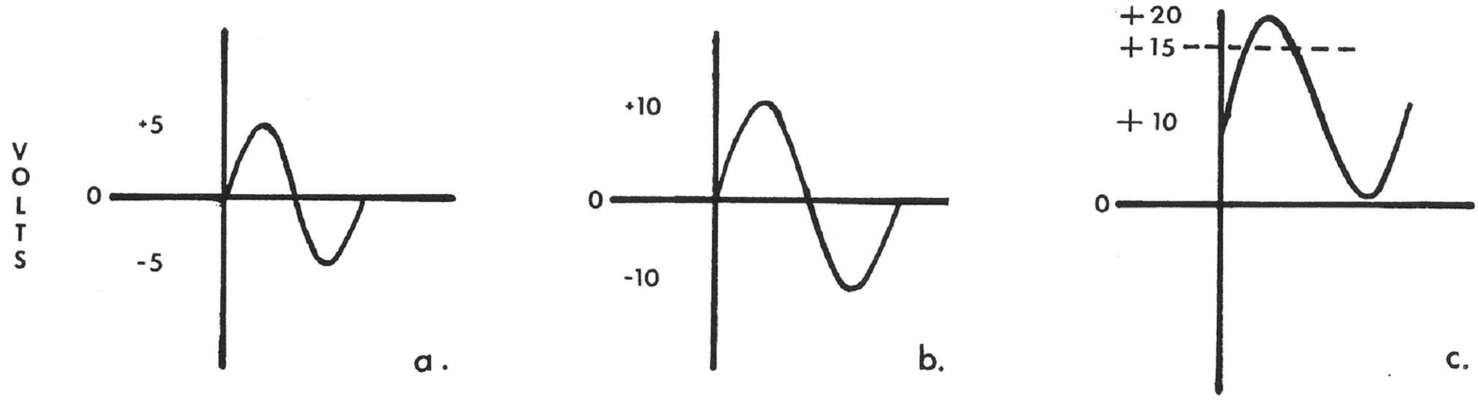
Although you could in fact use this application as shown, it's rather module-ineffective, using up an entire VCF module merely to impart vibrato to a VCO. The true purpose of this example is to illustrate the kind of thinking that will become necessary to use a patchable synthesizer effectively. What you have done is produce an LF sine wave from a source (the VCF) where you might never have thought one available. This kind of thing is possible again and again with patchable synthesizers. Often you will find you need something and it's not immediately apparent that it is available; creative thinking (affectionately known as "klugeing"; the verb is *to kluge*) will help a lot.

Almost all waves that synthesizer oscillators generate, whether in their low or audio frequencies, have a 10-volt p-p amplitude. You cannot simply input wave upon wave into a filter (or any module) because the module will have limits beyond which it will introduce a type of distortion known as *clipping*. The usable range of most modules before clipping is approximately +15 to −15 volts. The following example will demonstrate clipping and then show a technique for avoiding it. The technique is known as *offsetting* and will be discussed thoroughly in Chapter 5.

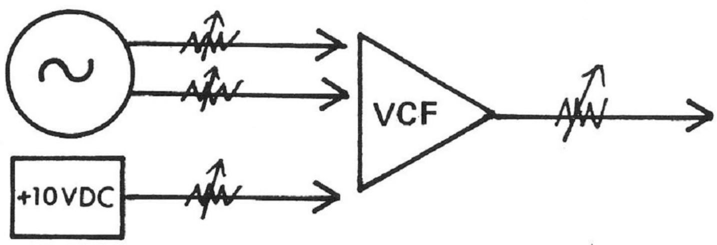
EXPERIMENT #29A: Clipping

Needed: multiple; positive DC voltage

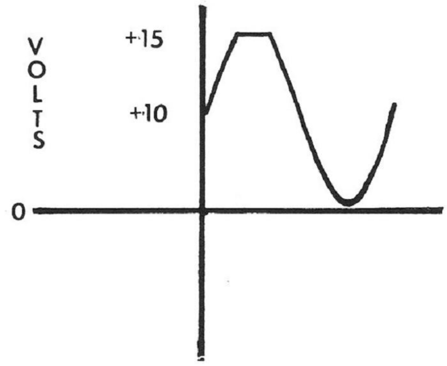
Patch an AF sine wave to a multiple. Patch two multiple outputs to two audio inputs to a low-pass filter. Open the filter all the way. Patch the VCF out. As you raise the first audio attenuator into the filter, you will hear the original AF sine wave. As you raise the second attenuator, you will hear the same frequency sine wave added to itself. If you used the technique of adding voltages shown on page 18 you would find that the effect of adding a wave to itself is to produce a wave of the same frequency and twice the amplitude. The difference is graphically demonstrated in the diagram below; a is the original wave, b the wave added to itself. The p-p amplitude of the doubled sine wave is 20 volts but, since the wave voltage fluctuates between +10 and −10 volts, no clipping occurs.



Patch +10 volts to an audio input to the filter. The effect of this is to offset the already amplified sine wave shown in b by an additional +10 volts. You have added +10 volts to whatever appears at the filter audio input. The entire wave has been raised +10 volts. This is graphically demonstrated by part c of the diagram. Since the filter clips at +15 volts and the sine wave now rises to +20 volts, clipping will occur. As you raise the attenuator over the audio input into which you have



patched the +10 volts, listen for a difference in timbre. The wave will not sound merely like a louder sine wave; you will hear harmonics being added. Since the VCF cannot accept signals that have a higher total voltage than +15 volts without clipping, the additional voltage is lopped off by the filter's limit. The result is graphically shown in this diagram:



The effect is to make something like a quasi-square wave out of the input sine wave and, since a square wave has certain harmonics associated with it, a timbral change will be perceived.

Clipping is one of various types of distortion. *There is nothing wrong with it and it may be used effectively in various ways. The important thing is to be aware of it.*

Any time you input waves into the audio inputs to the filter whose combined voltage exceeds some voltage (typically either +15 or −15 volts), you will have clipping. Thus merely inputting two unattenuated sawtooth waves fully would give you a +20 volt input and cause clipping. The effect here would be to cause a more hollow sound, because the resultant quasi-square wave would have fewer harmonics than either of the component sawtooth waves. (In case you are thinking you will almost always have clipping, that's what attenuators are for. Although a given wave may have possible p-p amplitude of 10 volts, you can input that wave into the filter and then attenuate the amplitude down to anything you want.)

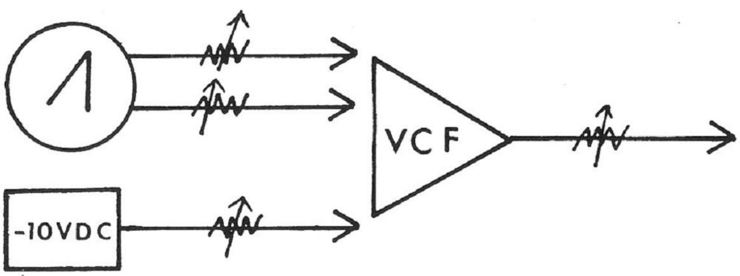
Just as you can cause positive or negative clipping by adding a positive or negative voltage to an audio input to the filter, so you can abate clipping by using a voltage to offset the combined audio voltages.

EXPERIMENT #29B: Abating clipping

Needed: multiple; negative DC voltage

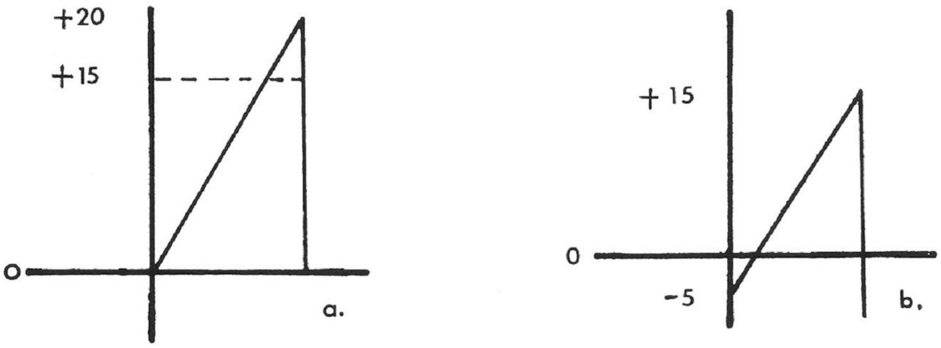
Patch an AF sawtooth into the multiple; come out the multiple with two patchcords to audio inputs to the filter. Now you've effectively got a sawtooth wave into the filter whose voltage range is from 0 to +20 volts, and clipping occurs.

Patch −10 volts to another audio input to the filter and slowly raise the attenuator over that input.



You are inputting negative voltage to the audio mixer to the VCF. When −5 volts have been input, the sawtooth wave will have been offset 5 volts (will have gone from a to b in the following diagram).

The p-p amplitude of the sawtooth hasn't changed, but its relative position within the filter has and clipping no longer occurs.

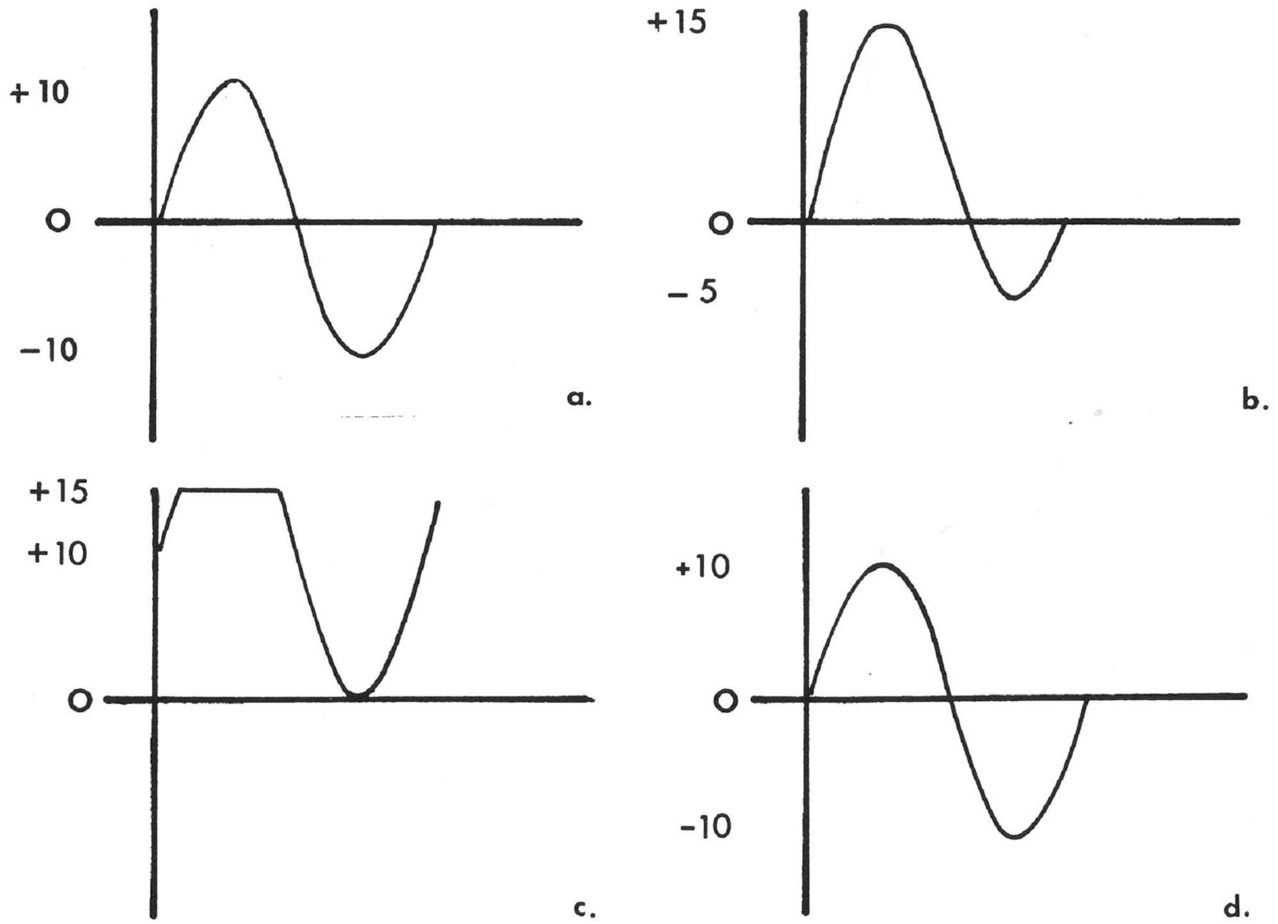


Although atypical, there's no reason you can't input an *LF* wave into an audio input into the filter. The effect it has will depend on the voltage characteristics of the input LF wave, but the general effect is to increase or decrease the voltage of whatever other wave(s) have been input into audio inputs to the VCF, in proportion to the voltage change of the LF wave.

EXPERIMENT #30: Voltage-controlled distortion

Needed: multiple

Going through the multiple, input one AF sine wave into two audio inputs to the filter. This will effectively give you one sine wave whose p-p amplitude is 20 volts, varying from +10 to −10 volts (diagram a). Now input a slow LF *sawtooth wave into an audio input to the filter. The effect will be to slowly raise the voltage level of the sine wave. When the sawtooth has reached half of its maximum voltage (i.e., +5 volts), the total positive voltage in the filter will be +15 volts and clipping will*



begin (diagram b). Clipping will continue as the sawtooth goes through its final 5 volts, reach a maximum (diagram c), and then instantaneously stop as the sawtooth drops to 0 volts (diagram d), beginning its cycle again.

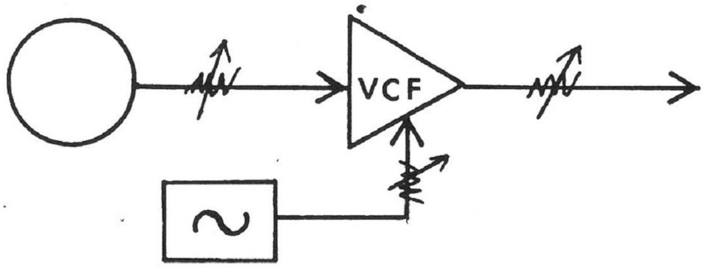
Just as you can control a VCO with an LF sine or triangle wave to impart vibrato, so you can control the VCF with an LF sine or triangle wave to impart timbral modulation. This is the primary reason that many synthesizer manufacturers pre-patch a sine wave to a control input to the filter. As the LF wave controls the filter, the F_C rises and falls at the frequency of the LFO, by an amount determined by the setting of the control attenuator into the filter. The timbre gets brighter or duller as harmonics are added or subtracted.

In addition, different effects will occur depending on the setting of the resonance slider. As Q becomes higher and higher, the rising and falling F_C will cause the filter to approach and recede from oscillation, thereby emphasizing different harmonics.

EXPERIMENT #31: Timbral modulation

Patch any AF wave into an audio input to the filter. Patch the filter out.

Set an LF sine wave to oscillate at about 3 Hz., the F_C at 1 KHz., the control attenuator into the filter at a level of 1, and the resonance attenuator closed. Become acquainted with the basic effect of timbral modulation.



Now methodically vary all four parameters (the VCO's frequency, the initial F_C, the control attenuator setting, and the resonance attenuator) to experience many different possible effects of timbral modulation.

In Experiment 22 you learned that inserting a dummy plug in the KBD CV input to the filter stopped the filter from tracking the KBD. Patchable synthesizers do not limit you to either no filter tracking or tracking at 1 volt/octave.[22] In a given situation you might want filter brightness to change but more slowly than the 1 volt/octave change of a VCO.[23] What is needed is an attenuator between the KBD CV output and the input by which the keyboard controls filter tracking at the 1 volt/octave rate.

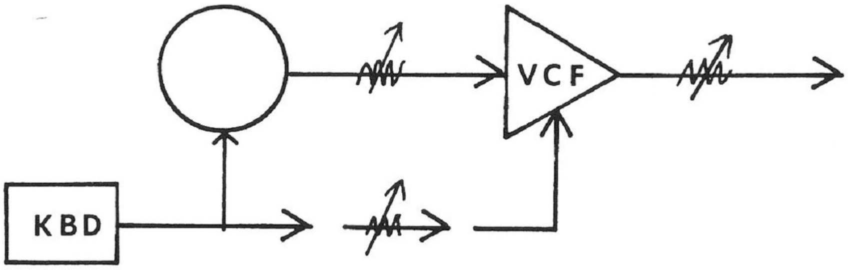
EXPERIMENT #32: Filter tracking at rates other than 1 volt/octave

Patch a complex AF signal into the filter. Open the filter and patch the filter out. Assuming your synthesizer has a pre-patch that allows for filter tracking of the

[22] Some hard-wired synthesizers have a switch that allows filter tracking at rates other than 1 volt/octave—e.g., the Multimoog, whose switch allows filter tracking at either .5 volt or 1 volt/octave.

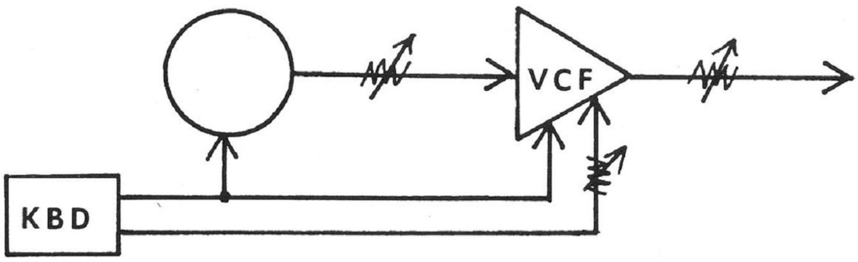
[23] For example, in a certain frequency range the timbre of a bassoon follows the pitch at a rate of about .7 volt/octave. If synthesizing a bassoon, you would want similarly to attenuate filter tracking.

keyboard, play the keyboard to familiarize yourself once again with the sound of the filter tracking at a rate of 1 volt/octave.



Since in this experiment you want the keyboard control voltage to control the filter's brightness at a lesser rate than 1 volt/octave, dummy out the pre-patched KBD CV to the filter and patch the KBD CV output to an attenuated *control voltage input to the filter. With the attenuator all the way closed, the filter will not track the keyboard at all; all the way open it will track the keyboard at 1 volt/octave. At any point in between it will track at a rate less than 1 volt/octave.*

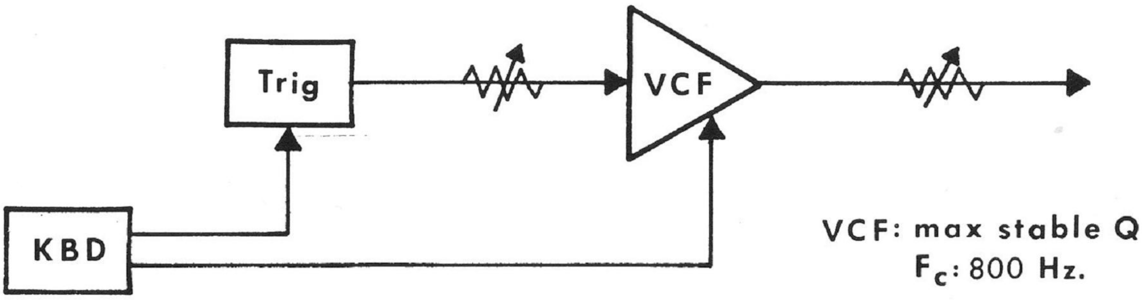
What if you wanted the filter to track at a rate greater than 1 volt/octave? Use the regular pre-patch to give you 1 volt/octave tracking and *a patchcord from the KBD CV output to an attenuated control voltage input to give you additional voltage to the filter every time you play the keyboard. This gives a very "hot" sound to a lead synthesizer line.*



EXPERIMENT #33: Percolator

Needed: trigger output jack

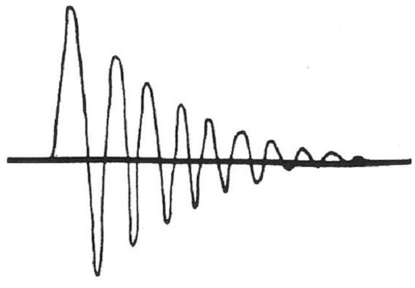
Finally, here's a cute little patch that illustrates maximum stable resonance, *the point just above which the filter starts to oscillate.*



Make the filter oscillate at a frequency of about 800 Hz. by using the resonance attenuator as in previous experiments. Slide the resonance attenuator slowly to the left, just to a point where the VCF no longer emits a pitch but where if you nudged the resonance attenuator just the slightest bit to the right the VCF would begin to oscillate. This is the point of maximum stable resonance, the point beyond which adding any resonance at all will cause the filter to oscillate.

At this point if the filter receives any trigger (section IV-E of this chapter), it will oscillate for a few cycles and exponentially decay. A simple way to input a trigger into the filter is to patch the trigger output to any audio input to the filter. If you do so *and play the keyboard (which should act as a control voltage to the VCF), you will get the famous "percolator" sound. Note that the higher the frequency at which the filter oscillates, the shorter and more percussive the event.*

What is happening here is that the noise in-herent in the filter is being filtered. Since the filter is highly resonant, a trigger allows the noise in the narrow pitch zone around the F_C to pass. The trigger inputs energy which dissipates exponentially. This is called a damped waveform. It's like giving a child on a swing a sudden push and then walking away. This is a form of pitched noise, of which you will learn more in Chapter 5.



DAMPED WAVEFORM

III. THE VOLTAGE-CONTROLLED AMPLIFIER

The first thing to understand about a VCA is that it does not amplify in the traditional sense of the term: the output of a VCA is not greater than the input. In fact, with one exception,[24] the greatest output the VCA can deliver is a voltage equal to that of the audio input.

We typically have a VCA in the audio path to control the gain (generally to shape the volume) of the audio signal. Depending upon the control voltage input to the VCA, the *shape* of the audio signal output from the VCA will be changed. The ratio of output to input is called the *gain* of the VCA, so the greatest gain possible with a VCA is a gain of 1.0, also called unity gain.[25] Unity gain says that the ratio of $\frac{\text{output voltage}}{\text{input voltage}} = 1.0.$

In fact, as you will see, in almost all cases the gain factor of a VCA will be less than unity gain. Since a VCA does not amplify, its purpose might be clearer if it were called a voltage-controlled attenuator. Its primary *purpose* is to allow the amplitude of the total instantaneous voltage appearing at the CV mixer to determine the output amplitude of the audio signal appearing in the audio mixer at that instant. How it does this will be discussed shortly.

Like the VCF, the VCA has separate audio and control voltage inputs that are summed in separate audio and control voltage mixers. The instantaneous sum of the signals appearing at the audio inputs appears in the audio voltage mixer; the instantaneous sum of the control voltages appears in the control voltage mixer. These two sums are then sent to a *multiplier,* a special submodule within a VCA (and *the* unique circuit in a VCA) that multiplies the sums of the instantaneous voltages in the audio and control voltage mixers and outputs a final voltage that determines how much and at what rate the VCA opens.

Just as with the VCF, no signal will pass through a VCA if it is completely closed. The only way a VCA opens is if some positive control voltage appears in the control voltage mixer. If the sum of the total voltage appearing in the CV mixer is either zero or negative, the VCA will not open and no signal appearing at the VCA's audio inputs will pass through the VCA. Thus, if the module is being used, the general way the VCA works is that the

[24] If +10 control volts are input to the exponential control input, the output may well be above +10 volts; it may even exceed +15 volts, the limit which most VCAs can accept as total input signal before clipping results.

[25] The VCA supplied with Moog modular systems (No. 902) has a gain factor of 2.0.

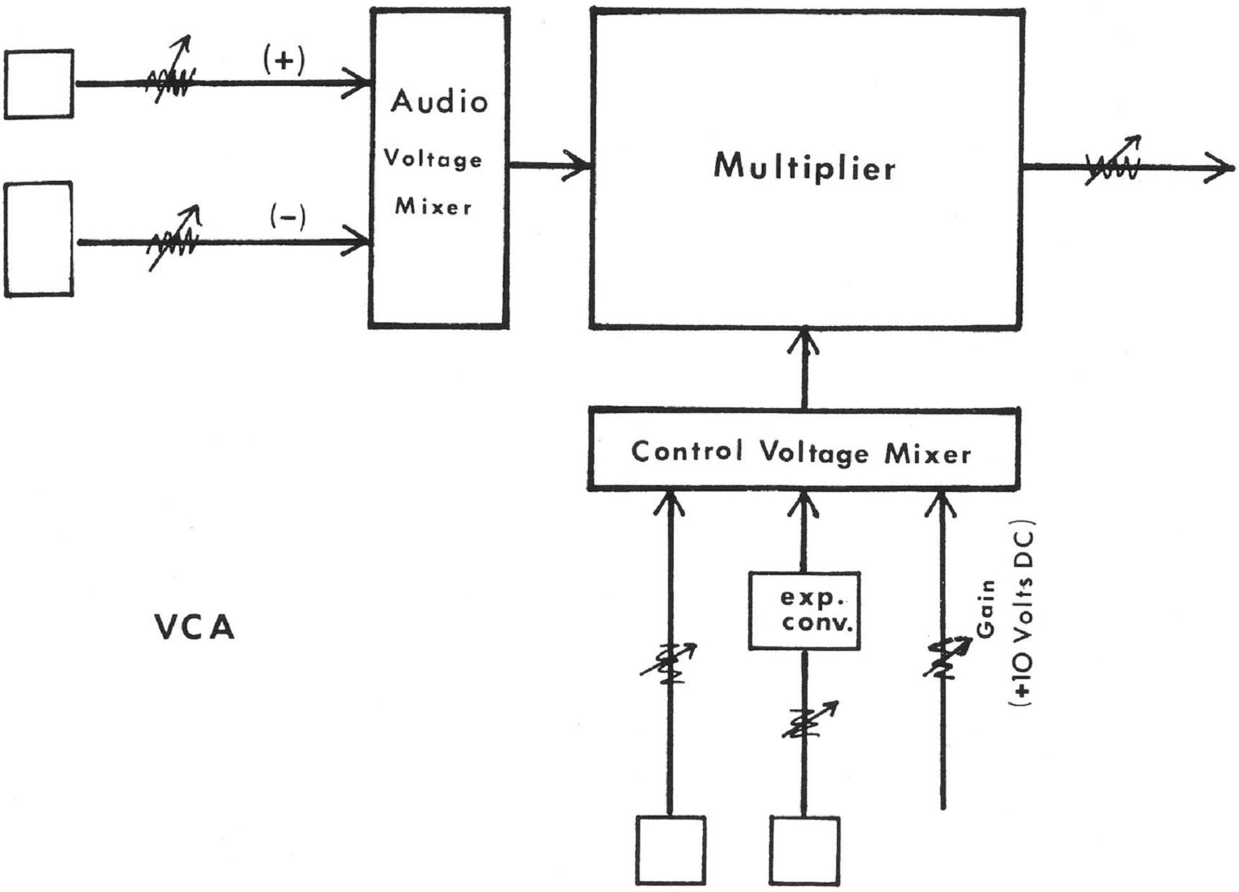


FIG. 3-11

audio signal is constantly present in the audio mixer; whether it passes through depends upon whether a positive control voltage appears in the control voltage mixer.

A. Inverting Audio Input

Audio inputs to VCAs are attenuated so you can allow all or only a part of each audio signal into the audio mixer. Note that in Figure 3-11 one audio attenuator has a (+) associated with it while the other has a (−). The first audio input is typical, but the other one is an *inverting input*. Any audio signal input into this jack will be totally inverted before reaching the audio mixer. Synthesizer VCAs frequently have one inverting audio input. The properties of inverters will be discussed in Chapter 5. For now you should know that totally inverting a wave has the effect of reversing its voltage polarity at every point. Thus a sawtooth wave that originally looked like that in Figure 3-12a will, if input into an inverting audio input to a VCA, look like Fig. 3-12b when it reaches the audio voltage mixer to the VCA. The aural effect of mixed waves that are out of phase may be changed because of the inversion (remember Experiment #11B). For example, if you were to add a

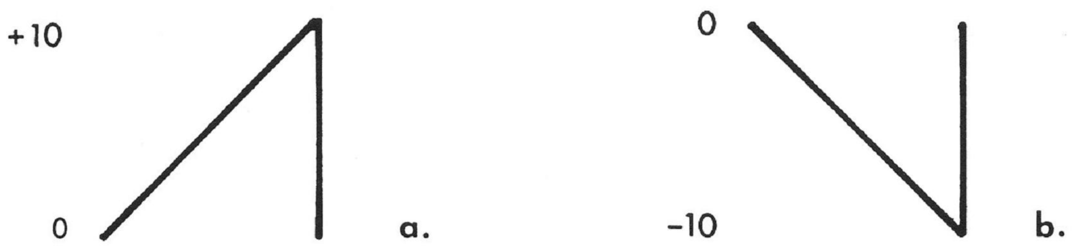
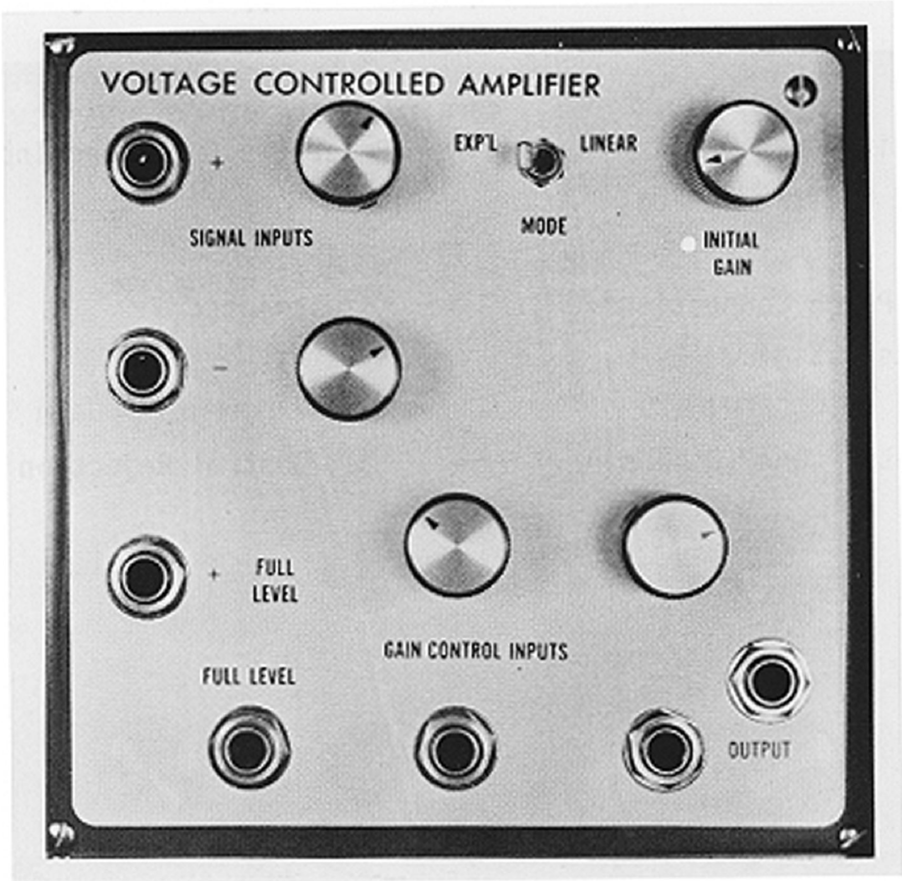


FIG. 3-12



E-mu Module 2000 VCA

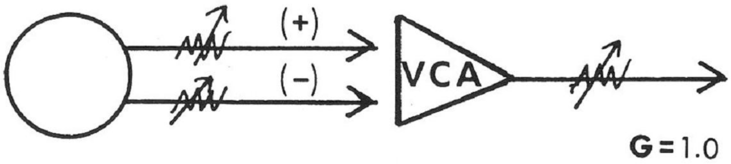
sawtooth to itself you would normally expect the output to be a sawtooth of twice the amplitude (twice as loud—remember diagrams a and b in Experiment #29A).

EXPERIMENT #34: The inverting audio input to a VCA

Needed: multiple

Patch an AF sawtooth into the multiple; come out of the multiple with two patchcords and patch one into each audio input to the VCA. Set the gain control (which generally supplies +10 control volts) all the way to the right and patch the VCA out.

Open the attenuator over the positive audio input. This is the original AF sawtooth. With the experience of the diagram in Experiment #29A, you would think that raising the other audio attenuator would double the amplitude, and thus the volume, at the VCA output. Slowly raise that second audio input attenuator.



The volume decreases rather than increases and, if you're careful, there will be one point at which the sound disappears altogether. At this point the signals cancel each other out.

Just as the oscillator coarse frequency attenuator is a source of +10 control volts for a VCO, just as the filter coarse frequency attenuator is a source of +10 control volts for a filter, so the gain (G) attenuator associated with the VCA is a source of +10 control volts for it. When that attenuator is all the way off, it supplies no control voltage to the VCA; all the way open it supplies +10 control volts. Voltage varies between 0 and +10 depending on the setting of the gain attenuator. You heard the effect of that constant control voltage in the previous experiment. Without it there would have been no positive control voltage appearing in the VCA's CV mixer, and no matter how many sawtooth waves appeared at the audio inputs there would have been no output from the VCA. Any voltage it supplies to the CV mixer will be added to any other control voltage coming into the other control inputs.

B. Two-Quadrant Multiplier

A VCA is sometimes referred to as a two-quadrant multiplier. In the chart shown here, voltage in the VCA's audio mixer is plotted against control voltage.

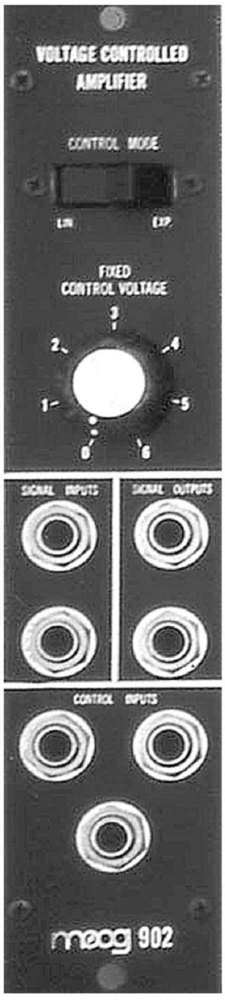
audio (−) control (+)	audio (+) control (+)
audio (−) control (−)	audio (+) control (−)

In the upper right-hand quadrant both audio and control voltage are positive, and the VCA will output. Even if the audio voltage has a net negative value, the VCA will output so long as the control voltage is positive; this is represented by the diagram in the upper left quadrant. No matter what the audio voltage polarity, there is no VCA output if the control voltage is negative (the two lower quadrants). Since the VCA outputs only two of the four quadrants, it is called a two-quadrant multiplier.

As usual, voltages input into control inputs may be LF waves. The type of output from the VCA will depend on the nature of the voltage change of the LF waves in question. With no other control voltage a positive-going LF sawtooth, gradually rising from 0 to +10 volts, will slowly open the VCA to its maximum and then instantly close it, beginning the cycle again. You would hear whatever had been input into the audio mixer to the VCA rise in volume as the LF sawtooth wave, functioning as a control voltage, gained amplitude and fell to 0 volts. If the control wave has a negative component (e.g., a typical sine or triangle wave, which fluctuates from +5 to −5 volts), as soon as that wave hits 0 volts and starts to go negative there will be no output at all from the VCA.

Verify this for yourself by inputting any AF wave into an audio input to the VCA and an LF sine or triangle wave into a control input to the VCA. Raise the attenuators over both inputs into the VCA as well as the attenuator out. The sound rises and falls during the first half of the wave's cycle, but there is no sound at all during the negative half.

However, remember that what determines if there is any VCA output is not that a *component* of a given control voltage is negative, but rather the *total* instantaneous control voltage in the VCA's CV mixer at any given moment. Thus if you were to open the gain attenuator halfway, the CV mixer would start with +5 volts already in it. Then as the sine or triangle went from −5 to +5 volts, the total voltage in the CV mixer would go from a minimum of 0 volts (when

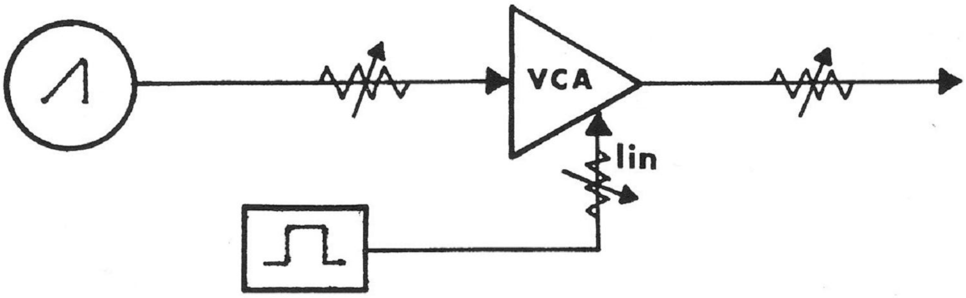


Moog Model 902 VCA

the LF wave was at -5 volts it would be added to the $+5$ already in the CV mixer) to $+10$ volts (when the LF wave was at its maximum of $+5$ volts *that* would be added to the $+5$ volts already in the mixer). Since total control voltage appearing in the VCA's CV mixer would vary from 0 to $+10$ volts, there would be a continuous VCA output except for that instant when the total CV was 0 volts. Opening the gain attenuator just a little more would give you a continuous VCA output. Try it.

Thus the fact that any individual control voltage is 0 or negative does not automatically mean there will be no output from the VCA; it is the *total* net control voltage in the CV mixer to the VCA at any given instant that will determine if the VCA opens at all and, if so, how much.

EXPERIMENT #35: Understanding a VCA



Patch an AF sawtooth into an audio input to the VCA, and an LF square wave (whose frequency should be about 1 Hz.) into the linear control input to the VCA. Open all appropriate attenuators and listen to the output. You will hear the saw-tooth for a given time (about half a second) and then you will hear silence for an equal time, as the LF square wave drops to and remains at 0 volts.

The two waves—the AF sawtooth and the LF square—might be shown as in Figure 3-13a and 3-13b. (Note that these are not to scale, as a might be 500 Hz. while b is only 1 Hz. For this purpose scale is not needed.)

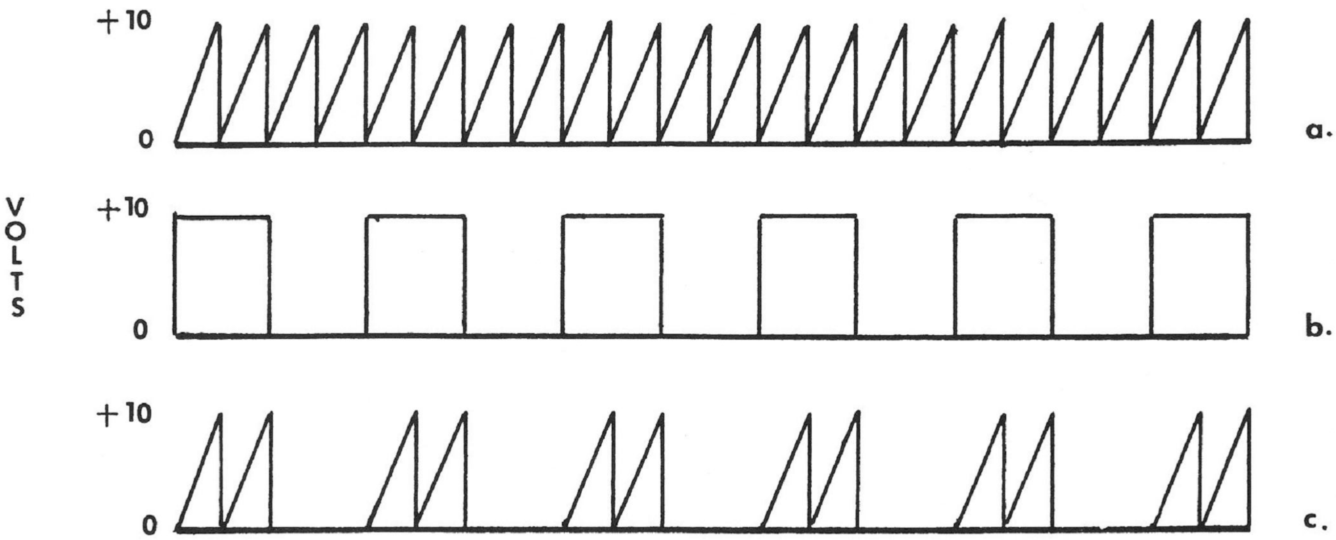


FIG. 3-13

The VCA outputs only when there is positive control voltage in its CV mixer (Figure 3-13c). *This is what is meant by the statement earlier that the primary purpose of the VCA is to allow the amplitude of a control voltage to determine the output amplitude of an audio signal.*

Figure 3-14 is one final example. When the attenuator over the linear control input is halfway up the control voltage is halved (b) and the VCA output (c) is *directly proportional to the control voltage.* Note that even though the input audio voltage remains constant, since the control voltage has been halved the VCA output is halved (when there is any output at all). If the control voltage were input into the exponential CV input rather than the linear, the VCA output would still be dependent upon the amplitude of the control voltage input, but the output would be 10 db/control volt instead of in direct proportion (see part C).

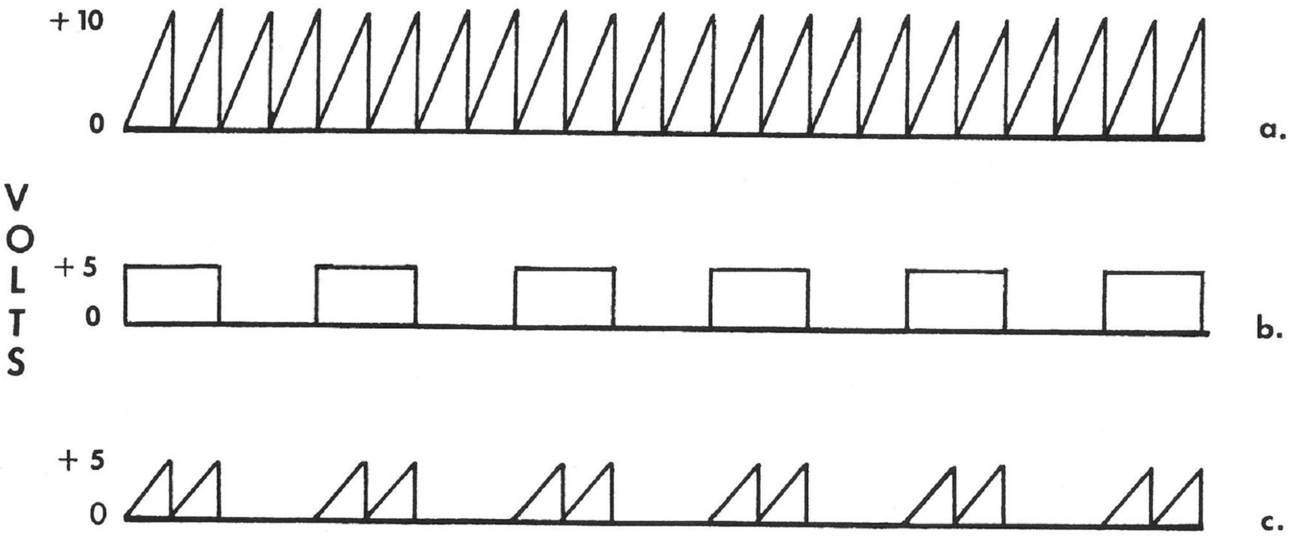


FIG. 3-14

C. Linear and Exponential Control Inputs

Just as the circuitry for the attenuator over one audio input is different from that for the other (because the other inverts the audio signal), the circuitry of VCA control inputs frequently differ from each other because one of the two typically includes an exponential converter. Patchable synthesizer VCAs typically have both linear and exponential control inputs. No matter what the original shape of the CV waveform entering an exponential control input, it will be altered by the exponential converter and appear as something else in the CV mixer to the VCA. An example: If the CV appearing at the exponential control input were originally a positive-going LF sawtooth (Figure 3-15a), it would appear as in Figure 3-15b by the time it reached the CV mixer to the VCA. It has been exponentially converted. As a practical matter this will generally mean that it has been delayed at the beginning and accelerated at the end of its cycle (an exponential curve generally is slow at the beginning and rises rapidly toward the end; see Figure 3-16).

FIG. 3-15

The exponential input is of importance because our perception of change in loudness is exponential rather than linear. Figure 3-16 is a graph of the effect of the linear and exponential control inputs to the VCA. Recall from Chapter 2, section III-C, that when a control voltage is input into the linear control input the VCA opens in direct proportion to that control voltage; when input into the exponential mode, the VCA opens at the rate of 10 db/volt.

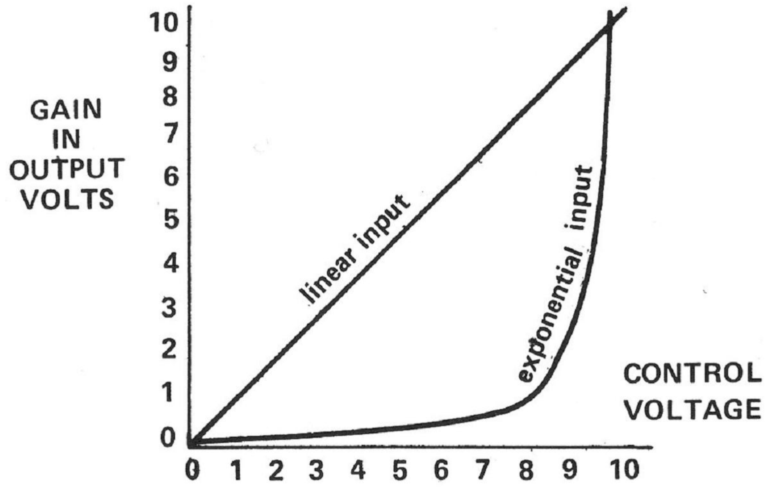


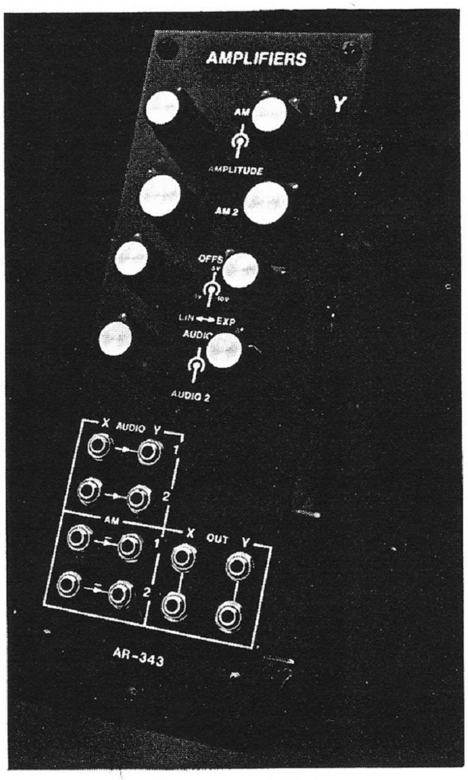
FIG. 3-16

There is a considerable difference in perception of the rate that volume increases or shape changes[26] between these two modes, as Experiment #36 shows.

EXPERIMENT #36: Linear vs. exponential control inputs to a VCA using an LF wave

Input an AF sawtooth into an audio input of the VCA. Use as your control voltage a slow LF sawtooth. First input this LF sawtooth to the linear control input to the VCA, open the attenuators including the one to the mixer out, and listen to how the volume increases.

Now repeat the experiment inputting the LF sawtooth into the exponential control input to the VCA.



Aries AR-343 Dual VCAs

Listening to the difference between these two inputs should help you understand the graph in Figure 3-16. Did you notice how volume increased *evenly* when the LF sawtooth was input into the linear mode and slowly at first and rapidly later when it was input into the exponential mode?

Here's another way to help you understand the differences in these two control inputs. Figure 3-17 is a kind of inverse graph of Figure 3-16, in which change in db is plotted against change in voltage for both the linear and exponential control inputs. The straight line in Figure 3-17 indicates a slope of 10 db/volt, the rate at which control voltages input into the exponential mode of the VCA will open the VCA. The linear input, con-

[26] The idea of the VCAs being used as a module to shape a sound will become clearer after a discussion of envelopes and envelope generators in the next section of this chapter.

versely, opens the VCA *much* faster than the exponential.[27] At +2 control volts into the exponential input the VCA has opened a total of 20% of the VCA's dynamic range;[28] in the same time a control voltage input into the linear input has opened the VCA almost 90% of

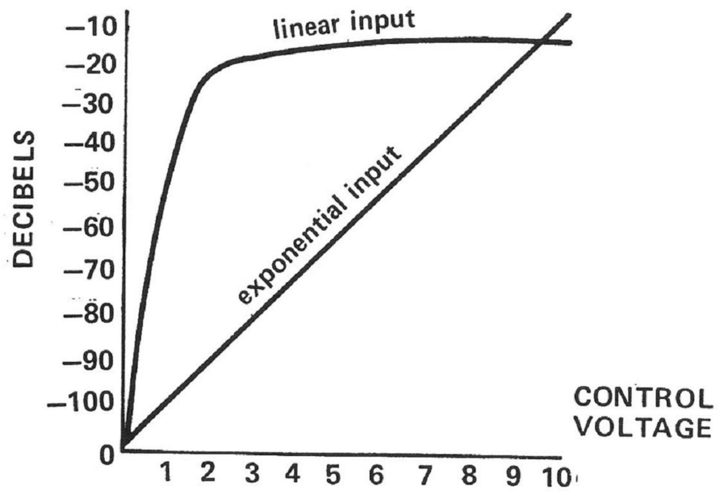
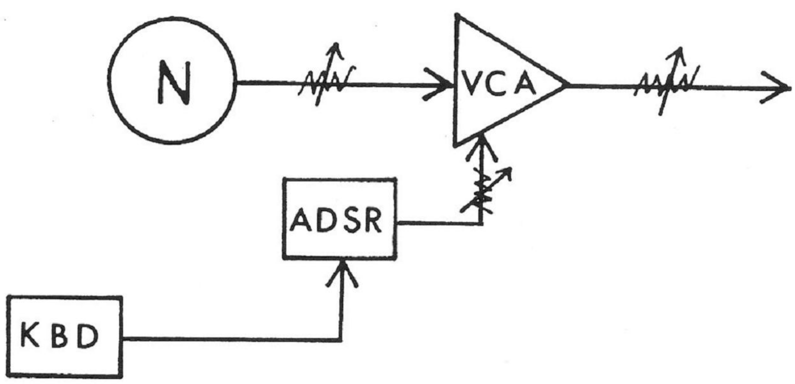


FIG. 3-17

its dynamic range. In other words, the linear input gives a very uneven distribution of the VCA's opening *in terms of decibels—that is, in terms of what we hear.* To obtain a perceptually even distribution of volume change, use the exponential input.[29]

EXPERIMENT #37: Linear vs. exponential control of a VCA using an envelope generator



Initially patch the ADSR output into the linear control input of the VCA and listen with awareness to the nature of the sound as you depress a key. Then, keeping all other parameters exactly the same, close the attenuator of the linear control input, patch the ADSR output to the exponential control input of the VCA, and listen with awareness to the nature of that sound.

Are there any differences? Can you describe them?

[27] If you're wondering why the slope of the exponential input looks linear and of the linear input looks exponential, consider what is being plotted against what. If the exponential input opens the VCA at a *constant* rate of 10 db/volt and you plot db against voltage, you *can't* get anything except a straight line as the slope. The linear input opens the VCA in direct proportion between a given control voltage and an *output voltage.* That was plotted in Figure 3-16. The slope of the linear input looks "exponential" when plotting input voltage against gain (rather than output voltage), which is what Figure 3-17 does.

[28] The *dynamic range* of many VCAs, the range between its minimum and maximum output, is approximately 100 db, about the same as the dynamic range of a symphony orchestra.

[29] Notwithstanding the above, in many applications it would be more musically effective to input an ADSR to a linear rather than an exponential control input to a VCA. Experiment for yourself.

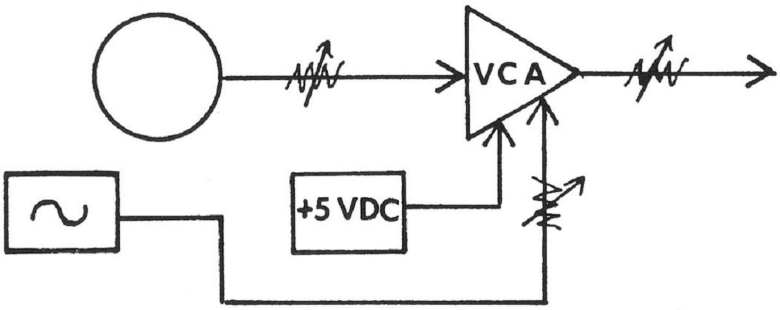
When setting up a patch that uses the VCA, always make an *A–B comparison* between the linear and exponential modes. The sound will generally be perceived as more percussive, louder, and shorter in duration when put through the exponential mode than when input through the linear.

D. Tremolo

Although amplitude modulation is discussed in some depth in Chapter 6, one special type of AM will be demonstrated here. It is commonly known as *tremolo* and consists of controlling the dynamic range of an audio signal with an LF sine or triangle wave. We have already heard this type of control; when applied to a VCO it was called vibrato, when applied to the filter it was called timbral modulation.

EXPERIMENT #38: Tremolo

Needed: +5 VDC



Patch any AF wave into an audio input to the VCA, and an LF sine or triangle wave into the linear control input. Because you don't want periods of silence, offset the control signal by setting the gain attenuator of the VCA halfway open. After opening the appropriate attenuators, experiment with different frequencies as well as different attenuator settings into the VCA for the LF wave.

IV. ENVELOPE GENERATORS (EG)

Aries AR-312 Envelope Generator

How is it that when you hear a tuba play a low C and you hear a piano play a low C you can tell that one note is from a piano and one from a tuba? After all, they are both playing the same note.

One reason is that the different shapes of the piano and tuba encourage different resonant frequencies and thus accentuation of different harmonics. Another, perhaps more important, reason is the difference in "shape" of the sound. Your ears receive musical "cues" about the nature of a sound. For example, a pitch from a tuba takes about one-fifth of a second to reach full amplitude, while a pitch from a piano takes only a millisecond or so to reach full amplitude. The initial portion of a musical event—the time it takes to reach maximum amplitude—is called the *attack* stage. Even a trained musician might be hard-pressed to tell the difference between a tuba and a piano playing the same note if he or she could not hear the attack of the note. The first two milliseconds are crucial in identification of a sound.

Envelope generators (sometimes called *contour generators* or *transient generators*) typically are of two types: ADSR and AR.[30] They provide the ability to give shape to any voltage-controlled parameter. A graphic representation of the shape of the parameter is called its *envelope;* since the ADSR and the AR create envelopes, they are called *envelope generators.* The shape of the envelopes created by the envelope generators is determined by the position of the attenuators on the ADSR and the AR. The envelope generators fire when they receive a *gate* signal from any of a number of devices on a synthesizer (see section III of Appendix A), the most common being the keyboard. Although they should be used as any voltage is used, the most typical use of an envelope generator is as a control voltage to a VCF or a VCA.

A. Envelopes

An LF square wave as a control voltage to a VCA causes an output from the VCA that is instantaneously high and then instantaneously low (Experiment #35). As was said in section III of this chapter, the instantaneous control voltage amplitude determined the instantaneous output amplitude of the audio signal input into the VCA. A graph of the output amplitude (in volts) vs. time for Experiment #35 would look like Figure 3-18.

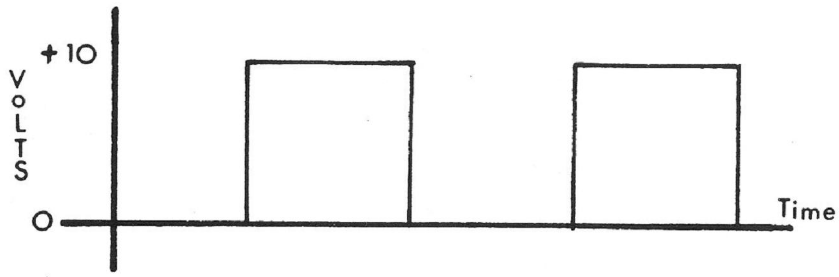


FIG. 3-18

A graph of the way a parameter varies over time (the "time domain") is a visual representation of its envelope. The horizontal axis on such a graph always represents time; if we speak of an amplitude envelope we generally mean the way VCA output varies over time. Figure 3-17 is an amplitude envelope where the control voltage into the VCA is an LF square wave. A filter envelope and a pitch envelope show respectively how the F_C and frequency vary over time. The envelope will always be represented diagrammatically as the total of the instantaneous control voltages to the parameter. If the total of the instantaneous control voltages is simply an LF wave, as in Figure 3-18, then the envelope for that parameter will have the same shape as the LF wave.

Figure 2-7 in Chapter 2 showed the effect of an LF sine wave controlling a VCO; since it is a graph of how a parameter (frequency) varies over time, we can show its pitch envelope (Figure 3-19).[31] Individual envelopes have discrete beginnings and endings;[32] in order to show a beginning and an ending, choose the lowest voltage as the point to start. Figure 3-19 gives the same information as Figure 2-7; however, it starts at the lowest voltage (output in frequency) rather than the point of 0° phase.

[30] However, there are several other configurations. Many Moog products and the Oberheim SEM provide only one attenuator for both decay and release time. The Korg MS-20 and E-mu Module 2350 offer initial delay times before the attack stage (effectively five-stage envelope generators). E-mu, Aries, and Serge-Modular offer voltage control of the timing and sustain parameters on their four-stage envelope generators.

[31] Technically these pitch envelopes would be correct only for linear (rather than exponential) control inputs; this liberty is taken for the purpose of teaching pitch envelopes.

[32] Exceptions: when playing in the "single" triggering mode (see part F of this section); when retriggering occurs before the last stage of a prior envelope is completed (see part E of this section).

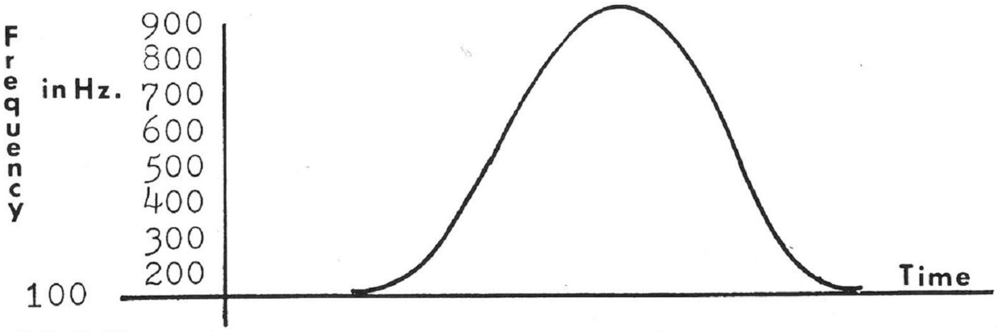


FIG. 3-19

Figure 3-20 is an example of a filter envelope whose initial F_C is 10 Hz. and whose control voltage is a +10 volt LF sawtooth wave.

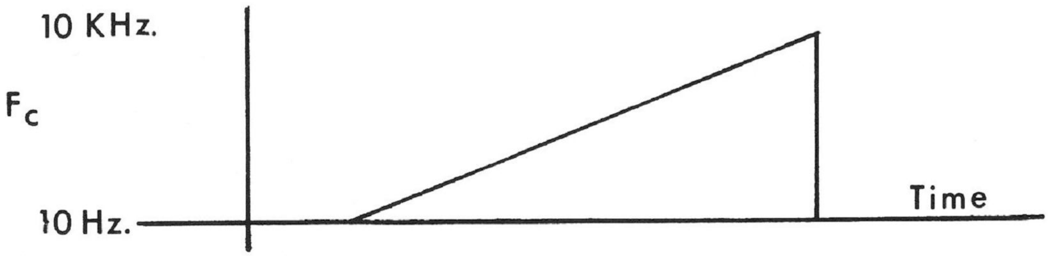
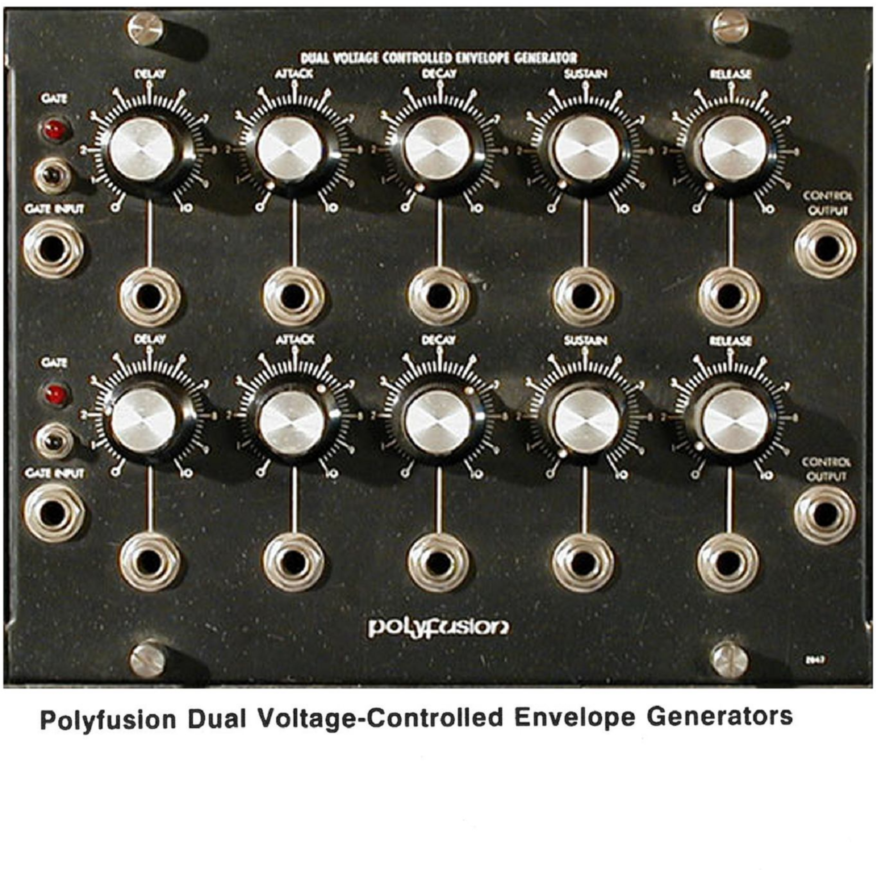


FIG. 3-20

As shown in Figure 3-18, the envelope of a parameter controlled only by an LF wave will look like that LF wave itself.

B. The AR Envelope Generator

Although LF waves help create certain effects, oscillator-produced waves are limited in their shapes and therefore in their effect as control voltages. No matter what its frequency or amplitude, a square wave is still going to be instantaneously high and low. How much more interesting sounds would be if we could further vary the shapes of the voltages we use to control voltage-controlled devices!



Polyfusion Dual Voltage-Controlled Envelope Generators

That, of course, is exactly what envelope generators do. Although the most common *uses* of envelope generators are to control a VCF and/or a VCA, their most important aspect is the *life* they impart to events, hopefully making them interesting and realistic.

Figure 3-21a shows a typical envelope created by an LF square wave. In a given application the sound might be more interesting if we could vary the rise and fall times to those shown in Figure 3-21b. That envelope indicates that it takes some time for the amplitude to reach its maximum voltage and some time for it to fall to zero, instead of its doing both instantaneously. The input audio signal *gradually* rises and falls in volume. This demonstrates the primary characteristic of the AR generator. It serves as a source of control voltage with manually variable individual parameters—that is, the rise and fall times (*A*ttack and *R*elease times) can be varied by the attenuators on an AR generator.

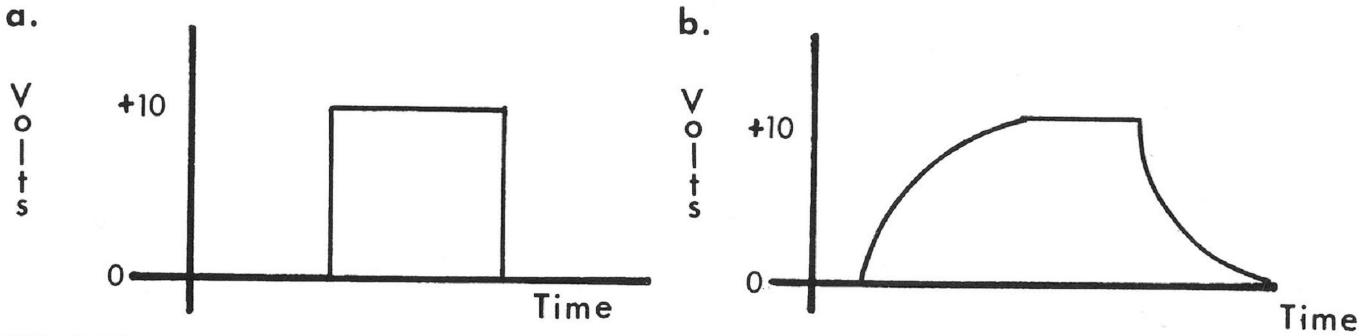
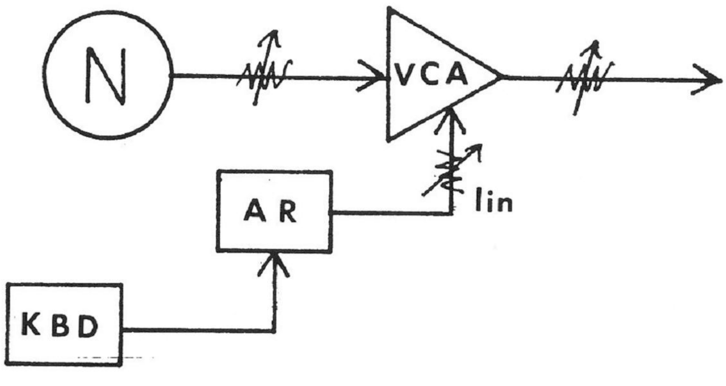


FIG. 3-21

EXPERIMENT #39: Introduction to envelope generators as control voltages

Set up the patch shown here. Make sure both AR attenuators are all the way down (or to the left): setting 0,0—see page 5.



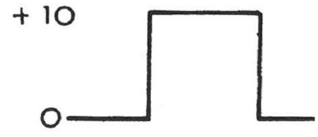
Depress a key.

As you will learn later in this section, the keyboard can supply a *gate signal* to all envelope generators. Once you raise the control attenuator into the VCA that has the AR patched into it, all the AR needs to begin functioning is to receive a gate. You signal it with a gate when you depress a key.

On both the AR and ADSR the attenuators are in minimum position when they are all the way down, maximum when all the way up, and at various midpoints in between.[33] Since the AR attenuators were at setting 0,0 in the preceding experiment, both attack and

[33] If you are using a synthesizer with rotary (round) attenuators, minimum is all the way to the left, maximum all the way to the right.

release time characteristics were minimum—that is, the *time* they took to complete was the absolute minimum of which the AR is capable. A minimum attack time is essentially instantaneous. As soon as the gate signal is received, the parameter being controlled by the AR (in this case amplitude) takes the minimum time to reach the voltage allowed by the control voltage attenuator.



With a minimum release time, as in Experiment #39, as soon as the AR fails to receive a gate signal (when you stop depressing a key), the parameter being controlled by the AR instantaneously returns to its original position.

Redo the previous experiment with the following difference: Set the attack attenuator to its maximum position (setting 4,0).

Depress a key.

The time the control voltage now takes to reach its maximum is the longest it can be with the AR generator. Since the release time attenuator is at a minimum, as soon as you let go of the key the control voltage instantaneously goes low; with no control voltage the VCA shuts down and the audio signal (in this case, noise) that is always present in the audio mixer of the VCA has no way to be output.

Redo the previous experiment with the following difference: Put the attack attenuator at minimum and open the release attenuator all the way to maximum (setting 0,4).

Depress a key.

Since the attack attenuator is at a minimum, attack time will be instantaneous; since the release attenuator is at a maximum, the release stage will take the longest it can, supplying slowly diminishing control voltage to the VCA. The sound slowly dies away.

Redo the previous experiment with the following difference: Open both attack and release attenuators to their maximum positions (setting 4,4).

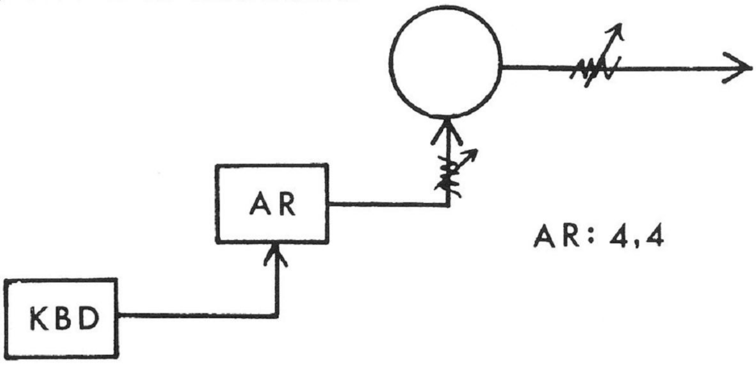
Depress a key (which you should hold for a short while to get the effect).

The horizontal line in the envelopes indicates the levels at which the control voltage sustains after it has reached its maximum. On typical ARs voltage always goes to +10 volts. On voltage-controlled parameters +10 volts is virtually always enough to sweep the entire spectrum of the parameter.[34] If you want the AR generator to sustain at a voltage less than +10 volts you must attenuate it via a control voltage attenuator. (Experiment #40).

[34] +10 control volts will raise an exponential VCO from 20 Hz. to 20 KHz.; it will raise an F_c from 10 Hz. to 10 KHz; it will open a VCA from −100 db to unity gain; it will sweep a voltage-controllable pulse width from one end of its duty cycle to the other.

EXPERIMENT #40: Sustain attenuator for an AR envelope generator

Set up the synthesizer as shown here:



Open the control attenuator into the VCO all the way.

Depress a key.

In this configuration the AR generator will very slowly output a control voltage that will raise the pitch of the VCO. Since the attenuator is wide open, the frequency of the VCO will go all the way up and out of the range of hearing. The AR generator always goes to +10 volts. By lowering the CV attenuator into the VCO, you can keep the same attack and release settings but have a maximum effective voltage of anything you want.

Set the CV attenuator such that the pitch of the VCO will rise one octave, two octaves, and four octaves when a key is depressed. When you have done this, the AR voltage that controls the VCO will be reaching a maximum of +1, +2, and +4 volts respectively.

By now you can see how the AR is replacing the LF waves we have been using for control voltages up to this point. An envelope generator creates a control voltage with certain parameters that are manually variable. On an AR those parameters are the attack and release portions, which create an envelope that has many more musically interesting possibilities than those created by periodic LF waves.

C. Timing Signal #1: The Gate

The AR (and the ADSR) must have some kind of signal indicating when to begin controlling a parameter and when to stop controlling it. Since this signal indicates a point in time to begin and another point in time to stop, it is called a *timing signal.* The two types of timing signals envelope generators use are called *gates* and *triggers.*

A gate is simply an on–off signal. In order to start, an envelope generator must receive a gate signal, in the same way that an automobile engine must receive a signal from the ignition. Conversely, unless a gate signal is received by an envelope generator, it either will not start or, if it has already started, will shut down at a rate determined by the release setting.

Figure 3-22 shows a typical AR envelope and associated gate signal.

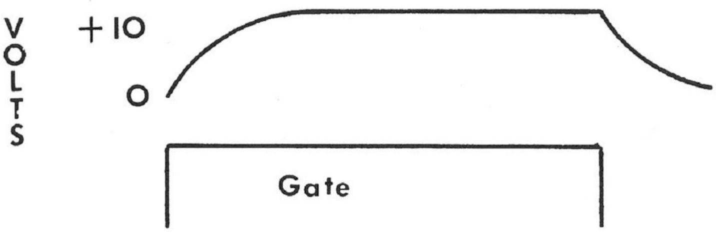
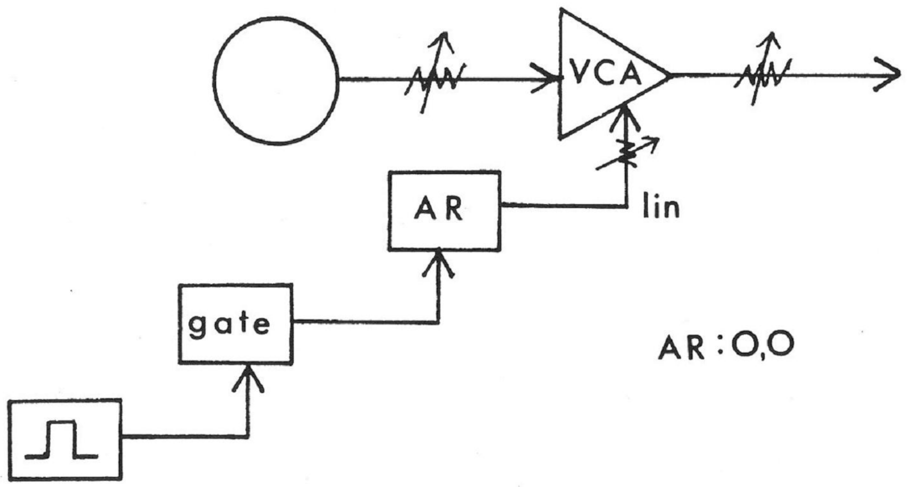


FIG. 3-22

The AR starts as soon as it receives a gate signal (e.g., from a keyboard), opens at a rate determined by the attack attenuator to a level of +10 volts,[35] and begins to close at a rate determined by the setting of the release attenuator as soon as it no longer receives a gate signal.

Since a gate signal is an on–off timing signal, a gate can come from an LF pulse. An LF pulse is either high or low, so it can signal an envelope generator to start when it is high and stop when it is low.

EXPERIMENT #41: LF pulse gating envelope generator

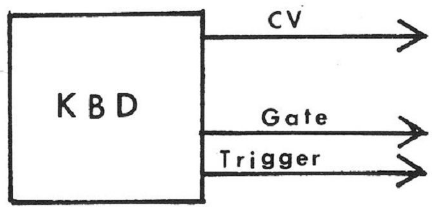


Patch an LF pulse into the gate input of an envelope generator. Patch any AF signal or noise into an audio input to the VCA. Set both AR attenuators at minimum and open the other attenuators.

The AR will go on and off at a rate determined by the frequency attenuator of the LFO. If that frequency is low enough, you can raise the attack and release to any point and hear the AR generator start and stop opening the VCA.

Any signal that has a fast rising edge, a sustained duration, and a falling edge can be a gate. Thus an LF pulse wave can be a gate but an LF sine wave, with no sharp edges, cannot function as a gate.

The most common source of gate signals is directly from the keyboard. Every time a key is depressed, the keyboard outputs at least one control voltage (which is frequently pre-patched to each VCO and to the VCF) and two timing signals, a gate and a trigger[36] (the gate being typically pre-patched to both envelope generators, the trigger to the ADSR only), all of which may be available at external jacks. Whether these keyboard outputs are all used depends on your need, but they are available each time a key is depressed.



Another source of gates on many synthesizers is a "Manual" button, located near the envelope

[35] Moog and Korg products have negative-going gates; there is a 10-volt output when no key is pressed down, 0-volt output when a key is pressed and a shorting-trigger ("*s-trigger*") is activated, shorting the normally high gate to ground.

[36] Moog keyboards do not output triggers because Moog envelope generators (even ADSR) don't use triggers. The discussion in part E of this section about how an ADSR must receive a trigger to function as a four-stage envelope generator does not apply to Moog products.

generators. Depressing this button provides a gate to the envelope generators that starts when the button is depressed and stops when it is releasad.

D. The ADSR Envelope Generator

An *ADSR* (ADSR stands for *Attack, Decay, Sustain,* and *Release*) is a more complex envelope generator than an AR, because it has four variable parameters instead of two. The *A*ttack and *R*elease parameters of the ADSR function exactly as they do on the AR. The *D*ecay parameter controls the timing of a voltage fall after the attack to a variable sustain level. The attack, decay, and release are *timing,* not voltage, parameters. *S*ustain sets the voltage level at which the ADSR will output a voltage once the attack and decay stages have completed, for as long as a gate is received; it is variable up to +10 volts.

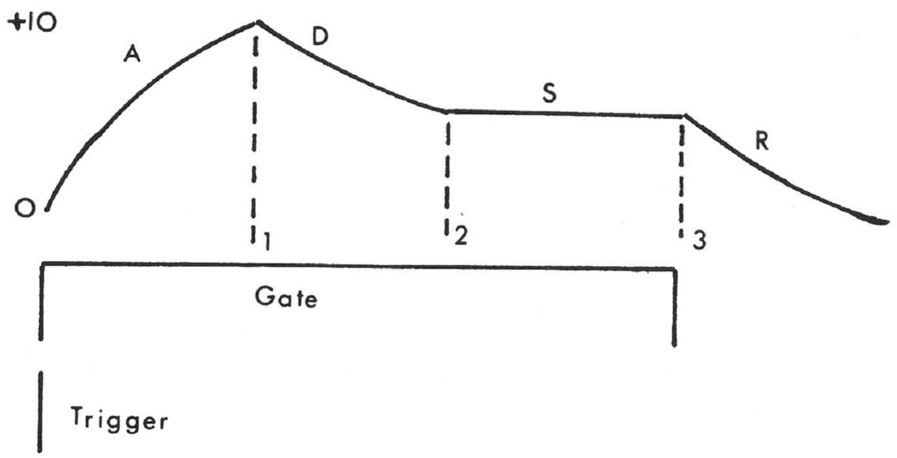


FIG. 3-23

Figure 3-23 shows a typical ADSR envelope and its relationship to a gate and a trigger. The gate has the same function with an ADSR as it does with an AR: it tells the envelope generator when to start and stop. The ADSR will begin firing the instant it receives a gate signal and will stop when the gate is no longer received. The ADSR will instantly go to its release stage at whatever point in the cycle the gate stops.

As with the AR, in the attack stage the voltage always goes from 0 to +10 volts, the position of the attack slider determining only the *time* it takes to make that journey. Voltage decay will take any time from instantaneous to several seconds, depending on the position of the decay attenuator. The decay will be to a voltage determined by the position of the sustain attenuator. That attenuator does *not* determine how long the sustain lasts—it is not a timing attenuator, as are all the other attenuators on an ADSR. How long the ADSR "fires" is determined by how long the gate lasts (e.g., how long a key is pressed beyond point 2 in Figure 3-23). If the gate should stop at point 1 in Figure 3-23, the ADSR would bypass both the decay and sustain stages and go directly to release. The envelope would then be like that in Figure 3-24a rather than that in Figure 3-23.

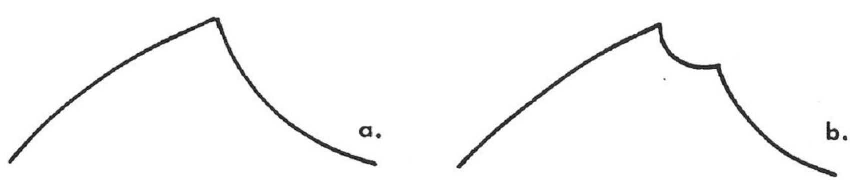
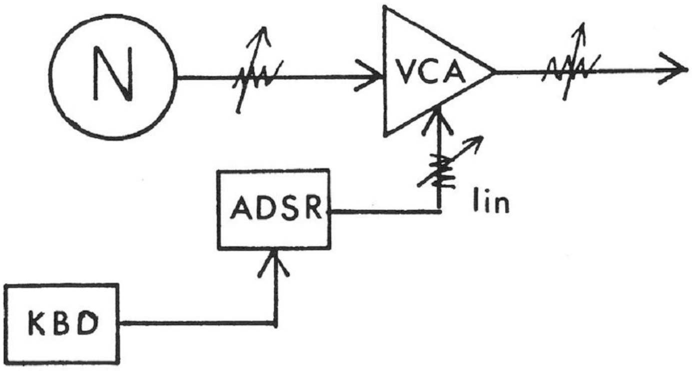


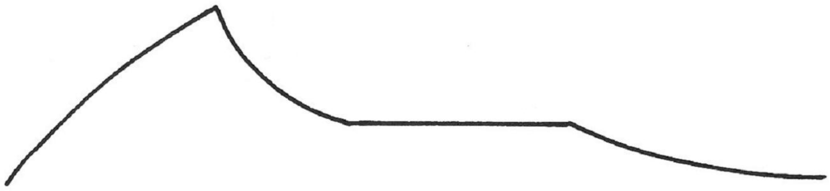
FIG. 3-24

If the gate should stop at point 2, then the envelope would look like Figure 3-24b. It is only if the gate stops at point 3 that the envelope will look and sound like that in Figure 3-23.

EXPERIMENT #42: The effect of ADSR attenuator settings

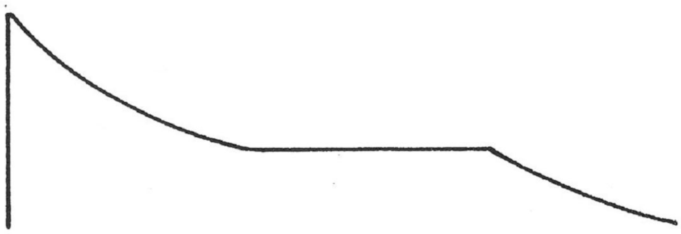


Set up the patch shown here. Initially, set the ADSR to 4,4,2,4. Depress a key. The amplitude envelope sounds like this:

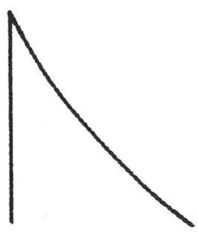


Since the parameter being controlled by the ADSR is amplitude, the volume will take maximum time to reach +10 volts; it will then take maximum time to decay to the sustain level, which is about +5 volts. When you release the key, the ADSR will take maximum time to go from the level sustained to 0 volts.

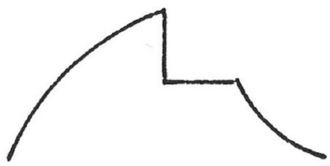
Now set the attack attenuator to minimum, the total setting being 0,4,2,4. Depress a key and hear the effect of the envelope below.



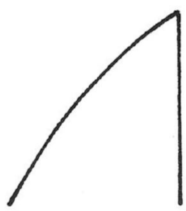
Hit a key and instantly let it go, as though you were playing staccato. Although the attack stage will complete since it is virtually instantaneous, no other stage will get a chance to finish and the effect will be to go immediately to the release stage. This is an example of the ADSR firing until the gate signal stops, at which point it goes directly to its release stage.



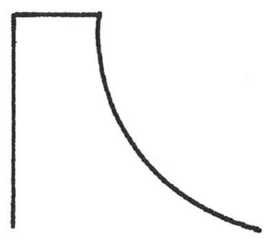
Set the attenuators to setting 4,0,2,4. Depressing a key (or pushing the Manual button) supplies a gate. The ADSR takes maximum attack time to get to +10 volts, but since the decay attenuator is at 0 time, there is no decay. As soon as +10 volts is reached, the ADSR goes from the attack to the sustain portion of its cycle.



Try slider setting 4,0,0,4. Now the sustain attenuator is in its minimum position, a sustain of 0 volts. As soon as the attack portion is completed, the ADSR output drops to 0 and there is no voltage to release from. The effect is the same as if the release attenuator were all the way down. If sustain is 0, release will have no effect



unless *the ADSR stops receiving a gate before the sustain stage takes effect. In that case release will immediately take effect (see Figure 3-22).*



Try attenuator setting 0,0,4,4. Since attack is at minimum, it will be instantaneous, but since sustain is at maximum, there can be no voltage decay. The voltage sustain is at maximum and voltage cannot decay to maximum. Any time sustain is maximum, the decay attenuator has no effect.

Keeping the attenuators set at 0,0,4,4, play a series of notes rapidly. Note that you are beginning a new envelope before the old one has completed (because of the long release setting). This is called re-triggering *and is one of the cases in which each individual note does not have its own envelope.*

Finally, try an ADSR setting of 0,0,0,0 into the exponential control input to the VCA, for a clicking sound.

This has been just an introduction to the possibilities of an ADSR envelope generator. Just keeping the different attenuators of the ADSR on either 0, 1, 2, 3, or 4 (i.e., not including intermediate points like 1½), there are 625 different settings possible (5^4). Take some time now to experiment with other settings. When you hear a sound, draw the envelope created and describe to yourself what is happening.

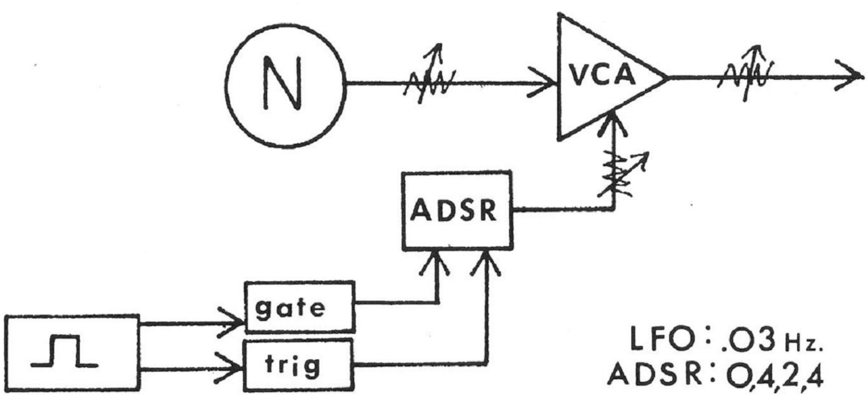
E. Timing Signal #2: The Trigger

A trigger is any signal used to start the operation of some device. The most common use of a trigger is to start the attack cycle of an ADSR.[37] The ADSR *will* function if there is no trigger, but it will function as a three-stage envelope generator rather than a four-stage envelope generator; in that case, the setting of the decay attenuator will serve as though it were an attack setting. Since sustain is variable, the "attack" won't always go to +10 volts. It will go to whatever level the sustain attenuator is set at within a time period set by the position of the decay attenuator. Thus with no trigger the ADSR would function as an AR with variable sustain.

EXPERIMENT #43: ADSR as three-stage envelope generator

Needed: multiple

Set up the synthesizer as shown.



[37] Examples of other uses of a trigger: to ring a VCF (Experiment #33), to start and stop a sequencer, and to determine oscillator synch/reset functions.

The LF square wave should be very LF.

Patch an LF square wave into the multiple; bring one patchcord out of the multiple and into the gate input and another into the trigger input to the ADSR. The effect of this is to provide both a gate and a trigger to the ADSR from the LF square wave. The envelope looks like this:

Now remove the patchcord between the multiple output and the trigger input. The ADSR now receives only a gate signal from the LF square wave. Listen again.

The entire attack stage has been bypassed. The setting of the decay attenuator becomes the length of time the ADSR will "attack" to reach a maximum voltage as determined by the sustain attenuator setting. The ADSR settings are 0,4,2,4. When the ADSR received a trigger, it began an instantaneous attack, a slow decay to a voltage level of about 5 volts, and then slow release. Without the trigger, the decay setting (which is at 4) becomes the time for the attack. Thus there is a very slow attack, which goes not to +10 volts but only to the level at which the sustain attenuator is set. The envelope now looks like this:

Without a trigger the ADSR functions as an AR with variable sustain. Instead of always going to +10 volts, it can vary between 0 and +10 volts.

Any signal with an extremely fast (essentially instantaneous) rise time can function as a trigger. If you looked at a series of triggers on a superb oscilloscope they would look like Figure 3-25.

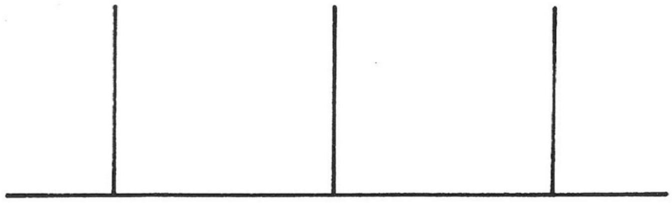
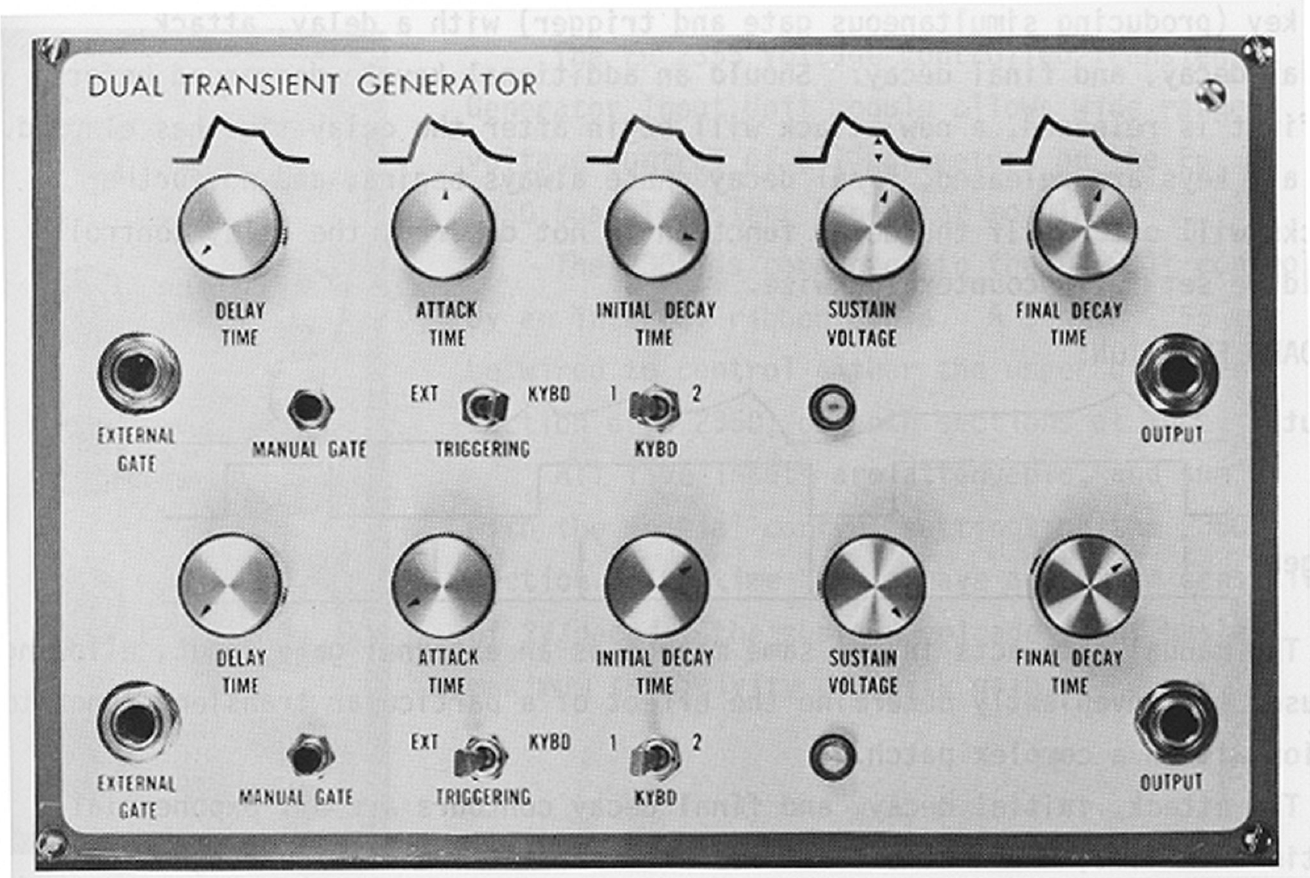


FIG. 3-25

The critical elements are that the rise time be essentially instantaneous and that the signal reach a particular voltage threshold (generally +10 volts). If a pulse wave serves as a trigger, everything that the pulse does after the initial rise is irrelevant. The duration of the pulse doesn't matter. The internal circuitry of the ADSR needs only the rise time; if this rise time is fast enough and high enough, the ADSR will allow the signal to be a trigger and ignore everything that happens to the signal later. Thus a pulse wave can be both a trigger and a gate. How long the gate lasts will be determined by how long the pulse is high.

As with a gate, the most common source of triggers is the keyboard. There is almost always an internal patch of a trigger from the keyboard to the ADSR; every time a key is depressed, the ADSR receives a trigger. This may or may not be used (it is obviously not used, for example, if the ADSR is not in the patch), but it is always available. Since the requirements of a trigger are an instantaneous rise time to some positive voltage, any LF pulse can serve as a trigger. A Manual button also typically has a trigger hard-wired to the ADSR. Pushing it gives a trigger to the ADSR and, for as long as you hold it, a gate to both envelope generators (see Experiment #A-3 in Appendix A).



E-MU Module 2350—Dual Delayed Transient Generator

F. Single and Multiple Triggering

Envelope generators receive triggers from keyboards in one of two ways: by either single or multiple triggering. By far the more common is multiple triggering, wherein individual timing signals are supplied to the envelope generator every time a note is pressed, even if other notes are being pressed and it's a *one-voice synthesizer*.

Remember, from footnote 36, that Moog envelope generators don't need to receive triggers in order to fire and behave like 4-stage envelope generators. Since they do not receive triggers, when a keyboardist plays most Moog products its envelope generators will "see" only one long gate and not refire *unless* the keyboardist takes her or his finger completely off the key for a time, no matter how short, between *every* note. The fingers need to learn a different style of playing on a Minimoog than on an Oberheim OB-1 or an ARP Solus.

Some synthesizers, like those in the KORG PS- series or the ARP 2600, have a switch that allows the user to select either single *or* multiple triggering.

EXPERIMENT #44: Types of triggering

Set up the same patch as in Experiment 42, but use an AF wave as your audio signal. If your synthesizer has only one voice, hold down one key and play other notes simultaneously. If you hear individual envelopes, the keyboard is multiple triggering; if you hear only the original envelope, it is single triggering.

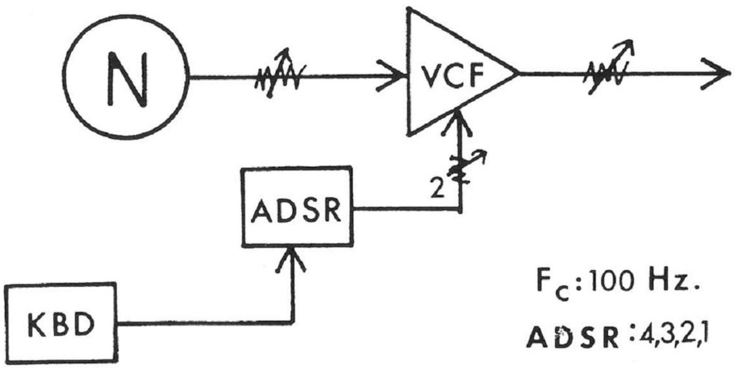
If your keyboard offers single triggering, note that the musical effect of single triggering in this experiment is similar to legato—one obvious attack and a subsequent series of notes perceived to be within the "umbrella" of the first note. (See Section IV in Appendix A.)

G. Exercises and Applications

Before we proceed to exercises and applications, some basic review may be helpful. Envelope generators typically function as any control voltage does. They can, by voltage control, raise[38] the frequency of VCOs, raise the F_C of a VCF, open a VCA, sweep a pulse width, or control any other voltage-controllable parameter at rates determined by the timing attenuators and to a voltage level determined by the sustain attenuator and the attenuator into the module being controlled. The importance of envelope generators to synthesis cannot be overemphasized. Because of the great control flexibility they provide, many synthesists say they would like to have as many envelope generators as all their other modules combined.

EXPERIMENT #45: Envelope generator review

Set up the patch as shown.



Press a key and, as you listen to the effect, verbally describe exactly what is happening.

Here is a narrative description of what happens when a key is pressed, but don't read this until after you have tried to give your own description.

When the key is pressed both timing signals, a gate and a trigger, are instantaneously sent by the KBD to the ADSR. The gate tells the ADSR to begin operating, the trigger tells it to function as a 4-stage envelope generator.

The only module being controlled by the ADSR is the VCF. Since the level of the control attenuator into the VCF is 2, both the voltage to which the attack travels and the sustain stage will be attenuated. Instead of going to +10 volts, the attack will go only to about +5 volts. Since the initial F_C is about 100 Hz., the filter will slowly (attack setting is 4) open to about 3200 Hz. (footnote 3 in this chapter). During that stage, noise of higher and higher frequency will be heard. As soon as the attack is completed the filter will begin to close at a rate set by the decay attenuator (level 3—a slow decay) to the voltage level set by the sustain attenuator and the control attenuator. The sustain attenuator is at level 2 (about +5 volts), the control attenuator is at level 2 (allowing about half the control voltage into the VCF), so the filter will remain open with about 2 or 3 control volts getting to it (F_C of about 400–800 Hz.) until the key is released. At that point the F_C returns to its original starting point (100 Hz.) at a rate determined by the release attenuator (level 1—a fast release).

There are several parameters that will affect the output of a module controlled by an envelope generator:

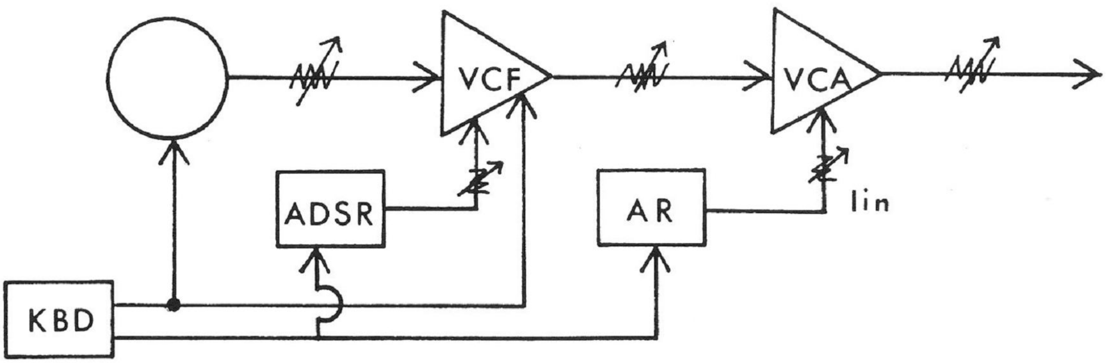
[38] Envelope generators can also lower these parameters if inverted; see Chapter 5, section II-C.

1. The envelope generator timing and level settings.
2. The level of the control attenuator into the module.
3. The initial setting (e.g., initial F_C or initial gain) of the module. (See Section V in Appendix A.)

While you have this particular patch set up on the synthesizer, take note of the changes in sound that occur as you vary the ADSR parameters. With the ADSR at 1,2,3,2, input a pulse or sawtooth wave into an audio input to the filter and vary both the initial F_C control and the control attenuator into the filter. You will get different "brass" instrument sounds, depending on the initial CV setting of the VCO. A 100 Hz. pulse with an ADSR setting of 0,1,2,3 will give a primitive electric bass. Try an AF sine wave, an ADSR setting of 1½,2,4,2½, a low control attenuator level, and a little timbral modulation for a flute effect.[39]

Experiments 46 to 54 are generalized, with a basic description of what occurs and the expectation that you will spend much time becoming acquainted with the various possibilities within each experiment.

EXPERIMENT #46: Control of several modules by envelope generators



Set up the typical live rock performance patch diagrammed above.

In this patch the filter and VCA envelopes can be different. Begin your experimentation by setting the AR at 00 and note how abrupt the sound is. To soften the sound a bit open the AR attenuators to 11 (Chapter 1, paragraph 15).

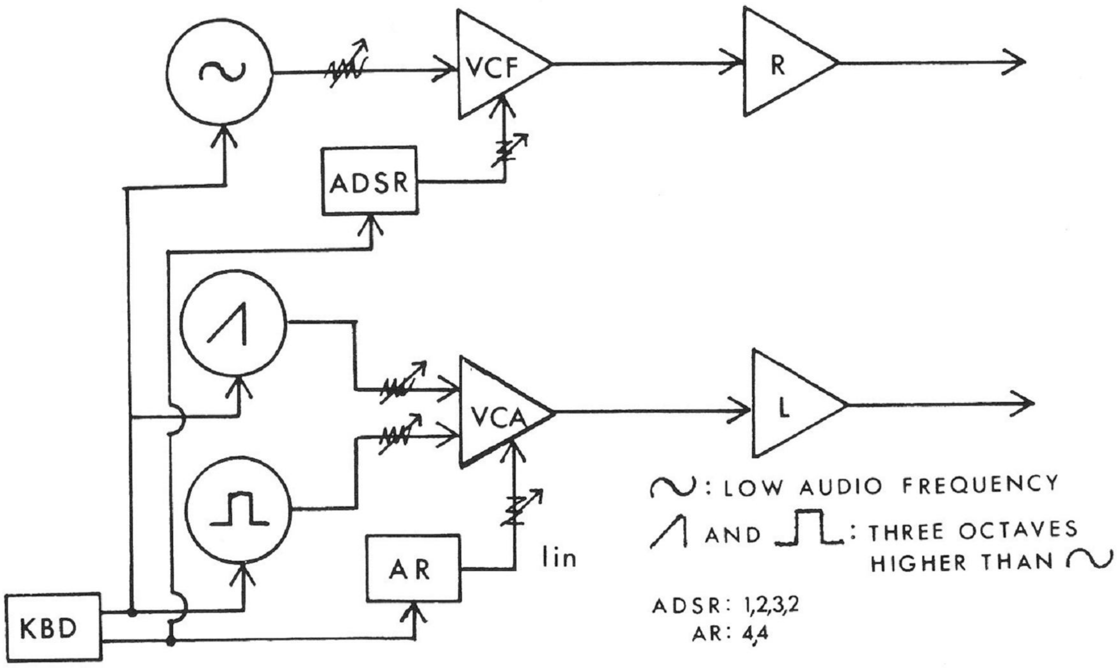
Begin to listen to sounds differently. Ask yourself what happens to the pitch, the timbre and the volume of a given sound—an acoustic instrument, a natural sound from your environment, a sound you make. How would you set the envelope generators and what parameters would they control in order to reproduce those sounds?

EXPERIMENT #47: Two signal paths using different envelopes

Envelope generators allow you to input one signal into a VCF, another into a VCA, and to hear both as distinctly separate. The patch below is an example of a "bass and strings" patch in which, because the attack of the ADSR is fast and of the AR very slow, the signal input into the VCF will be heard and finished before the

[39] *The Source Book of Patching and Programming from Polyphony* is available. See (11) in Bibliography.

"strings" input into the VCA is even heard. If you play fast, all you will hear is the "bass"; if you hold a key down, the "strings" will come in.



Note that in this patch the "bass" comes out the right side and the "strings" the left. They could just as easily be mixed instead of having this binaural effect.

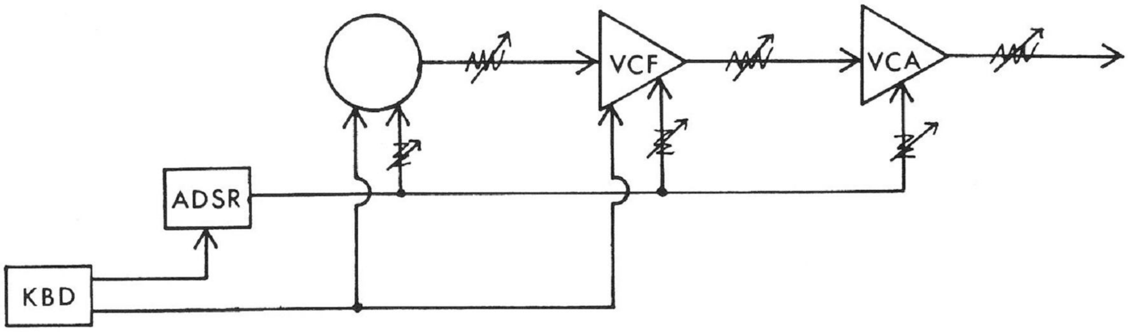
EXPERIMENT #48: Pitch envelopes

Most acoustic instruments have some frequency change within each note. For example, a string goes slightly sharp immediately after it is plucked and then returns to its original frequency. It is the simulation of just these subtleties which creates realism in synthesis.

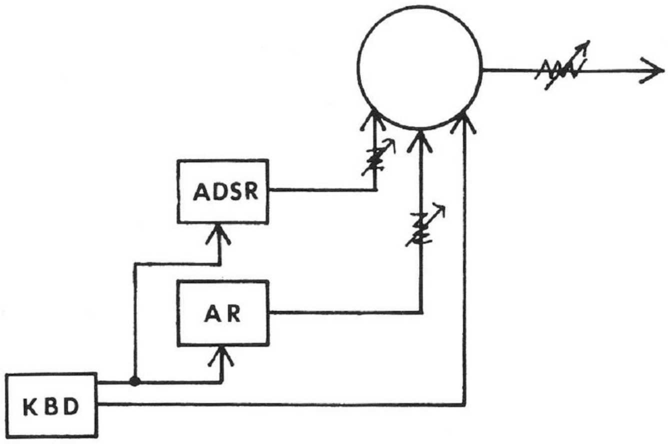
The settings of the A, D, and R attenuators will determine the time *the VCO takes to make the pitch changes; the amount of those changes is determined by the level of the control attenuator into the VCO. The pitch that the VCO holds until released is determined by the sustain attenuator.*

In the patch shown here, pitch, timbre, and amplitude are all controlled by the ADSR. Experiment with many different settings and conceptualize to yourself what's going on.

Remember that you could open the control attenuator into the VCO to a level of ¼ or ½, into the VCF to a level of 3, and into the VCA to 4. Thus the same envelope will create different control effects on different modules.



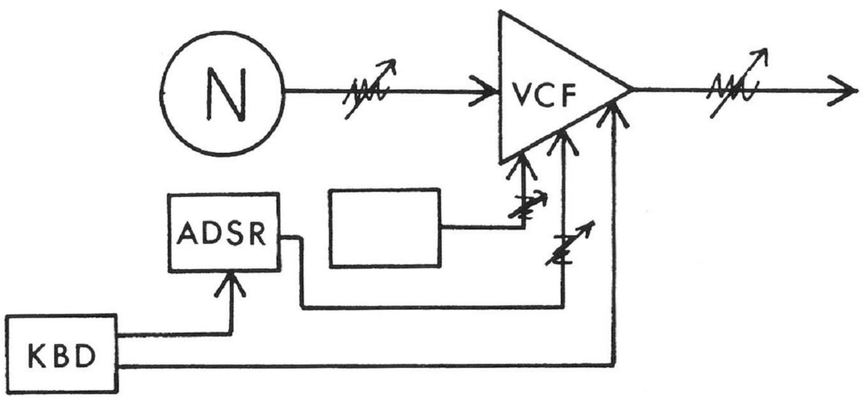
EXPERIMENT #49: *Complex envelopes using more than one envelope generator per voltage-controlled module*



The two envelope generators are controlling a VCO *in the above patch only as an example, because we perceive changes in pitch more easily than timbral or amplitude changes. The effect of having both attack attenuators at different settings is* not *to change the total time it takes for the attack stage to take place. What the two attacks do is create the possibility of an attack time to a given voltage and then a drop, but then another attack as the second envelope generator continues with its attack. This technique is discussed in some detail in Chapter 4, section A, because it gives you the possibility of an envelope more complex than that of the ADSR. You can have an attack, a decay, another attack, decay, sustain, and release. If you have two or more ADSR envelope generators, the resultant envelope will be even more complex. The levels of the control attenuators will be critical here. Experiment with different envelope generator attenuator settings as well as different control attenuator levels; then go on to controlling other modules with two envelope generators. Remember that the outputs of those envelope generators are available elsewhere (for example, via patchcord and a multiple you can still control any voltage-controllable parameter with these same envelope generators).*

EXPERIMENT #50: *Envelope generator control voltage with other LF control voltage of a module*

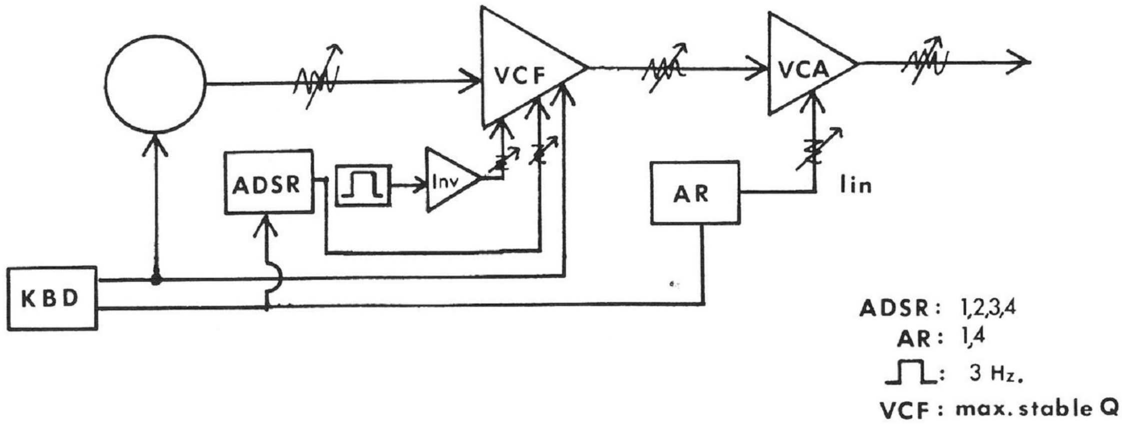
Since each of the principal voltage-controlled modules has a control voltage mixer, you can combine various control voltages to have different effects.



The patch shown here is meant to be a general one, with you deciding which LF wave and which ADSR attenuator settings to use. At any given instant the sum of the LF wave voltage, the ADSR voltage, the initial F_C voltage, and the KBD CV into the VCF will determine the F_C.

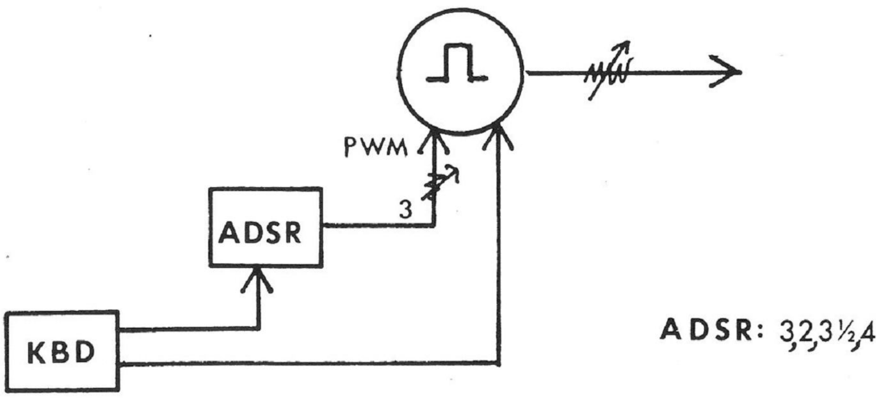
As an example, an interesting combination of envelope and LF voltage into the VCF is shown in the following diagram.

Needed: inverter



The effect here will be akin to an echo, because the long release is interrupted about three times a second by the inverted LF square wave.

EXPERIMENT #51: Envelope generator control of pulse width



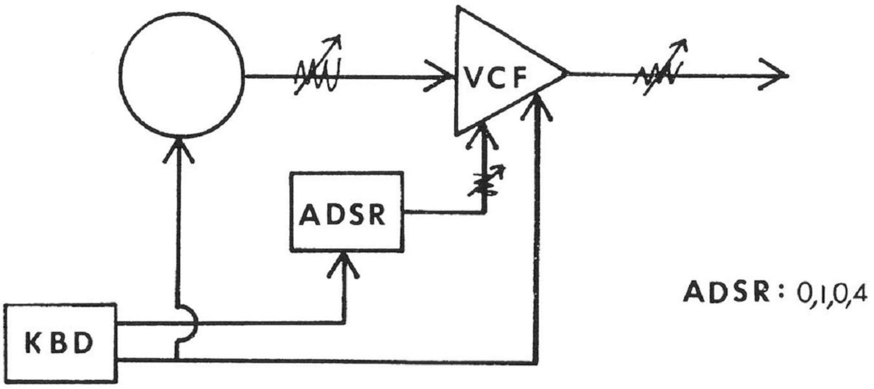
If you open the control attenuator for pulse width modulation over level 3, you will drive the duty cycle of the pulse over 100% and hear nothing.

As you will hear upon setting up this patch, controlling the pulse width with an envelope generator is tantamount to causing a change in timbre. As the control voltage output by the envelope generator changes, the pulse width varies. This produces different harmonics, each different pulse width having its own characteristic timbre (Experiments #8 and 9).

EXPERIMENT #52: "Staccato" and "legato" reversed

Envelope generators deserve a great deal of experimentation—they are so versatile. The patch shown here illustrates the kind of thing you'll run across if you experiment on your own in depth.

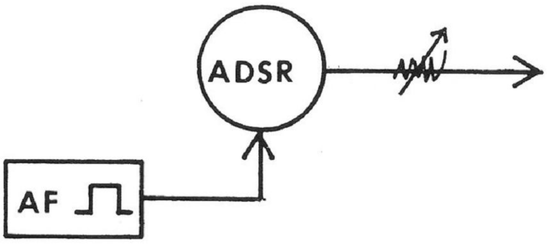
The ADSR settings are the critical aspect of this patch. If you press a key and hold it down, you expect the note to sustain, but in fact you get a staccato; the initial



decay goes quickly to 0, and sustain and release don't appear at all. If, instead, you hit a key quickly, like a normal staccato, the sound will die out for a long time, legato. The decay goes immediately to the long release.

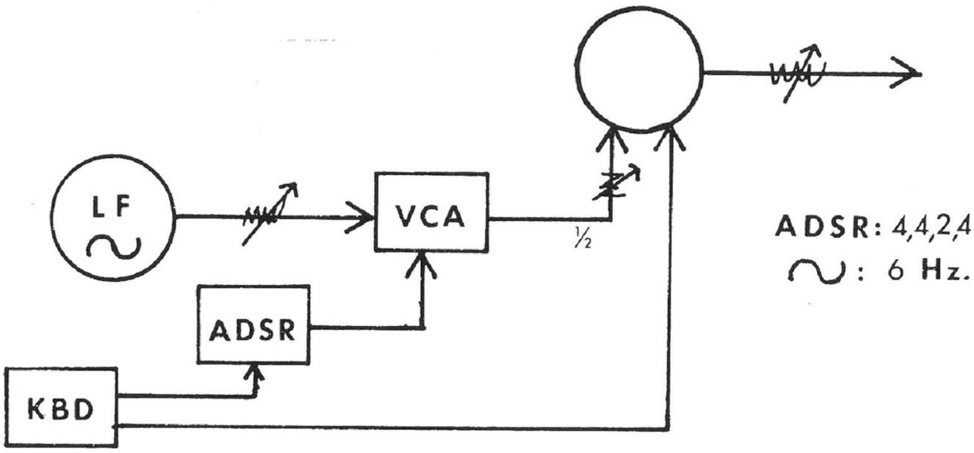
EXPERIMENT #53: Waveshaping

Although you can modify and control the various waves output from VCOs, there is not much opportunity actually to create initial waves of different shapes (and therefore different timbres) on most voltage-controlled synthesizers. However, when you gate an envelope generator with an AF wave you can create different waveshapes, which may then be further processed if you desire.



You will find that different timbres are created depending on the settings of the envelope generator attenuators. Note the use of an AF wave as a controller and an envelope generator as a signal generator!

EXPERIMENT #54: Delayed vibrato



The LF sine wave will begin controlling the VCO once the attack stage of the envelope generator has reached a certain level. The delay is proportionate to the attack time. The attenuator level setting (½) is critical.

chapter four

Three Approaches To Synthesis

I. SIMULATION OF INSTRUMENTS

It is relatively easy to imitate an instrument in such a way that a listener would be able to identify the instrument as a *simulated* tuba or clarinet. It is quite difficult to simulate an instrument with real accuracy. A primary reason for this difficulty is the complex way in which the harmonics of acoustic instruments vary over time. Figure 4-1 is a diagram of the manner in which the first twenty harmonics of a bowed violin vary in the first 3/10 second.[1] The first harmonic (the fundamental) is at the rear and the twentieth harmonic at the front. Note that virtually none of the harmonics has a smooth or typical ADSR-type envelope. Indeed, the fundamental has more than ten rises and falls (excursions) within the first 1/10 second. Only the most complex modular analog synthesizer would have the capability of creating such a complex envelope, and that envelope is only one of twenty necessary to create the harmonics in correct proportion to each other.

It might seem hopeless to aspire to create fine instrumental sounds, and yet there is no denying that many analog synthesists create superb instrument simulations with a synthesizer as relatively simple as a MiniMoog or an ARP Odyssey. Here are some hints as to how they do it, and how you can do it too.

1. Be a good listener. Do you *really* know what the instrument you wish to simulate sounds like? Certainly you can distinguish a trumpet from a guitar, but to do a credible simulation of a trumpet you must become intimately involved with it. You must have

[1] Reprinted from *Computer Music Journal,* vol. 1, no. 2, 1977, "Lexicon of Analyzed Tones" by J. Moore and J. Grey, by permission of MIT Press. Copyright © The Massachusetts Institute of Technology.

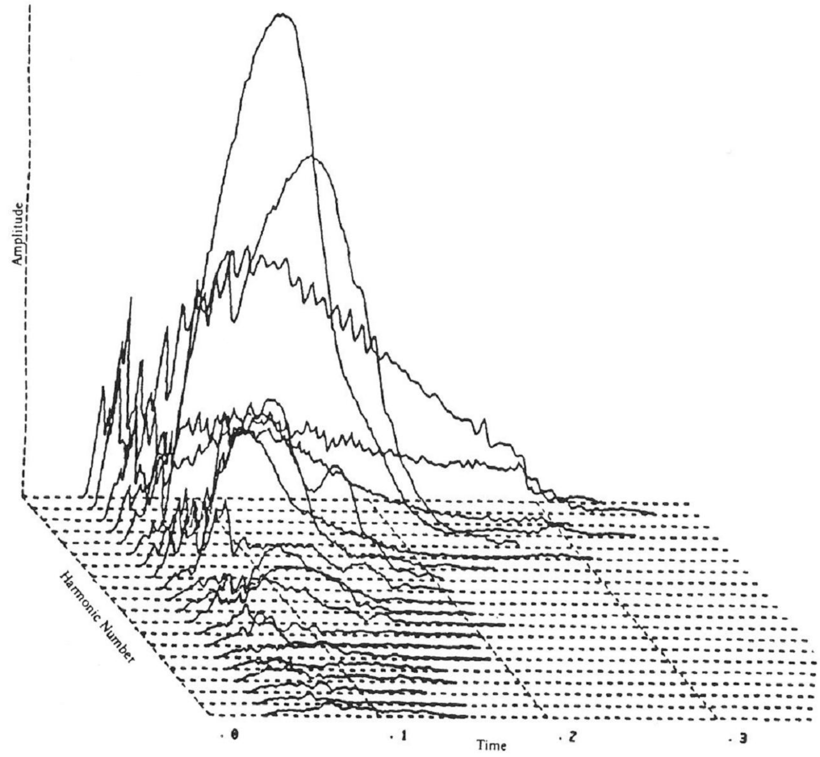


FIG. 4-1

listened to a trumpet *with awareness.* You may think you know what a trumpet sounds like, but unless you have listened with an ear to synthesizing it you haven't analyzed a trumpet sound rigorously enough.

To listen analytically with awareness, you must be able to describe the instrument's changes over time in pitch, timbre, and loudness. In so doing you will necessarily describe the instrument's envelope contours. Identify the instrument in a classical setting, in a rock setting, avant garde, solo. If possible take a portable cassette deck to the music department of a local college and ask a student who plays the instrument you want to synthesize to play for you while you tape the sound. Ask the student to play staccato, legato, high and low registers, and any or several pieces in her or his repertoire.

If you can't find a musician to help you, buy *The Instruments of the Orchestra,* a splendid two-record set on Vanguard (VSD 721/22) performed by the first desk men of the Vienna State Orchestra, with narration by David Randolph. Each orchestral instrument is demonstrated with regard to range, tone color, special characteristics, in typical solo passages and with the full orchestra. The set is especially helpful for understanding the limits of the various instruments.

2. In your initial experience with the synthesizer (and this might take years rather than months), you will probably find that the sound you have created is something like the instrument you want but not as true as you would like it to be. Allow yourself to create a caricature of the instrument rather than a perfect instrument. You wouldn't expect to pick up a trumpet and play it well within a few months; don't be less rigorous with the synthesizer.

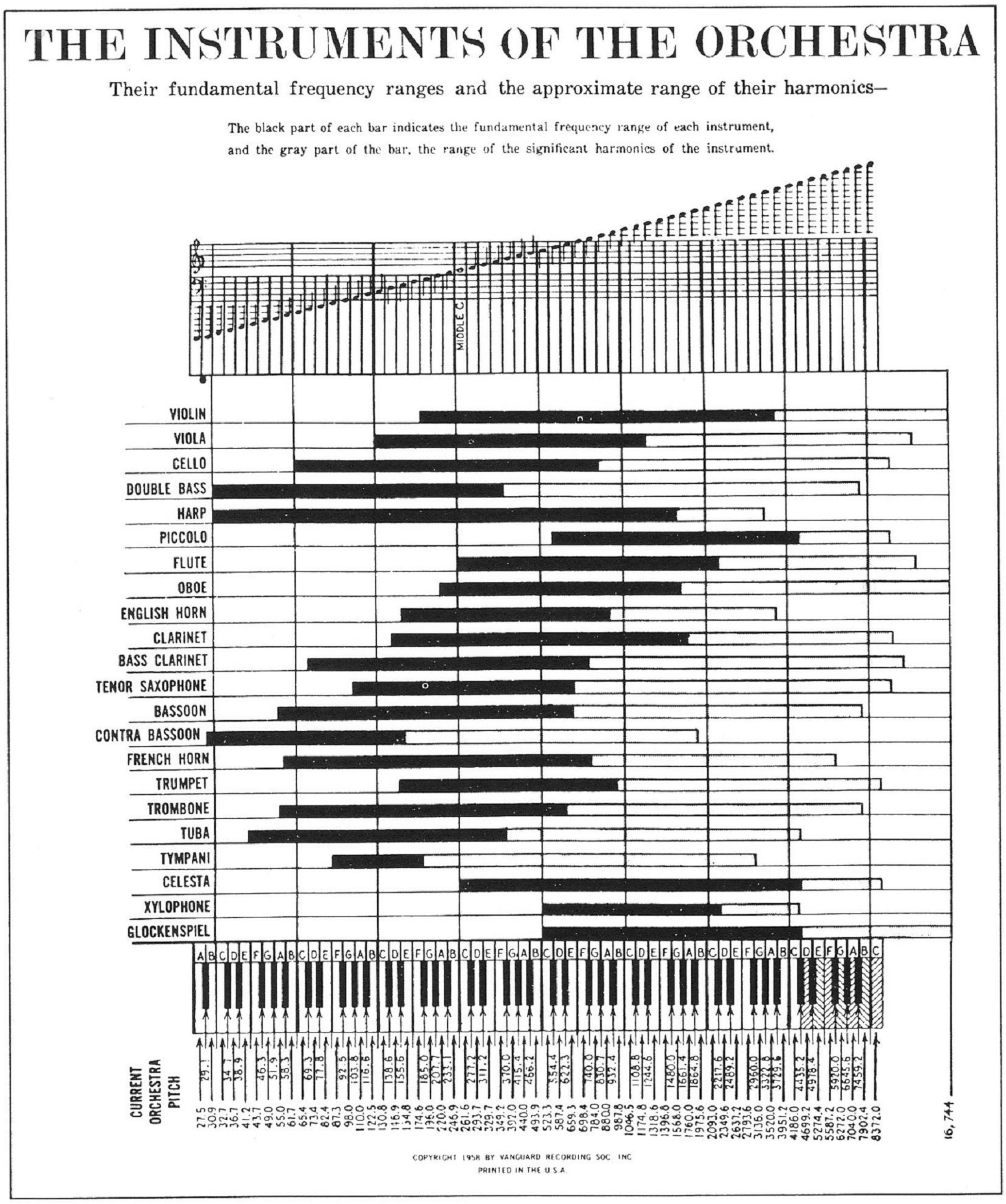
If you are dissatisfied with the sound you have, ask yourself what's wrong with it. If you were making soup and you didn't like it, you might narrow the problem down to: it needs salt. The way you know it needs salt is that you know the qualities of salt and what it will do for your soup. If you don't like the sound because something wrong has happened within a given period of time, change only that portion of the result you didn't like in that period of time. The primary reason you must be a good listener is that you must be able to say what ingredients make up the sound you're trying to simulate and analyze and then correct that which does not satisfy you.

One of the things you will hear if you listen to acoustic instruments is noise. A violinist puts rosin on the bow and drags it across the strings and it makes noise; a tuba player blows air into the instrument and that makes noise. Beginning synthesists attempting to simulate acoustic instruments commonly achieve a much "cleaner" sound than a real instrument can produce. Try using small amounts of noise to control a VCO or VCF, or as an audio input to the VCF.

3. Memorize the following: *The basic waves from which simulated bowed strings and brass instruments are created are sawtooth and rich harmonic content pulse waves. Pulse waves with a duty cycle of 20–40% are valuable for creating double-reed instruments and plucked strings (use envelope generator control of pulse width for those plucked strings).* Try a flute using either a sine wave or two sawtooth waves, heavily filtered, one tuned an octave higher than the first (because of the importance of the second harmonic in a flute). Saxophone and acoustic piano are very hard to simulate because of the unusually complex nature of their harmonics.

4. Table 4-1 gives the pitch range of all the orchestral instruments. To imitate an

TABLE 4-1*



* Table 4-1 reprinted by permission of Vanguard Recording Society, Inc.

instrument effectively, you must play within its pitch range. Generally, tune a VCO such that the lowest note of the keyboard corresponds to the lowest note of the instrument. That will give you maximum playing flexibility. However, if the passage you are to play has a small pitch range (about an octave), tune the lowest note you will play to the middle of the keyboard for convenience.

You may find that you have synthesized a believable instrument within a given pitch range (an octave) but that it loses credibility outside that range. Analog control voltages tend to be precise over a range of about an octave; with more than that they are less precise.

5. Know the material you are going to play. Be aware of the context in which the piece will be played. Is the patch you have created right for the *type* of instrumental passage called for? If the score calls for playing legato and you are using multiple triggering (Chapter 3, section IV-F), or for a fast meter and your attack times are slow, you won't get the effect you want. Everything must be tempered by what is to be played.

6. Each instrument tends to emphasize certain harmonics, but the harmonics emphasized vary depending on the pitch the instrument sounds. For example, a French horn playing D above middle C (294 Hz.) emphasizes the second and third harmonics. If you simulate a French horn in that register (pitch range), set the VCF's resonance attenuator so that those are the harmonics emphasized. However, if the French horn were playing an octave higher, it might not necessarily emphasize the second and third harmonics. You will have to experiment to determine for yourself and decide which harmonics should be emphasized for a given instrument in a given register. Table 4-2 lists some examples of which harmonics particular instruments emphasize in a given register.[2]

TABLE 4-2

Instrument	Pitch	Harmonics emphasized
Viola	440	1 strongest, also 2–6
Cello	220	1, 2, 3
Double bass	146	1–6, (3 strongest)
Harp	220	1 strongest, 2, 3
Clarinet	220	1 strongest, 3, 5
Flute	440	1
Oboe	523	1, 2, 5 (2 strongest)
Bassoon	175	3 strongest
French horn	294	2 and 3
B-flat trumpet	349	1 and 2
Tuba	233	1 and 2
Trombone	233	1, 2, 4

7. On one occasion this was my train of thought as I synthesized a tuba:

Since a tuba is a brass instrument, I'll start with a sawtooth wave into my low-pass VCF. I put the VCO in the correct pitch range, having checked the material I'm going to play. Table 4-2 gives me the formant of a tuba so I set the F_C around the second harmonic and play with the resonance control just a little bit to see if I like it.

The mouthpiece of a tuba is large and there is a lot of tubing which the player blows through to get sound. Therefore the attack time is going to be quite long, much longer than for a trumpet, a brass instrument with relatively little tubing.

I set up some beginning ADSR settings: a long attack, some decay, sustain, and release. The ADSR controls the filter. Now I ask what's wrong. Did it start

[2] Charles D. Culver, *Musical Acoustics*, 4th ed. (New York: McGraw-Hill, 1956).

realistically? No, the attack time was a bit fast, so I slow it down. Also I hear a buzzing that I don't hear with a real tuba. The filter is opening too much, so I lessen the amount of ADSR into the filter. I listen to the sound and define what is wrong with it. Then I correct that and nothing else.

Lessening the ADSR doesn't make the sound better as I had thought it would, so I'll lower the intitial filter cutoff frequency. Now the attack sounds realistic. The sound has a buzziness now when it sustains so I lower the sustain level on the ADSR. That's better; in fact, it's far too good. A real tuba player puts wind into his instrument. I put a small amount of noise into the VCO. I try a small amount in the filter. It's very subtle, hard to pick out; yet I think this has added to the realism of the patch.

As I listen to a real tuba player I can distinguish the player from the horn, so it occurs to me to control the filter with two envelopes—one will be the player and one the horn.

The particular synthesizer used offers one ADSR and one AR envelope generator. Since I've been using the ADSR for the horn, I add AR control to the filter for the player. That doesn't sound right no matter how I set the attenuators so I reverse it, letting the ADSR be the player and the AR the horn. I give the AR a long attack time and the ADSR very short parameters, about 1½,0,0, just enough to put a little additional "bite" at the moment the player blows through the horn. The AR is heard long after the ADSR has shut down.

Since I know that one of the uses of a lag processor is that it can function much like an AR envelope generator (Experiment #61), I substitute that for the AR. This frees the AR to do something else in the patch if that's required. This is thrifty use of the modules available. I've got a tuba sound that works fine and I'm using only one VCO, the VCF, the ADSR, and the lag processor.

In this example I used two control voltages to control the VCF and didn't use a VCA at all. In another patch I might use one control voltage to control both the VCF and VCA. In general, I want to use the least number of modules consistent with obtaining the most realistic sound.

This example is not necessarily the "right" way to create a tuba sound. It is the way I did it on a particular occasion. There is more than one way to simulate a given instrument. I might do this differently another time. You might do it differently.

8. Reverberation helps create a more "natural" ambiance for almost all instrumental simulation patches. Always experiment with adding Reverb as the last module sparingly, when you have a patch with which you are otherwise satisfied.

9. Although all the hints in this chapter are important, I want to particularly emphasize the importance of keyboard technique. No matter how good the patch is, if you play a violin like a piano it won't sound right. Nuance is that degree of control which an individual performer brings to the instrument. It is what distinguishes a Horowitz or a Perlman from a journeyman pianist or violinist. In terms of expressivity it is everything. It is one reason that Jan Hammer's MiniMoog sounds so much like an electric guitar.[3] Another reason is that Hammer plays characteristic guitar riffs.

10. The patches and brief descriptions below are meant as starting points. Because no two models of synthesizer sound exactly alike or have exactly the same control sensitivity, you will have to listen, work, and spend the time necessary to create effective instrument simulations on your synthesizer.

[3] See Chapter 5, footnote 9.

These patches are tailored to synthesizers that have only VCOs, a low-pass VCF, VCA, keyboard, LFO, and an AR or ADSR envelope generator. The more other modules (particularly filters and controllers) and *outboard devices* (like graphic and parametric equalizers) that you have available, the more you increase the possibilities for realism.

Figure 4-2a is the basic patch for brass; someone listening would think it sounded something like a trumpet. Figure 4-2b is a refinement. Note the settings in the latter for the ADSR controlling the filter: ¾,0,0,0, with control level of 1¾. The effect is to cause the filter to open very quickly a discernible distance and then immediately close. However, the filter doesn't close, because the AR is still opening it. The total effect is to create a "bite," an immediate attack followed by a second attack. The first represents the person initially blowing into the trumpet, the second the horn itself.

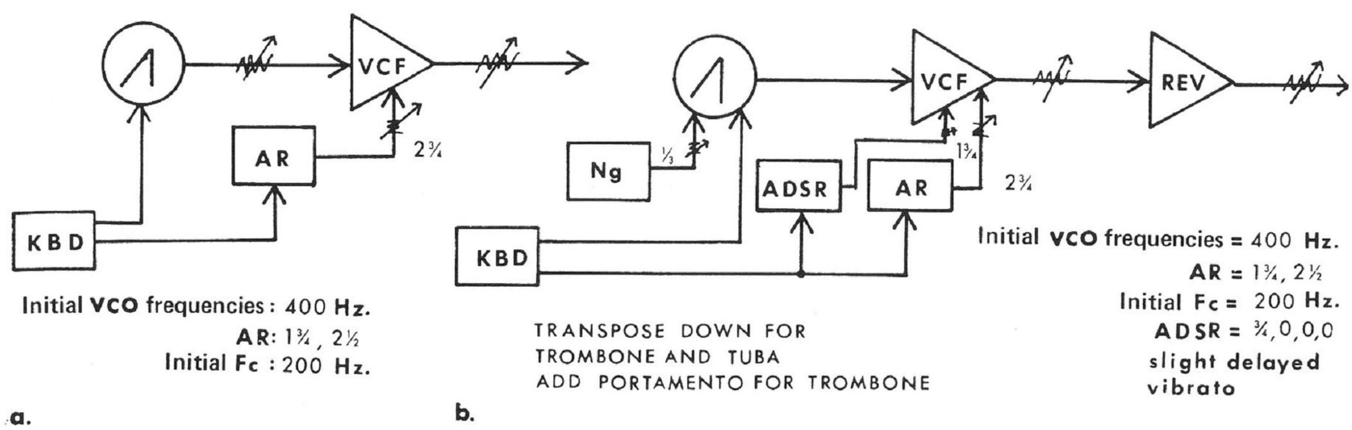


FIG. 4-2. Trumpet (and brass in general)

Note also the slight use of noise to create the imperfections referred to earlier, and the use of a slight delayed vibrato.

Use a transpose switch (if your synthesizer has one) and transpose down two octaves to get an initial tuba or trombone sound. Use portamento (Chapter 5, section III-B) with the trombone.

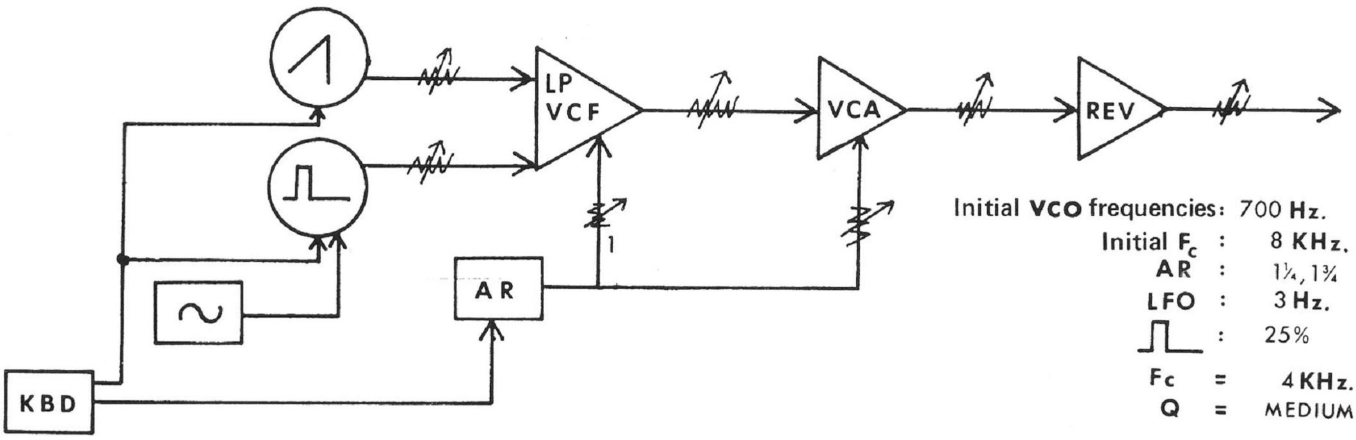


FIG. 4-3. Violins

When a violin is played a bow is drawn over strings that come over a neck onto a tailpiece. Since the strings are in a somewhat curved configuration, the bridge rocks imperceptibly on the face of the violin, causing the strings to move and create a vibrato. Thus there is a vibrato even if the performer does not consciously create one. The vibrato that is consciously created is a second vibrato. Create the first vibrato by slightly detuning the two VCOs causing a slow beat frequency. The second vibrato is created by an LF sine wave controlling a VCO. It does not control both VCOs because that would mean both violins always have the same vibrato, which is not what happens. This way a vibrato will be perceived but it will be just a bit different from the no-vibrato sound coming from the other violin (VCO).

Since the bow hits the strings and then comes away from them, we will use just a simple AR envelope for amplitude. The setting gives a slight attack and release.

The patch in Figure 4-3 succeeds or fails by the nuance given by the player. You can accent a note by rapidly inserting and deleting an additional vibrato. You'll play differently if it's a *solo* violin.

Lower the pitch range and initial filter cutoff frequency for the sound of a viola; lower them even more for a cello. Generally the larger the instrument the longer the attack time and less natural the vibrato.

Figure 4-4 shows the basic patch for the electric bass.[4]

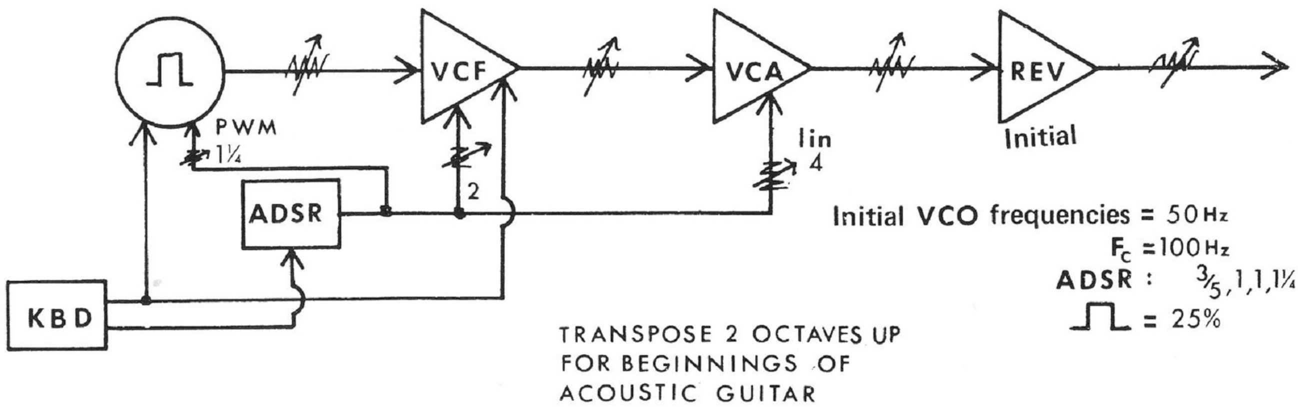


FIG. 4-4. Electric bass

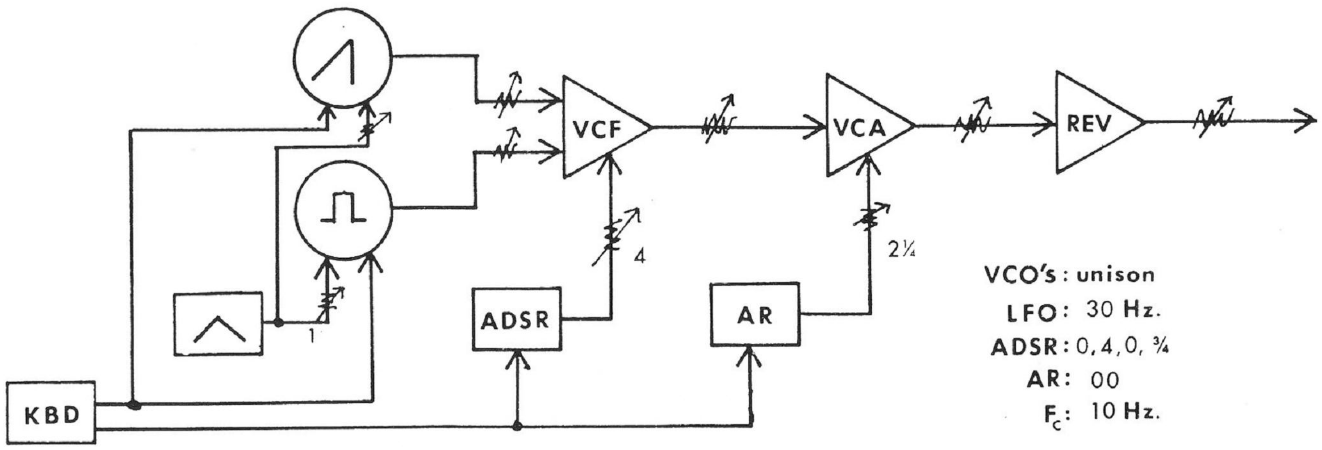


FIG. 4-5. Amplified electric lead guitar

The patch in Figure 4-5 is for amplified electric lead guitar. Keyboardists are used to playing laterally; that is the physical limitation of a keyboard. The physical limitations of a guitar are very different because of the way the frets are constructed and the tuning of the instrument. Furthermore, the lead guitarist doesn't play the same notes or riffs that a pianist does. Rock and other guitarists play diatonic scales, harmonic and melodic minor scales, and blues scales. Just as with a violin, pitch bend and vibrato are important ways of adding realism to a guitar patch, but they are employed very differently. Since the guitarist stays on one string and on the same fret when either bending a string or adding vibrato, the realistic way to add vibrato with an electric guitar patch is to use pitch bend only. To go slowly down in pitch, stay on the same note and change pitch with the pitch bend. In fact, pitch bend is the only keyboard control to add expressivity with this patch.

[4] See Roger Powell, "Synthesizing a Bass Tone," *Contemporary Keyboard*, October 1981, p. 66.

A guitarist bends a string down and plays either a half or whole note below the note he or she ultimately intends to sound. The guitarist will also play a minor third up and "between the notes." These changes should use only a pitch bend device.

The 30 Hz. LFO in the patch adds "distortion" to this electric guitar patch.[5]

II. SOUND EFFECTS[6]

THE GENERAL STRATEGY FOR SPECIAL SOUND STRUCTURES

The general strategy given below is a step by step procedure that will not guarantee that you will get the sound you need, but it will almost always give you a reasonable start. Basically, the steps are to first get the time structure, then the major sound component, followed by a major refinement, and then a number of lesser refinements.

STEP 1—TIME BEHAVIOR: *The first step is to provide the necessary control for the time behavior of the sound to be produced. This may seem simple and obvious, but what may not be so obvious is that it* must be done first. *For example, if you are going to produce thunder or gunshots, you can probably guess that you start with white noise, but if you try to refine the basic sound without providing the proper envelope, you will be in for a surprise. Things sound very different under different envelopes. Thus you need to set up an envelope generator to give short envelopes for gunshots, long envelopes for thunder, and there are cases where you need periodic envelopes. In such cases, an envelope generator can be periodically triggered, or a periodic waveform can be used to control a VCA.*

STEP 2—MAJOR SOUND COMPONENT: *The second step is to determine the major sound component and place it under the previously determined envelope. In many cases, you will use as the major component either a perfectly periodic waveform, or white noise. However, one must be prepared to use other major components such as heavily modulated waveforms. For effects that employ relatively long envelopes, you will be able to work at the basic sound as a static waveform and then try it under the envelope. In the case of short envelopes, you will have to work with the major component already under the envelope. In such cases, even if the effect is not to be periodic, it will be useful to employ periodic triggering during setup as this frees both hands for the manipulation of controls. When the proper combination of major sound and envelope are chosen, you can then switch back to single event envelope triggering. There is probably no single or best solution to the major sound component selection problem. You will find several possibilities and should write them all down in case something doesn't work well later and you have to backtrack.*

STEP 3—MAJOR REFINEMENT: *Upon completion of steps 1 and 2, you will have an effect that is probably close to what you want, or at least you have an idea as to the best you can do at this point. The next step is to make a major refinement. This step is major in that it will set the effect you intend to create apart from a larger group of possible effects. Yet it is a refinement—something important and yet possibly subtle, and it may take a lot of work to get it. There are many ways of getting this. Careful listening to the real effect and intelligent experimentation are probably the best ways, yet one should not be afraid to try things that seem unlikely if it does not mean tearing down all that has been accomplished so far. At the completion of this step, a listener should be able to say "Yes, that's a ____." It is not*

[5] For another interesting discussion of imitative and non-imitative synthesis and how to synthesize percussion, see Jim Aiken, "Modular Synthesizer Effects," *Polyphony,* May/June 1981 and July/August 1981.

[6] This entire subsection is reproduced from *Electronotes,* no. 80, pages 3-6, with the kind permission of its publisher. Bernie Hutchins. For more on *Electronotes* see the Bibliography.

necessary that he say "That's a very good ____," but only that he has a good idea what you are trying to make.

STEP 4—LESSER REFINEMENTS: *This step is rather obvious in what it is intended to accomplish. We want to make the additional refinements that will make the effect all the more useful and professional. While these minor refinements may be the most difficult, they are the most rewarding as they make a good job better. You can work on them with the knowledge that you have made a good cake, and even if you don't decorate it nicely, it is still a good cake. Very likely, you will find many possible minor refinements, and these should be carefully recorded, especially if you have to sell the sound to a buyer. His idea of a good ____ may not be the same as yours. It won't help you to know that you once made it a little different if you don't have the faintest idea how you did it. Always keep good records of how special effects were created.*

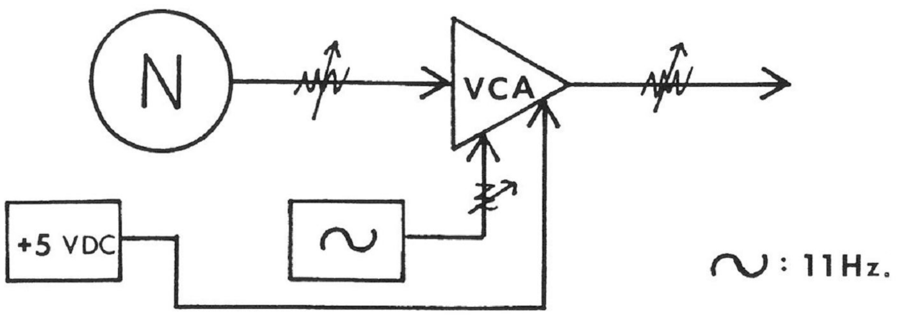
HELICOPTER SOUND

A couple of years ago a student was studying helicopter noise and just for fun we decided to see if we could learn anything about helicopter noise by trying to synthesize it. We will use this as our first example.

In the first place, we have to know exactly what we are trying to synthesize. A helicopter far in the distance has a very different (thump-thump-thump) sound than one close by, which contains the thump, but also a "swish" of the rotor blades and engine noise. To make the example more general, let's suppose we want the helicopter to fly up from far away and then go off again.

In step 1, we determine time behavior. The dominant time behavior has two aspects. First there is that of the distance of the helicopter. In the case where the helicopter comes closer, the overall amplitude increases, but more importantly the rotor swish and engine noise become apparent. The second aspect of the time behavior is periodic thumping sound as the rotor blades rotate. Thus we have the periodic thumping that is present in all cases, and other effects which will come in with one or more envelopes that represent a single flyover. It is natural to suppose that the flyover envelopes should control one or more VCAs, but another possibility can be suggested based on the physics of sound propagation. Why is it that we hear only the thumping when the copter is far away? Because the air attenuates high frequencies more than low ones, and there is more air further away—you say. Thus we can consider a greater distance through the air to correspond to a low-pass filter with a lower cutoff frequency. Thus we may want to use a VCF as a means of shifting the major and lesser components of the overall sound. This is just something we want to keep in mind at this point.

In step 2, we set up the major sound component. This is the "thumping" of the rotor blades. A little bit of experimenting will convince you that white noise confined to periodic bursts will give the general sort of effect we need. The setup of this diagram can thus be considered as our step 2 helicopter:



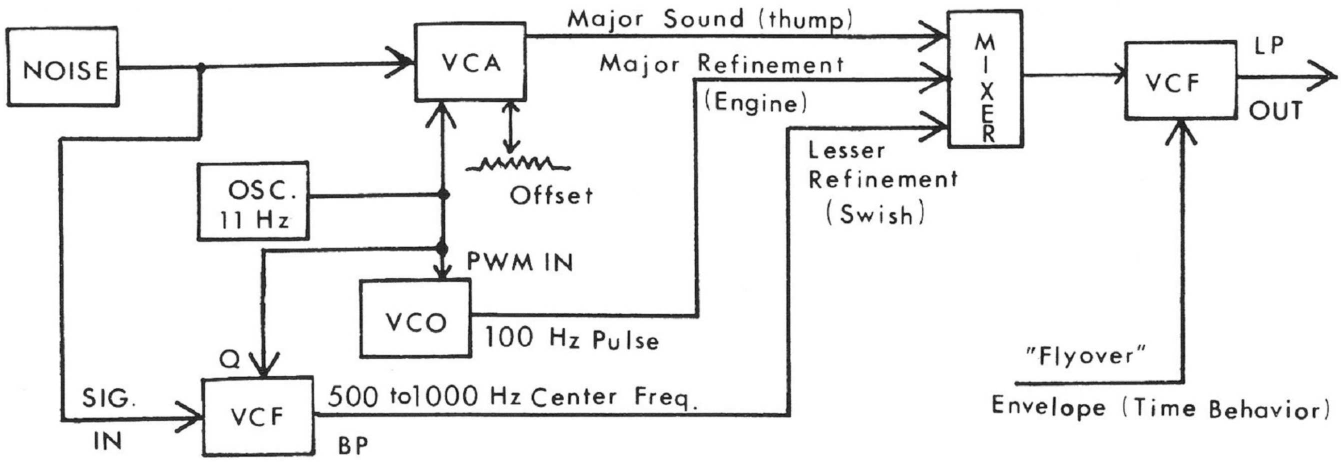
At this point, we can experiment with the offset of the VCA, which determines the relative duration of the noise pulses, and the frequency of the oscillator, which corresponds to the rotational rate of the rotor blades. This done we have a pretty good helicopter in the distance and we can make it fly a little closer (but not too close) by simply changing the overall amplitude of the noise thumps.

The third step is to make a major refinement, which in this case has to be to add in the engine noise. We could have considered the rotor "swish," but we do not expect to hear the swish without the engine running, so we do that first. If we think about how a standard piston type engine works, we can guess that the sound will be a rapid series of pops corresponding to exploding fuel in the cylinders. In fact, if you experiment with the waveforms you have available, you will find that the sound is most realistic if you use a pulse waveform (in the range of 80 to 150 Hz.) with a fairly short (10% say) duty cycle. Also, it helps if you use a little modulation of the pulse train. The modulation frequency should be the same as the "thump" (rotor rotation rate) frequency, and you may find that you prefer pulse-width modulation to frequency modulation in this case. In either case, the modulation should be quite small, or else you start to produce science fiction helicopters.

The fourth step is to add the lesser refinements. In this case we want to add in the swishing effect of the helicopter rotor blades. We know that in general wind-like and swishing effects are obtained by bandpass filtering white noise. This is what we will try here. We then have to consider such things as the bandpass frequency, the Q, and any modulation of the bandpass frequency. Here is where a good deal of experimentation is required, but there are a few things that we can work out ahead of time based on physical intuition. If the swish is produced by the blade, then we expect that the swishing sound will be a little different depending on the distance from the rotor hub because the blade is moving faster the further out you go. Furthermore, we expect some sort of Doppler shift because part of the time the blade is moving toward us and the other part of the time it is moving away. Of course, the opposite sides of the rotor move in opposite directions, but there is an overall shifting effect that is probably very complex if it were fully analyzed. In any case, we will try to use the VCF to simulate the overall effect. It is also possible to consider the use of variable phase shift networks, and even a VCF providing a frequency modulated "whistle" can be used, although these are probably to be considered additional lesser refinements.

The second diagram shows the overall setup used to simulate the helicopter sound. Here we have used the major sound and major refinement as discussed above. The lesser refinement (swish) is accomplished by using voltage-controlled Q on the bandpass VCF. This lesser refinement is interesting because in some ways it seems to do the whole job. It does sound like a helicopter locates somewhere. However, if you try to make the copter fly away by lowering the output level, you will soon see that it is not realistic because there is no attenuation of high frequency components. In practice, the individual components input to the mixer are adjusted separately. Then a satisfactory mix of these components is set up to simulate the helicopter at its closest approach. Then the flyover is simulated by raising and then lowering the cutoff frequency of the final low-pass VCF. An ordinary AR type of envelope can be used here if the time constants are fairly long, or the filter can be controlled with a manual knob, a keyboard, or a ribbon controller. [See Glossary for definition of ribbon controller.*] Of course, when the final patch is set up and working, you may still want to make some adjustments of the patch parameters to get exactly what you want.*

Two important points can be seen from this example. First, the combination of modules used (e.g., two VCFs) may be quite different from the combination used for



musical sounds. Secondly, the modules are controlled in many cases by what would be considered secondary control in musical sounds. In this example, the VCO is controlled in a PWM mode and the first VCF in a Q mode, unlike the usual pitch control modes in musical examples. These two facts should not seem strange when we consider that we have gone from the synthesis of musical sounds to a more general type of sound. We therefore must be careful not to restrict ourselves to the musical way of doing things when we are trying to produce special effects.

III. SYNTHESIS AS A MEDITATION

Until now we have, of necessity, approached synthesis from a rational, analytic viewpoint. In order to produce a sound, we consider the way pitch, timbre, and amplitude vary over time. This is the Western, scientific approach: Divide something into its constituent parts and you will be able to define and reproduce it. It is one approach, a good approach, but not the only approach.

Once you have learned the principles in this book so well that they have become a part of you, you are ready to approach the synthesizer with "beginner's mind." This is a Zen concept: If you have become an "expert," then you "know" what can and cannot be done and therefore limit yourself. The beginner's mind is empty, free of the habits of the expert, ready to accept, to doubt, and open to all possibilities.

This approach to the synthesizer is intuitive, nonlinear, a cybernetic meditation. The synthesist can become indistinguishable from the machine and the music. The instrument becomes an extension of the person. Your hands move and you have a broad idea of what will happen when you control or modify a module with another module, but you are not going anywhere. There is no goal. Sound happens. When the process has reached completion you know and you stop. The entire process becomes a gestalt rather than a series of independent actions.

A sound or series of sounds may have been created which do not require your active participation; you can "turn off your mind, relax and float downstream." Another patch will require a constant active interaction between you and one or more of the synthesizer modules and you become a part of, indistinguishable from, the music. Some time later you realize you have been in a trance-like state of consciousness, but at the time "you" did not exist. Such moments are grace.

By its very nature little can be written about the meditative approach to synthesis. It is real; it is valid; and it is an experience you can have if you open yourself to it.

chapter five

More Synthesizer Modules

Synthesists frequently use modules other than the basic ones discussed in Chapter 3. Some, like mixers and attenuators, are found on all synthesizers; others, like a noise source, ring modulator, and sample and hold, are on many synthesizers; still others, like a lag processor and envelope follower, are generally found only on modular synthesizers.

The modules described in this chapter are grouped either as the noise generator (the only signal generator other than a VCO), or as modifiers and controllers, which is the way in which these modules typically function. However, in a given patch a module may function atypically; for example, Figure 4-2b shows a noise generator, typically a signal generator, functioning as a controller. Keep in mind that there is nothing sacred about a module's being defined by its typical function.

I. NOISE AND THE NOISE GENERATOR (NG)

You have already been minimally introduced to the noise generator (primarily a signal generator), since it has been used in various experiments. In Chapter 2 it was defined as an oscillator that generates all frequencies simultaneously. Actually *white noise* occurs when at any given instant all voltages are present but their amplitudes are random. There is equal energy for each Hz.

Figure 5-1 is a graph of white noise. It indicates the presence of all voltages whose amplitudes vary randomly.

Listen to white noise by opening the appropriate noise attenuator(s) and patching the NG directly out.

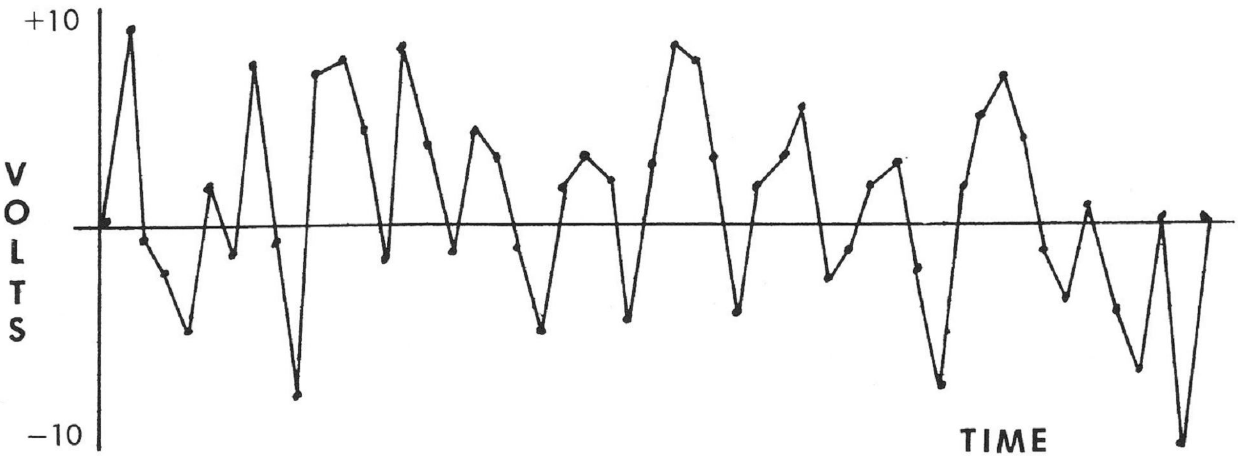
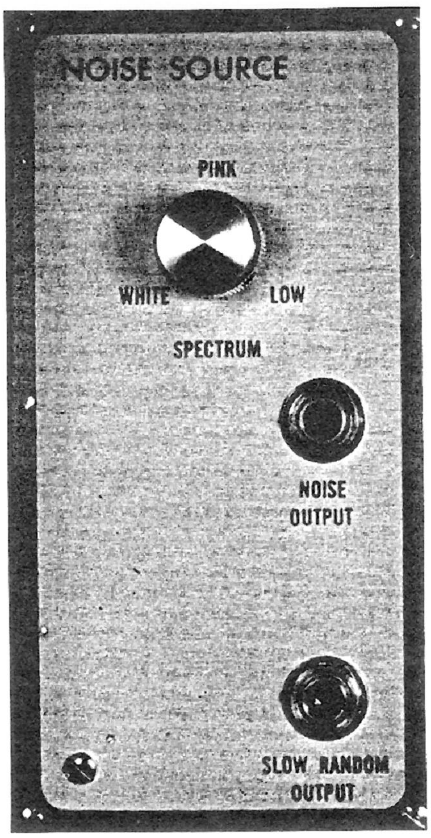


FIG. 5-1

If the probability of hearing any frequency is the same as that of hearing any other frequency, why do we perceive white noise as having the highs predominate over the lows? You can verify that this is true by varying the "color" (or spectrum) attenuator and hearing what noise with low frequencies accentuated sounds like.

The answer is as follows. The hearing range of humans is about 20 Hz. to 20 KHz., a span of ten octaves. As shown on page 33, the first six octaves contain a total of about 1260 Hz.; the next four octaves contain about 19,000 Hz. If we arbitrarily say that the first six octaves are low and the next four are high, there is much more energy—about 19 times as much—in the higher four octaves than in the lower six octaves. The lower octaves contain far fewer frequencies than the upper ones. Thus even though it is true that in white noise there is an equal probability of any frequency being heard, we hear the highs more because there are 19 times as many high frequencies that might be present as low frequencies.



E-mu Module 2400 - Noise Source

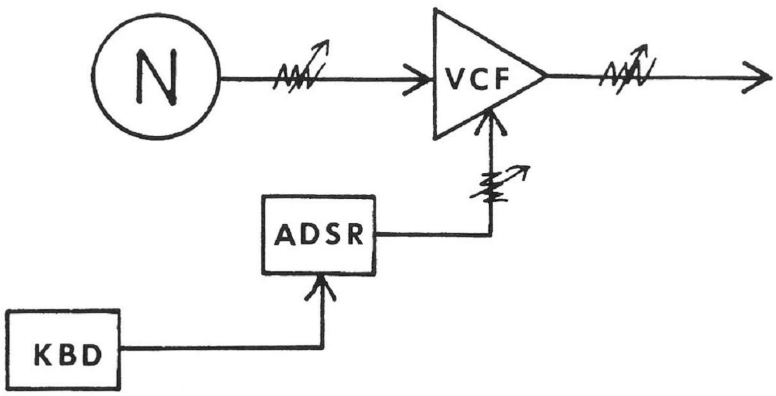
Pink noise is created when there is *equal energy* per octave. Pink noise is created if sufficient high frequencies are filtered from white noise such that each successive octave would have half the energy that it previously had. The color or spectrum attenuator is actually a filter whose slope is −6 db/octave.[1] Filtering even further, creating "red" noise, creates more energy in the lower octaves than in the higher; this is experienced as a low rumbling noise.

White and pink noise are typically used to provide percussive and wind-like sounds as audio signals, and to provide a random voltage source as either a controller or input to a sample and hold module.

Pitched noise is obtained by using a noise generator while utilizing the ability of a filter's resonance circuit to emphasize one very narrow band of frequencies; since all frequencies are present in noise, one band of frequencies will be emphasized with a highly resonant filter.

[1] The slope of a color attenuator can be said to be −6 db/octave if one is measuring voltage, as we have elsewhere in this book; if one is measuring power, the slope would be −3 db/octave.

EXPERIMENT #55: Pitched noise



All you need do in this patch is vary the position of the resonance and initial F_C attenuators, the ADSR attenuator into the filter, and (if you like) the ADSR settings to get different examples of pitched noise.

Noise is, of course, very important in the creation of percussive sounds, drums of all kinds, cymbals, gongs, and the like. Perhaps the simplest experiment involving noise and the filter is the familiar surf or rocket sounds created by the patch in Figure 5-2; vary all attenuators of the noise generator and the initial F_C and resonance attenuators.

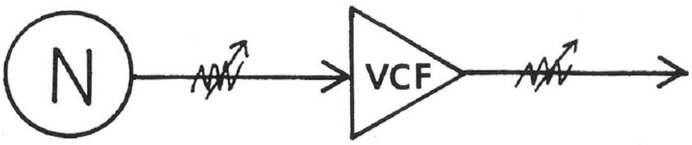


FIG. 5-2

If you control a high-pitched VCO with noise, a timbre is created which can then be otherwise modified to create interesting sounds. This is *not* pitched noise, but rather the control of a high VCO frequency by noise—many frequencies which (for practical purposes) occur simultaneously.

Another reason to control a VCO with noise arises when simulating acoustic instruments (see Chapter 4, section I).

Noise can be filtered, and used in conjunction with a pre-amp and envelope follower as a source of random control voltage (see Figure 5-3 and section 3-E of this chapter).

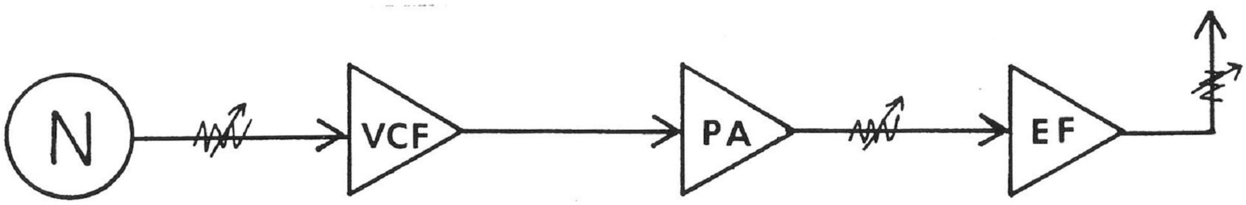


FIG. 5-3

This is particularly effective with low-frequency "red" noise. Many modular synthesizers have a slow random voltage output associated with their noise generator. This produces a control signal whose energy is greatest in the 1 to 10 Hz. range. A slight amount of slow random modulation in an acoustic instrument simulation adds warmth. A slow random output attenuated to a range of ±1 volt controlling an AF VCO would cause the VCO to "slip and slide" over a two-octave range at a rate of about 7 Hz. (the frequency of a normal vibrato).

II. SIGNAL MODIFIERS

A. Mixers

A mixer is a device that allows one to combine otherwise independent signals of varying levels into one (or more) final signal output from the mixer. In the experiments you have already done you have mixed signals many times. Examples are Experiments #28, 12, and 29, where you added signals to themselves, added signals from different VCOs, and added a negative offset to abate clipping, respectively.

By way of review:

Every VCO has a control voltage mixer that has several inputs and one output. If the mixer had three control inputs it would be called a "3 by 1" mixer (written 3 × 1). Typically two of the VCO's inputs would have attenuators associated with them (attenuated inputs); one input would be unattenuated (the one with keyboard control voltage pre-patched to the VCO). Note that with each VCO the output of the control voltage mixer appears as the instantaneous frequency of the VCO.

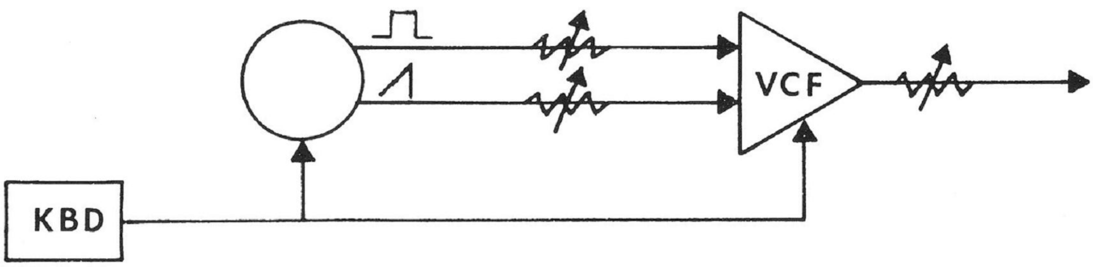
A VCF has two mixers associated with it: an audio voltage mixer, with attenuated inputs and one output, and a control voltage mixer, one of whose inputs is generally unattenuated (with a keyboard control voltage pre-patch). To use the VCF module merely as a mixer, input the signals to be mixed into the *audio* inputs to the filter and open the filter completely, letting all the input signals pass. Inputting them into the control inputs would only allow them to control the filter.

Aries AR-323 Dual Mixer

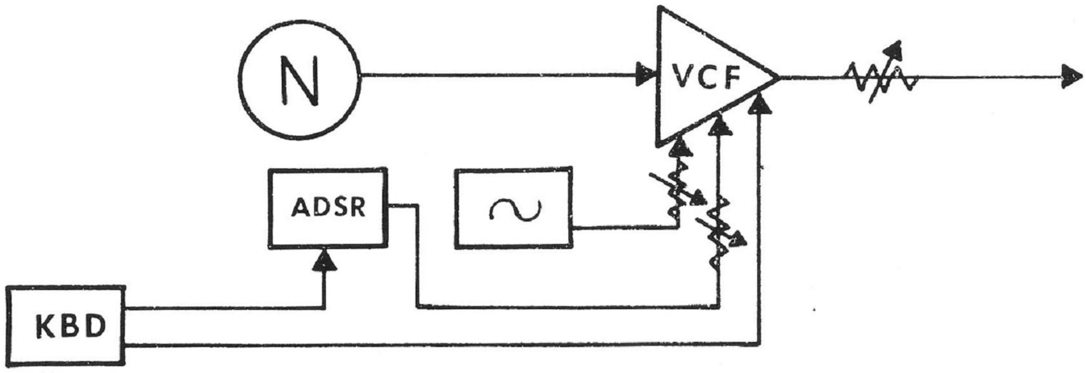
Similarly, a VCA has an audio voltage mixer and a control voltage mixer associated with it. Generally, all inputs are attenuated. You can use the audio inputs to mix two signals; the signal input into one jack may be inverted. If you will not otherwise be inputting control voltage, open the gain attenuator to fully open the VCA, and the mixed signals will be available at the output.

EXPERIMENT #56: Examples of uses of mixers

Remember that a VCF has two mixers associated with it: an audio voltage mixer and a control voltage mixer.



In the first diagram, a pulse wave and a sawtooth wave from the same VCO are input to the audio voltage mixer of a VCF. Raising one attenuator or the other gives you either audio wave. Raising both attenuators mixes the two waves, adding their voltages at every instant, and creating a third wave that will have its own unique timbre. Thus you can change the timbre of your audio signal in real time by merely raising or lowering the attenuators. You can also change the timbre by moving the pulse width attenuator associated with the pulse wave.



In the second diagram, raising the control attenuator over either the ADSR or the LF sine wave will give you a certain effect; raising both of them will give you their combined effect and, of course, raising one more than the other gives different effects.

You can combine the examples shown in the two diagrams for even more timbral flexibility (see section V in Appendix A).

You cannot use either a multiple or a "Y-cord" (a patchcord with two inputs and one output) to mix signals. Both of these devices will give you the *average* rather than the *sum* of the input signals.

B. Attenuators

An attenuator is a type of voltage multiplier. It can be thought of as in Figure 5-4, where V_I is the input voltage and V_O is the output voltage. The value of V_O will depend on the gain factor (the position) of the attenuator, which cannot be more than 1.0. Thus *any voltage input to the attenuator can only be cut—never boosted.*

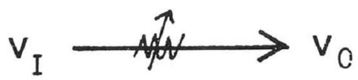


FIG. 5-4

Although audio and control inputs on synthesizers generally have associated attenuators, most outputs have no attenuators.

To attenuate an unattenuated input or output, use a "floating attenuator," an attenuator that has specific outputs so that it can be used between two other given points where an attenuator is needed.[2] (See section VI in Appendix A.) Floating attenuators are available only on quasi-modular and modular synthesizers.

A floating attenuator may also be used to attenuate external devices input to the synthesizer. For example, take the output from another synthesizer through a floating at-

[2] Aries offers a module called a "hex attenuator," which is really six independent floating attenuators for use with their modular systems.

tenuator and send it to the left side, put your synthesizer signal through the right side, and create a binaural effect (Figure 5-5).

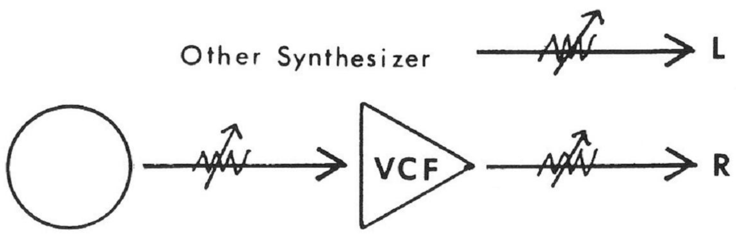


FIG. 5-5

C. Inverters

An inverter is a negative voltage multiplier; the gain factor is always −1 (Figure 5-6). Therefore, every input value is output with reversed polarity (see, as an example, Figure 3-11). If the input to an inverter is a symmetrical wave (i.e., sine, triangle, or A-C square wave), the output signal is 180° out of phase with the input signal (see section VII in Appendix A).

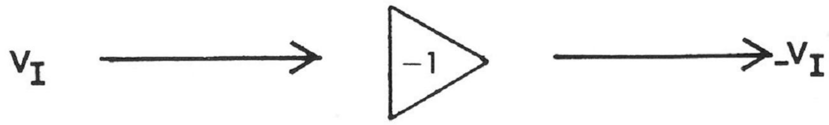
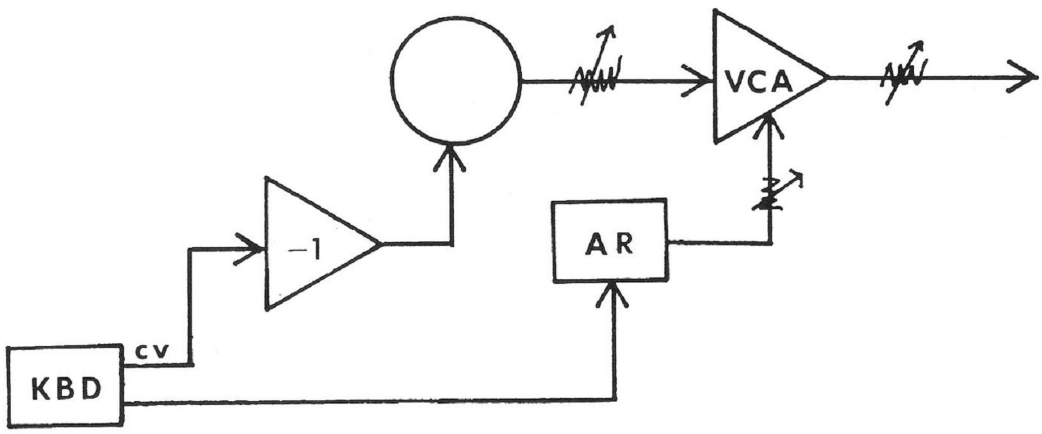


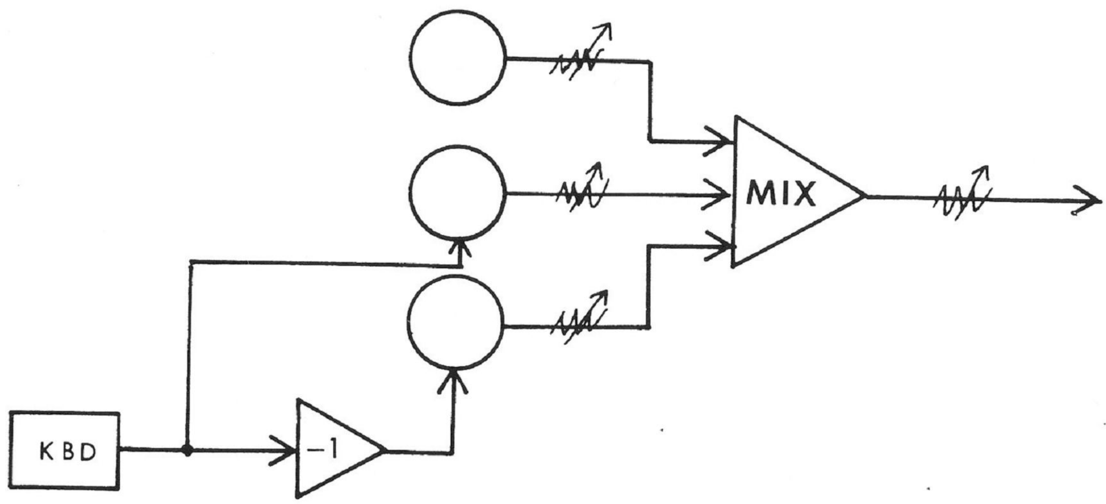
FIG. 5-6

EXPERIMENT #57: Keyboard inversion

You can invert the keyboard control voltage, which will cause the keyboard to play like a reverse keyboard.

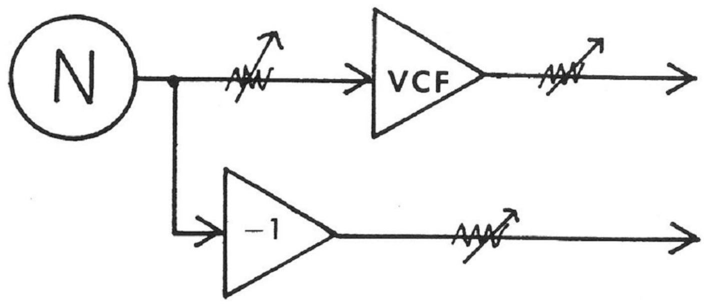


As a variation on the first diagram, keep the inverted keyboard as you have it, insert a dummy plug into the KBD CV input of another VCO, and tune three VCOs to unison. Now if you play the keyboard, one VCO will change pitch normally as the keyboard changes at the rate of 1 volt/octave, a second VCO will remain constant since the KBD CV has been dummied out, and the third VCO will be reversed, tracking at the rate of −1 volt/octave.



EXPERIMENT #58: High-pass filter

If your patchable synthesizer has only a low-pass filter, you can "kluge" a high-pass filter (see page 58) by mixing a filtered uninverted signal with a nonfiltered inverted signal.



There are two signal paths in the diagram. The upper is straightforward; noise is patched into the VCF and the VCF is patched out.

The other signal path is obtained by patching the NG output to an inverter input and the inverter out. The only tricky part is when you open the latter attenuator out. Do it slowly and at one point you will hear the low frequencies drop away. That's the point to stop raising that attenuator. If you raise it even more, the low frequencies will come back.

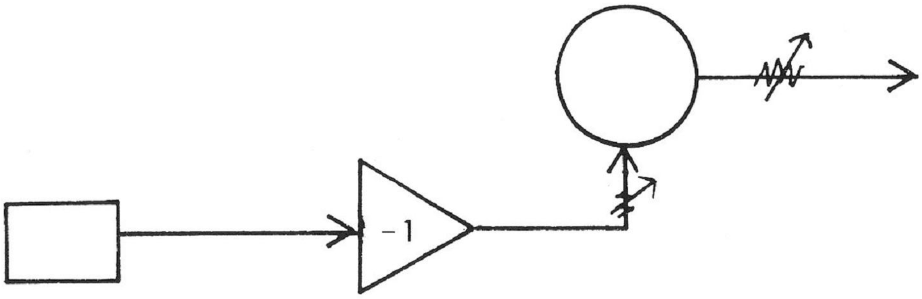
What's happening is that the signal passing through the filter, when mixed with its inversion, cancels out (page 68). This leaves only the high frequencies of the inverted noise coming from the mixer output.

Try this with audio frequencies from a VCO to hear other possibilities.

EXPERIMENT #59: Inverted control voltages

An LF square wave when applied to a control voltage input of a VCO will cause the VCO to alternate frequencies in proportion to the amplitude of the LF wave and the attenuator setting into the VCO. When the square wave is low, you will hear the original VCO frequency; when high, the higher frequency (Experiment 17).

You can invert the LF square wave, which would cause the VCO to alternate between its original frequency and a lower frequency, since the inversion would *cause the LF wave to vary between 0 volts and some negative voltage. Try the patch shown here.*



Any control voltage can be inverted and used to control any voltage-controllable module. However, you'll have to understand what's happening or you may become confused. For example, if you control a low-pass VCF with an inverted LF positive-going sawtooth wave, the VCF will have to be open to start with. If it were closed, then the sawtooth would begin there and gradually close it even more. Since the F_C was sub-audio to begin with, closing the filter even more would not have much result.

Start with different inverted LF waves controlling a VCO, since the changes will be most readily perceived there; then go to the VCF, VCA, pulse width modulation, or any other voltage-controllable module you have available.

D. Lag Processor (LP)

A lag processor modifies (causes changes in) both audio and sub-audio (control) signals. When an audio signal is input, a lag processor acts as a filter (which *may* be voltage-controlled) whose slope is −6 db/octave and whose range of F_C's typically varies from a maximum of about 1600 Hz. to a minimum of some low frequency. Any audio signal input will immediately have its higher harmonics attenuated at a rate of −6 db/octave, a much shallower slope than the −24 db/octave slope of a typical low-pass VCF (see Figure 5-7, and Experiment #A-5 in Appendix A).

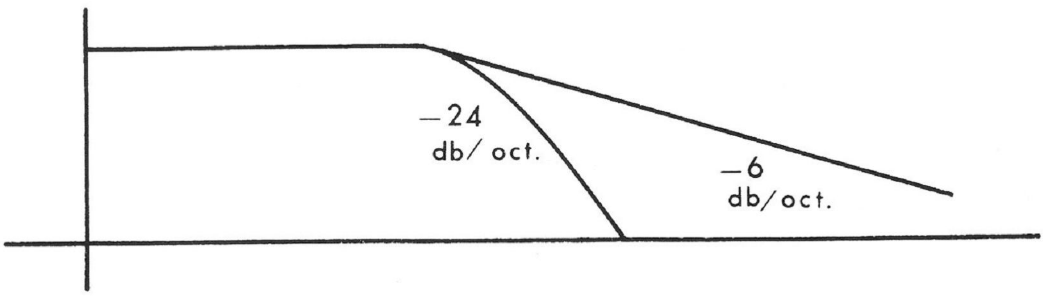


FIG. 5-7

It is the shallowness of its slope that makes the lag processor a viable processor of sub-audio signals (better than a deep-sloped VCF), because it will still pass LF signals relatively strongly. When an LF signal is input, the processor slows down all sudden changes, creating a different shaped LF wave, which will yield different results as a control voltage. Figure 5-8 shows the lag processor's effect on LF pulse and positive-going sawtooth waves. Because sine and triangle waves have almost no harmonics (in their AF) or sudden changes (in their LF), you would generally use a lag processor with pulse and sawtooth waves, or with any control voltage that has a vertical (instantaneous) component (like a gate).

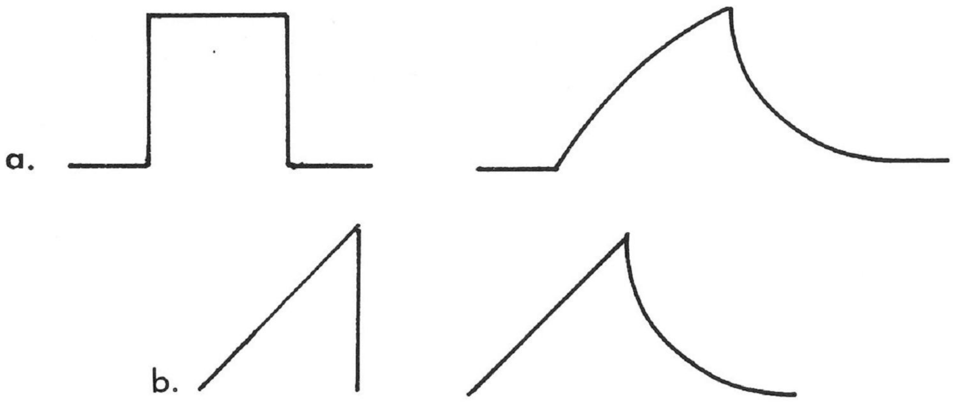
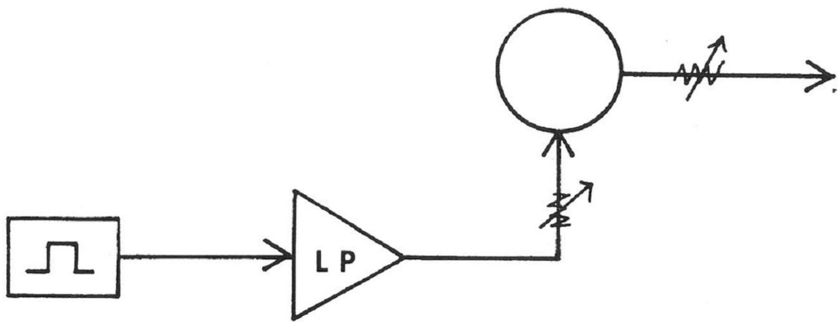


FIG. 5-8

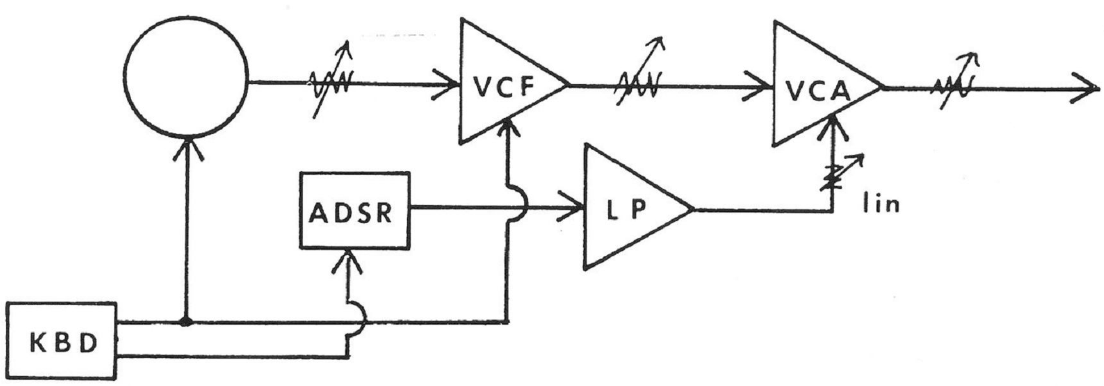
EXPERIMENT #60: Lagged control voltages



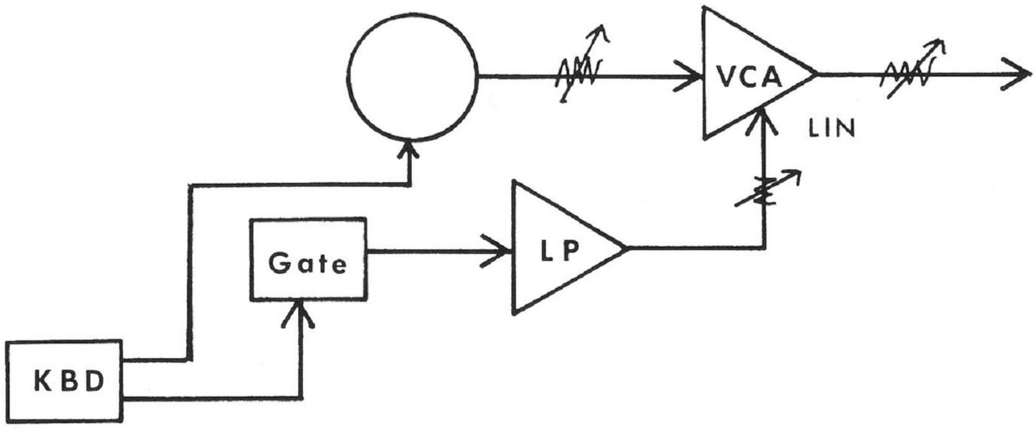
An LF square wave controlling a VCO will cause the VCO to change pitches instantaneously. In the above patch the pitch change will be the same, but the time taken will not be instantaneous. The more the LP attenuator is open, the more pronounced the time lag will be. Figure 5-8a shows what's happening to the LF wave controlling the VCO and accounts for the difference in rate of pitch change.

EXPERIMENT #61: Lag processor applications

The first patch shown will cause any sudden changes in the envelope to be lagged. This is another way to create different envelopes in real time. Instead of changing ADSR attenuators in live performance, just open the LP attenuator.



Say you need something like an AR but no envelope generator is available (they are busy elsewhere in the patch you're working on). If you patch the keyboard gate to the lag processor, the sudden changes as the gate rises to +10 volts and *falls from it will be slowed; the effect is to make an AR control voltage (see Figure 5-8a) without using your AR module.*



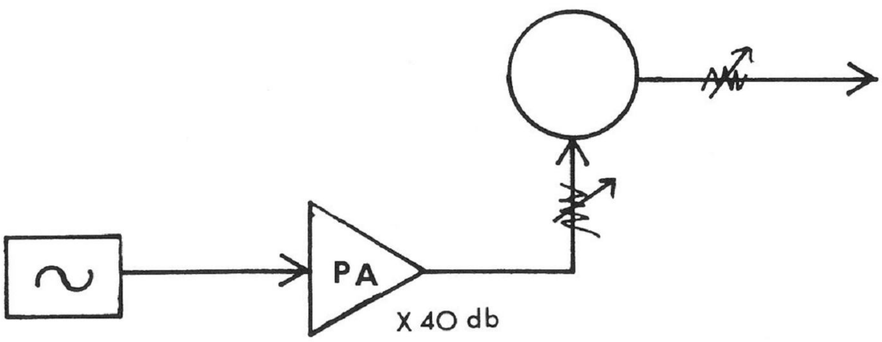
E. Pre-Amp (PA)

A pre-amp is the only module that normally amplifies on a synthesizer (a VCA really being a voltage-controlled attenuator—see page 66). Although most synthesizer modules operate in ranges of voltages, many external instruments that you may want to interface with synthesizers operate in terms of millivoltages (thousandths of a volt). Their output needs to be boosted before they can work well with a synthesizer: thus the pre-amp.

A pre-amp typically has a range switch associated with it which determines whether the incoming signal will be multiplied 100, 10,000, or 1,000,000 times (boosted 20, 40, or 60 db). Within those ranges there is an attenuator which will allow more or less amplification of the external signal. If you don't know the output of the external signal, it's best to start in the lower ranges and work your way up; that way you won't run the risk of blasting and distorting if you put a strong signal into the pre-amp input.

Although the pre-amp will typically be used with external microphones or instruments, there is no reason it can't be used with a synthesizer's own internal signals as an input, particularly if the signal input is a control signal.

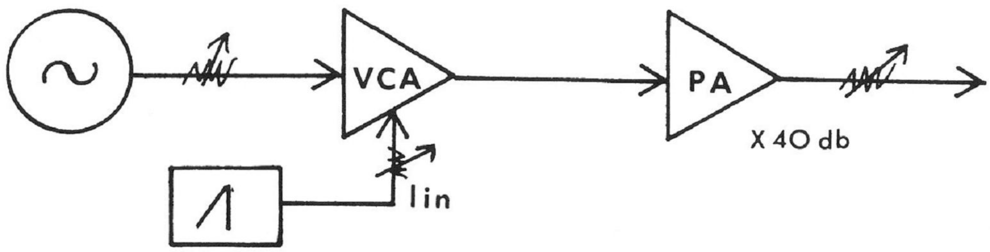
EXPERIMENT #62: The pre-amp as control voltage processor



In the simple patch shown here the sine wave will be amplified until it begins to distort; more distortion will make it more and more a square wave (see, for instance, diagram c in Experiment #30). Using it to control a VCO will give a trill instead of the vibrato you would expect from an LF sine wave.

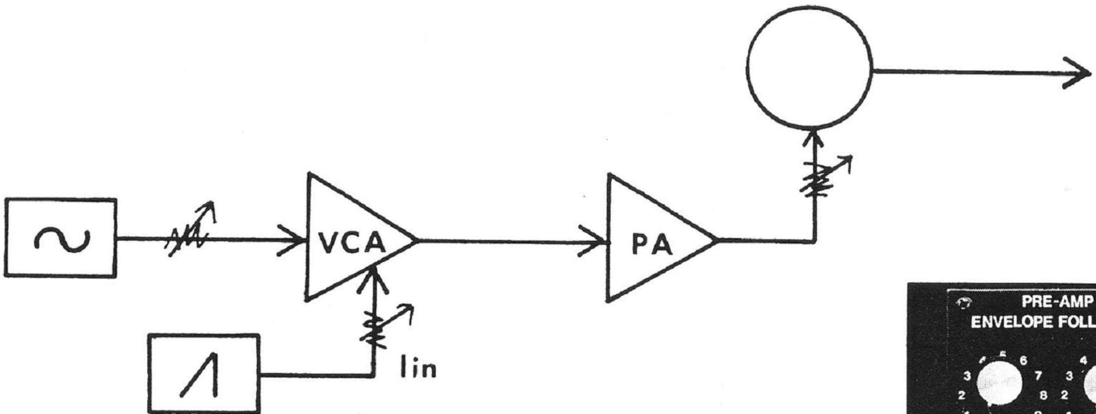
EXPERIMENT #63: Voltage-controlled distortion #2

We heard in Experiment 30 one example of voltage-controlled distortion. Here are two others, this time using the pre-amp. In the first diagram the VCA is closed when the LF sawtooth is at 0 volts.



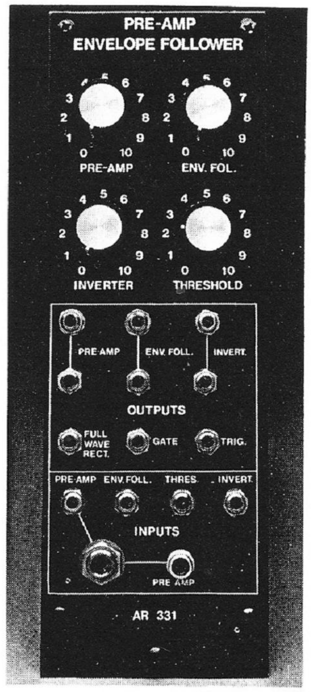
As it rises, the VCA opens and the AF sine wave is allowed to pass to the pre-amp. After a certain point the sine wave becomes more and more clipped, giving voltage-controlled distortion.

In the second diagram the LF sine is gradually distorted under control of the LF sawtooth, and the output used to control a voltage-controllable parameter. You might want to try diagramming the interaction between the sine and sawtooth waves where the sawtooth has a lower frequency, the same frequency, and a higher frequency than the sine wave, using the techniques shown on page 18. Note also the inputting of an LF wave into an audio input to the VCA.



Typically, however, you will use the pre-amp with an external signal in one of two ways: the synthesizer will modify the external signal, or the signal will control one or more synthesizer modules via a derived control voltage. We will deal here with the first of these, since the second will require the envelope follower, discussed later in the chapter.

When you use the synthesizer to change the perceived sound of an input signal, treat that signal exactly as if it were the output of a VCO. In other words, if you want to modify the timbre, input the signal into an audio input to the VCF (Figure 5-9a); to change the amplitude or shape, use an audio input to the VCA (Figure 5-9b) together with an envelope generator. Remember when we discuss ring modulation that an external signal can be an input. Since I don't know what, if any, external signals you have available for synthesizer processing, you will have to experiment for yourself here.



Aries AR-331 Pre-amp/ Envelope Follower

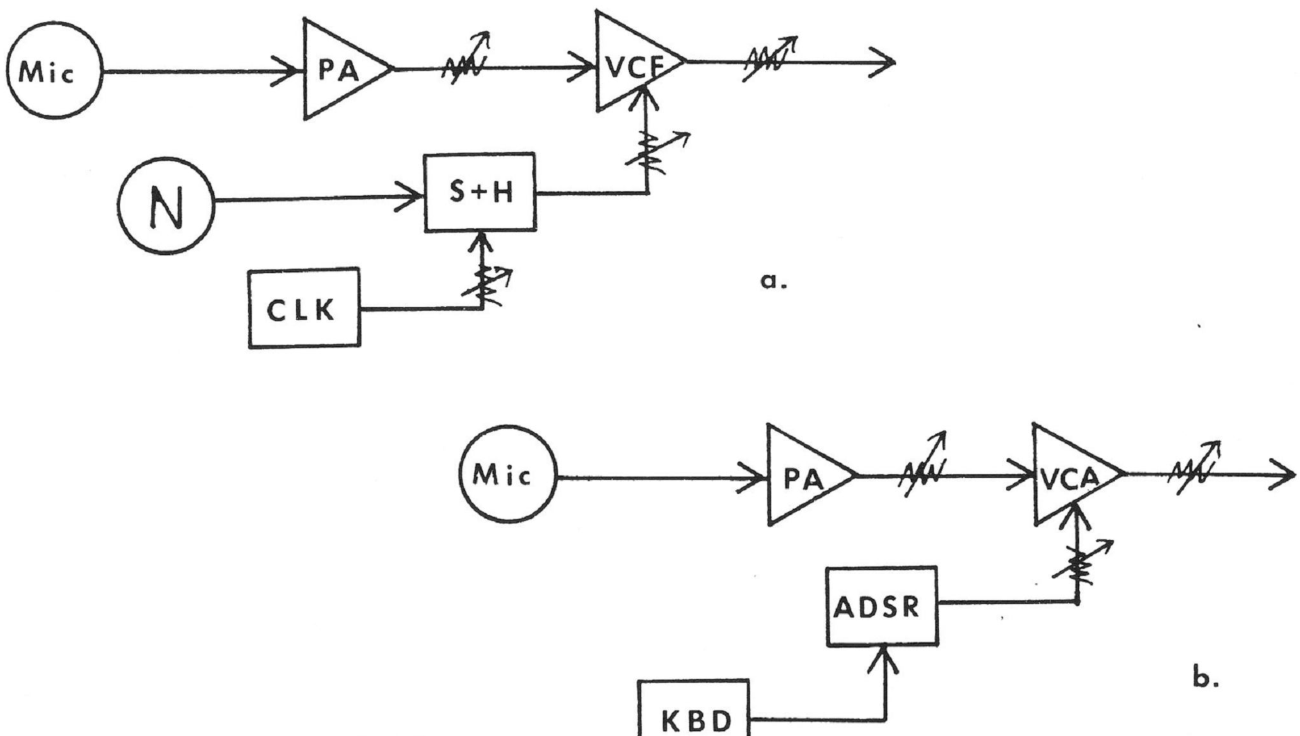


FIG. 5-9

F. Multiple

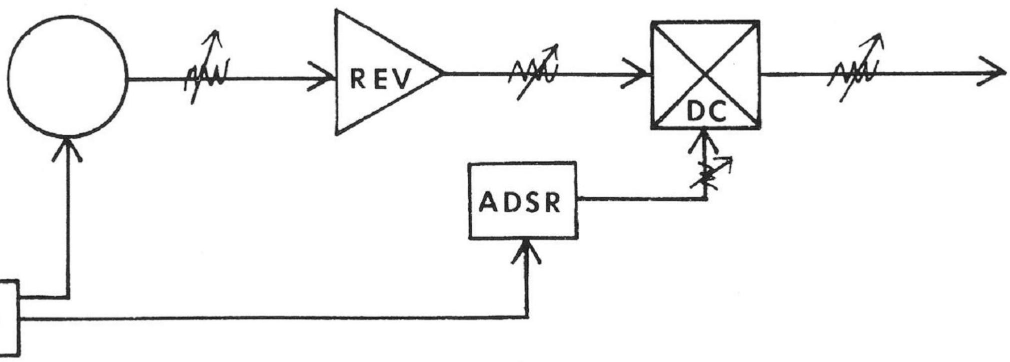
We have already used a multiple, a series of jacks wired in parallel, on many occasions. Use the multiple any time you need to route one output to more than one input and pre-patches can't help. Once an output is patched into any multiple jack, that output is available at all other multiple jacks. Do not use a multiple as a mixer.

G. Reverberation

Reverb is used to give music a more ambient feeling that, for various reasons, might not otherwise be there. Generally it is best used sparingly.

Typically you might use reverb as the last signal processor before the signal is output (e.g., Chapter 4, section I). Just open the reverb attenuators to taste. However, a reverb can also be used as an intermediate signal processor in a chain of processors.

EXPERIMENT #64: Reverb as an intermediate signal processor



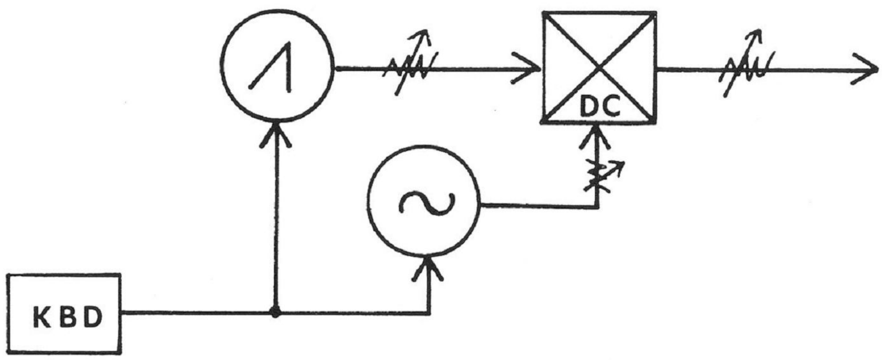
As you play successive keys, the reverb will hold some of the old pitch, even after you play a key with a new pitch. The reverb "stores" the sound. Even if you play the same key over and over again, certain harmonic changes will take place, yielding a qualitatively different sound than if reverb were not in the patch.

H. Ring Modulator (RM)[3]

A ring modulator, so called because early versions had a configuration of diodes within the module in the shape of a ring, is a voltage multiplier[4] (as is the VCA) which will have an output with *negative* control voltage as well as positive. There must be signal present at each of two inputs for there to be an output. As long as both inputs have some voltage *other than 0,* there will be an output; if either input has 0 voltage for an instant, then for that instant there will be no ring mod output.

When both inputs are AF waves, the ring mod multiplies the fundamentals and harmonics of the input waves and outputs the sums and differences of all their frequency components. The input signals themselves are suppressed. (A more detailed description of how the ring mod does this multiplication of frequency components will be given in Chapter 6, when amplitude modulation—of which ring modulation is a form—is discussed.) This results in the typical "clangorous" ring mod sound.

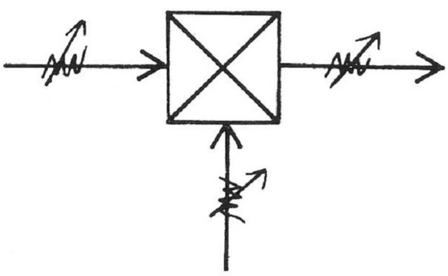
EXPERIMENT #65: The ring modulator



Experiment with different sound textures and densities by varying the frequencies of the two VCOs. If the RM on your synthesizer has a switch indicating "AC" (called "Audio" on the ARP 2600) or "DC" coupling, the switch should be in the DC position. Play the keyboard.

If you would like to hear the texture a little less dense, use a square wave (which has only odd harmonics) instead of a sawtooth.

[3] A ring mod will be designated by this symbol in block diagrams:



[4] The equation to determine output voltage is: $V_{out} = V_x \times V_y / 5$.

The simplest RM output you could have, where both inputs are AF, would be if two sine waves were input. You can get one sine wave by putting a VCF in oscillation (see Experiment #27) and use that and an AF sine wave from a VCO as the other. Since the sine waves have no harmonics, all you'll hear will be the sum and difference frequencies of the sine waves themselves.

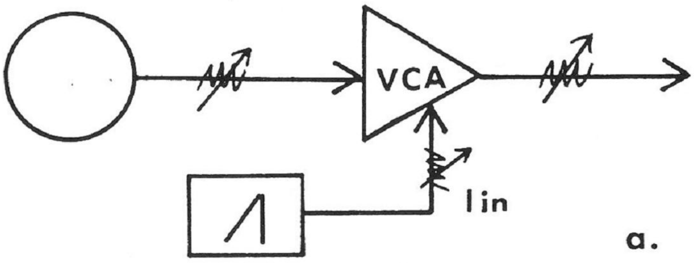
A dense RM output would result if you used two AF sawtooths as your inputs.

The patch shown in Experiment #65 is the standard RM sound. However, if you have access to its inputs, the RM is a *very* versatile module; producing clangorous sounds is just one of the various functions it can serve.

One of the most useful of these functions is that the RM can be a VCA. When both inputs are AF (as in Experiment #65), a modulation timbre is perceived as the output; when one input to the ring mod is LF and the other is AF, an audio frequency being amplitude-modulated by a control voltage (like a VCA) is perceived. Just treat one RM input as an audio input and the other as a control; it doesn't matter which is which. The ring mod should be "direct coupled" when being used as a VCA; any associated switch should be in the DC position.

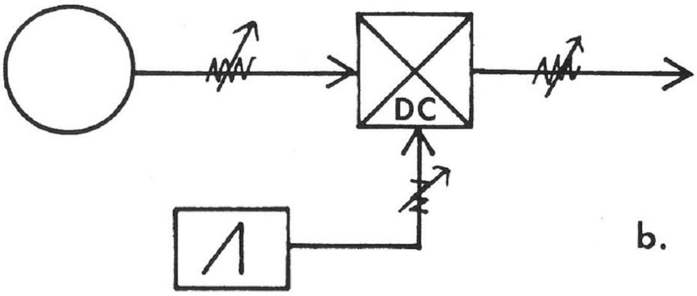
EXPERIMENT #66: The ring mod as VCA

Set up the patch shown in diagram a. Review the effect of an LF sawtooth controlling a VCA.



a.

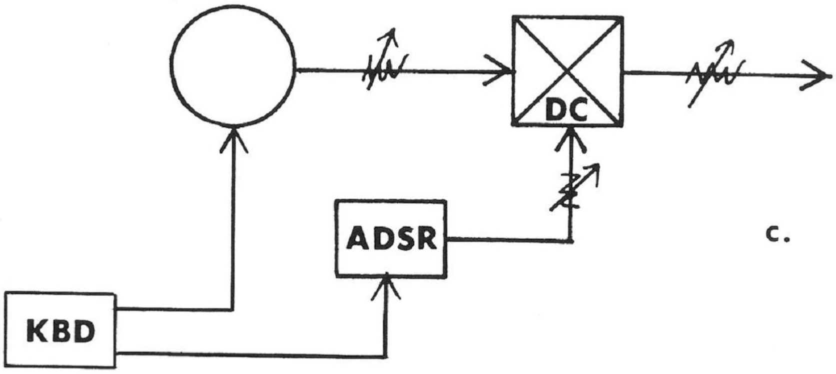
Instead of having both oscillators as inputs to the VCA, have them both go into a direct-coupled ring mod, as in diagram b.



b.

There should be no difference in the quality of the sound output between the VCA and the RM. When an LF wave that has a DC component (like a positive-going 0 to +10 volt sawtooth) is one input to the ring mod, and an AF signal is the other input, the ring mod will act exactly like the VCA. You can think of the LF wave as a source of "control" voltage, although with the ring mod it's just as accurate to say that the AF signal is "control" (!).

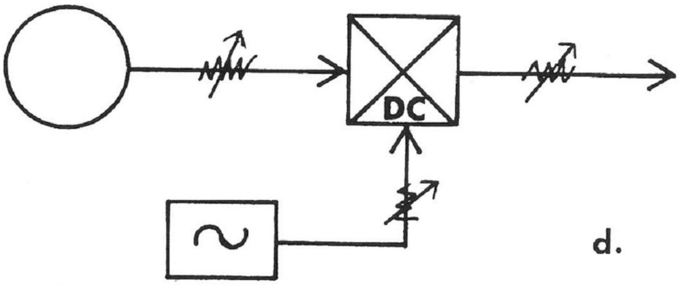
Now change the LF wave to an envelope generator output, as in diagram c. Play the keyboard.



c.

The ring mod still outputs the same as if it were a VCA. The envelope generator output is the equivalent of an AC wave with a DC component (see, e.g., Fig. 5-13). It constantly varies between 0 and +10 volts. So long as the "control" voltage has a DC component, the ring mod will function as a VCA.

Substitute for the envelope generator an LF sine wave, as in diagram d. Before listening to the output, think about what you would expect it to sound like.



d.

You might expect the volume of the AF signal input to the ring mod to rise and fall during the positive half of the sine wave's cycle and to output nothing during the negative half. After all, that's what the VCA would do (see Chapter 3, section III-B).

Now listen to the ring mod output. The sound rises and falls in volume but stops altogether only when the sine wave is at 0 voltage.

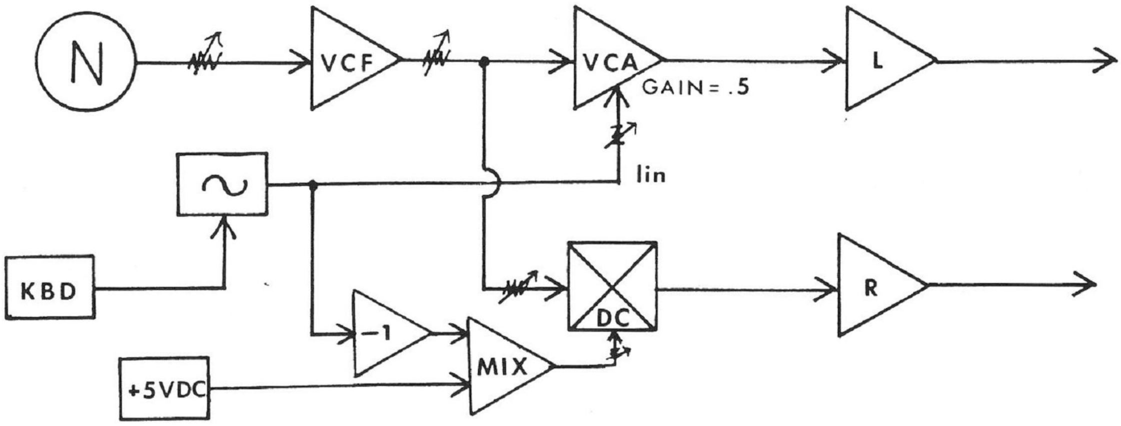
The ring mod, unlike the VCA, will output the AF signal even if the "control voltage" is negative. The ring mod is a *four-quadrant multiplier* (see section III-B in Chapter 3). It will not only output if the control voltage has a net positive value, it will also output if the control voltage has a net negative value. The only time the ring mod doesn't output is when one of the inputs is at 0 voltage. Thus you hear the AF signal at all points in the sweep of the sine wave except when it hits 0 volts. Note that the sine wave is at 0 volts *twice* every cycle.

Having two VCAs makes some wonderful effects possible, as Experiment #67 will show.

EXPERIMENT #67: Voltage-controlled stereo pan

With this patch the audio signal will shift from one side of the stereo field to the other, at whatever speed you want. In addition you can voltage-control it, so that a higher voltage (e.g., playing a higher key) will speed up the pan, a lower one slow it down.

The NG attenuators are opened, noise gets to an audio input to the VCA and out the left output. The gain attenuator of the VCA is halfway open, allowing a bias of +5 volts into the CV mixer of the VCA. The audio signal (noise) is split to one input of the ring mod as well.



The critical signal in this patch, however, is the LF sine wave. It is also split, controlling the VCA and going through an inverter to the ring mod. To do this, you will need to put the LF sine wave through a multiple. The ring mod is DC. Keyboard control of the LF sine wave is patched from the KBD CV output to an unattenuated input of the LFO.

Let's analyze the patch. The LF sine wave will raise and lower the control voltage in the VCA's CV mixer from +5 to −5 volts, but since there are already +5 volts there from the gain attenuator, the VCA will go from +10 to 0 volts; thus it will open all the way and close, but it will not be silent for half the time as it would be if the LF sine wave had not been biased up by the +5 volts from the gain attenuator.

The LF sine wave is inverted, so when it is high (i.e., when it is at +5 volts and the VCA is wide open) the ring mod will be receiving −5 volts; this would normally allow signal to pass through the ring mod, since it passes negative as well as positive control voltage. However, the ring mod has a +5 volt bias added to it, so the net voltage reaching the ring mod when the VCA is wide open is 0. At 0 voltage the ring mod outputs nothing.

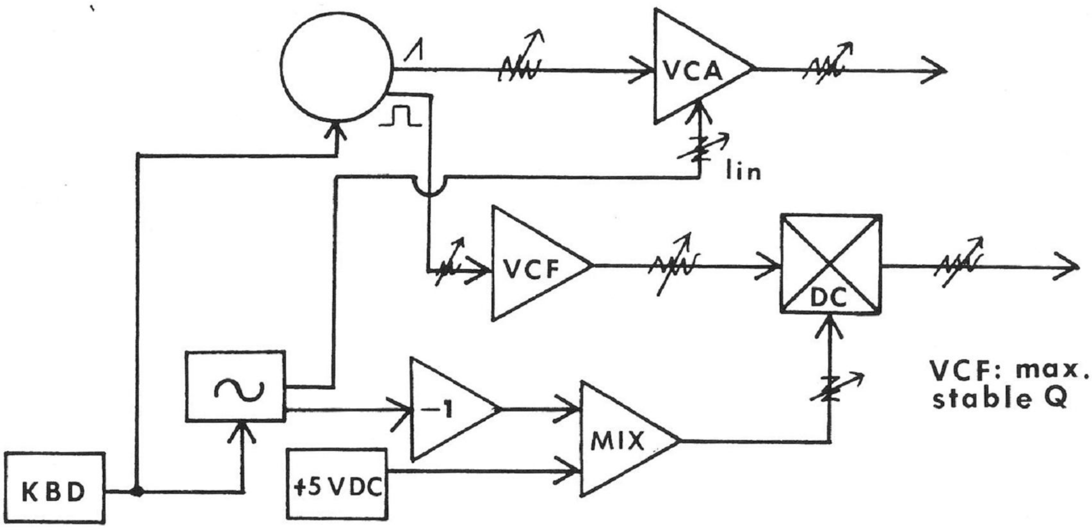
Conversely, as the LF sine wave reverses itself, going through a 180° phase shift, the VCA will close (sine wave goes to −5 volts offset by the +5 volts from the gain attenuator) and the ring mod will have maximum output (+5 volts from the sine wave plus (+5) volts from the DC offset). Thus as the VCA closes, the ring mod opens. During intermediate points in the cycle both the VCA and the ring mod will be opened or closed to varying degrees.

Since the ring mod is patched to the right side of the stereo field and the VCA to the left, they will create a stereo pan effect. Adding keyboard control of the LFO allows the stereo pan to be voltage-controlled, faster as you play higher (and increase the frequency of the LFO), slower as you play lower on the keyboard.

This general patch, using an LF sine wave to control a VCA and its inversion to control a ring mod (or another VCA), is *extremely* useful anytime you want to pan from something to something else.

EXPERIMENT #68: Voltage-controlled mixer

This patch is quite similar to the previous patch, except that the audio signal has changed. Treating the ring mod as a VCA, you have two VCAs input to a mixer with control over the individual amplitudes of each signal, in this case a voltage-controlled cross-fade between the sawtooth and the square wave.

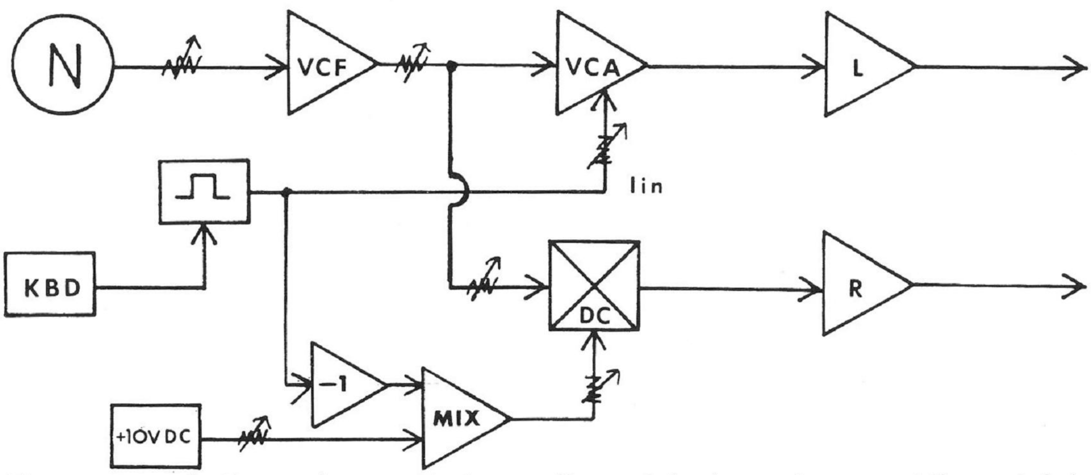


If you listen to them separately, you can't hear that their amplitudes are varying inversely, but the LF sine wave increases one as the other decreases at a rate that is voltage-controllable. The cross-fade is between two waveforms of the same VCO.

The square wave is input into a filter that is highly resonant—just short of maximum stable resonance. This is only so that you can better distinguish the square wave from the sawtooth when listening.

With some slight variations of the patch in Experiment #67, the ring mod can function as an electronic switch.

EXPERIMENT #69: The ring mod as electronic switch

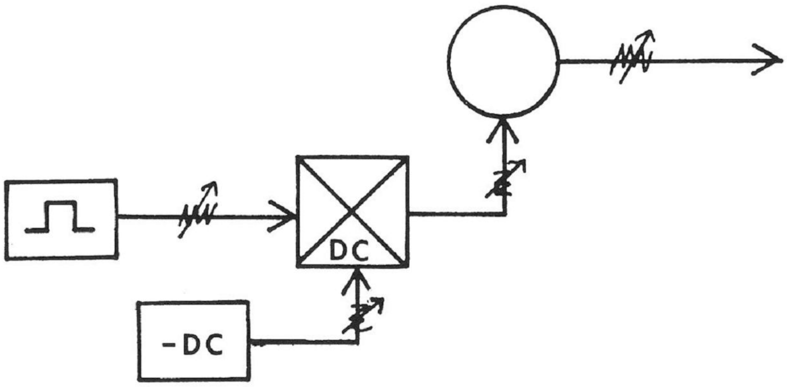


There are only three changes between the patch shown here and the patch in Experiment #67. Since to get an electronic switch effect we want a sudden switching instead of a gradual cross-fade, use an LF square wave instead of a sine wave. Since it will send the VCA from 0 to +10 volts, no initial gain is required; close that attenuator. When the voltage is 0 the RM outputs nothing, but at that instant the pulse at the VCA goes to +10 volts, so the VCA outputs and the RM shuts down. The converse is true and when the RM goes to +10 volts the VCA control voltage is 0 volts. Changing the pulse width of the LF square wave will result in more time spent on either the VCA or the ring mod.

You could also try this with an LF sawtooth controlling the VCA and an inverted and biased sawtooth controlling the ring mod.

Because of its ability to pass negative voltage, the ring mod can function as an inverter. Just input whatever signal you want inverted and a steady negative DC voltage.

EXPERIMENT #70: The ring mod as inverter



The patch is quite simple. Normally an LF square wave controlling a VCO would make it rise to a different frequency and fall to its original frequency at a rate determined by the frequency of the LF wave and to a depth dependent upon the level of the CV attenuator into the VCO. Those are true here also, but because the LF wave is multiplied by a constant negative voltage, its functions as an inverted LF wave, lowering the frequency of the AF VCO. Naturally you can use the ring mod as an inverter in any situation; just input negative DC voltage as shown in this diagram.

I. Balanced Modulator

Some ring modulators (e.g., Emu, ARP 2600) have a switch indicating either "AC" (audio) or "DC" coupling; that switch is a complex device. Any wave can be considered to be an AC wave with a DC component[5] (page 125). Here's an example of what that means. You know that an LF pulse from a VCO output generally varies from 0 to +10 volts. That pulse wave is the same as an AC pulse which has had added to it a bias of +5 volts. Figure 5-10a shows a typical pulse wave; 5-10b shows an AC pulse wave as explained in section III-A of this chapter. If we add to that AC pulse wave a bias of +5 volts, we have as a net result a typical positively offset pulse wave (as in Figure 5-10a); we have raised (biased) it up +5 volts. Most unmodified pulse waves generated by synthesizers are AC pulse waves *and* a bias (DC component) of +5 volts. Similarly most unmodified sawtooth waves generated by synthesizer VCOs are AC sawtooth waves *and* a bias of +5 volts.

When a ring modulator is A–C, the DC component of the wave is removed. The effect is to make a balanced AC wave out of the originally input wave. Sometimes you may hear a ring modulator referred to as a "balanced modulator"; actually it is best said to be a balanced modulator when it is AC (acoustically coupled), the steady-state DC component is removed from any input wave, and a "balanced" AC wave results therefrom. If the

[5] If the wave varies from, say, +5 volts to −5 volts, it has a DC component of 0 volts.

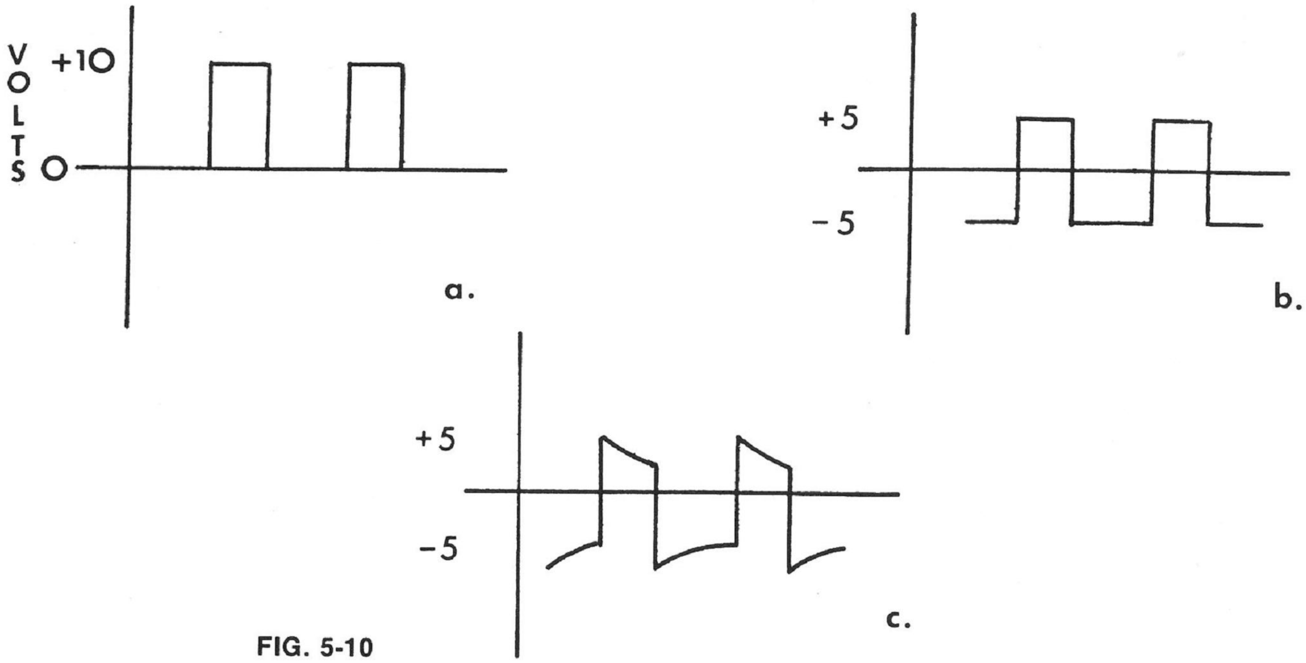
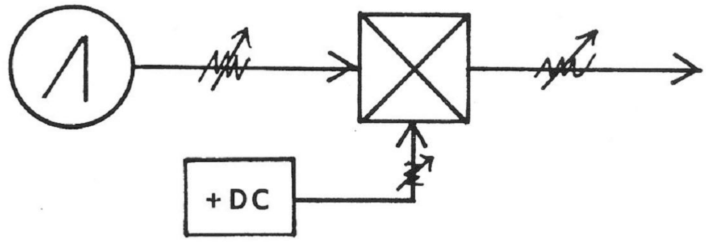


FIG. 5-10

module on your synthesizer is called a "balanced modulator," it probably is permanently AC-coupled and always removes DC components from input signals.[6]

EXPERIMENT #71: Balanced modulator



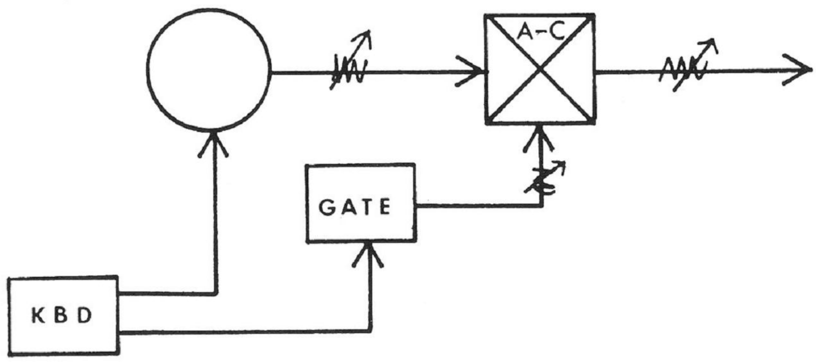
This brief experiment works if you have either a balanced modulator or a ring modulator with a switch that allows AC-coupling.

An AF sawtooth and positive DC are the two inputs. When a ring modulator is DC (direct-coupled), the AF sawtooth is output. On a balanced modulator, one with AC-coupling, the steady-state DC component is removed; thus one input has a net voltage of 0 and there will be no output.

Behind some ring or balanced modulator inputs are capacitors which have effect only when the modulator is AC-coupled. The two capacitors (one behind each input) are the primary electronic parts responsible for the removal of the DC component of any input. They do not necessarily remove the DC components either instantaneously or simultaneously; the time they take to remove the DC component is called the "time constant" of the capacitor. Sometimes the time constants of the two capacitors will differ; this means that the DC component of one input will be removed at a different rate of time than the other, depending upon whether a wave with a DC component is input to one jack or the other.

[6] Manufacturer nomenclature about ring- or balanced-modulators is somewhat confusing. Their important similarity is that they are both four-quadrant multipliers. Aries' balanced modulator is only direct-coupled; the ring modulator on the ARP 2600 can be either, depending on the setting of a switch. If in doubt, input a positive DC signal into the module; if there is an output then the module is DC (see Experiment #71).

EXPERIMENT #72: Differences in balanced mod inputs



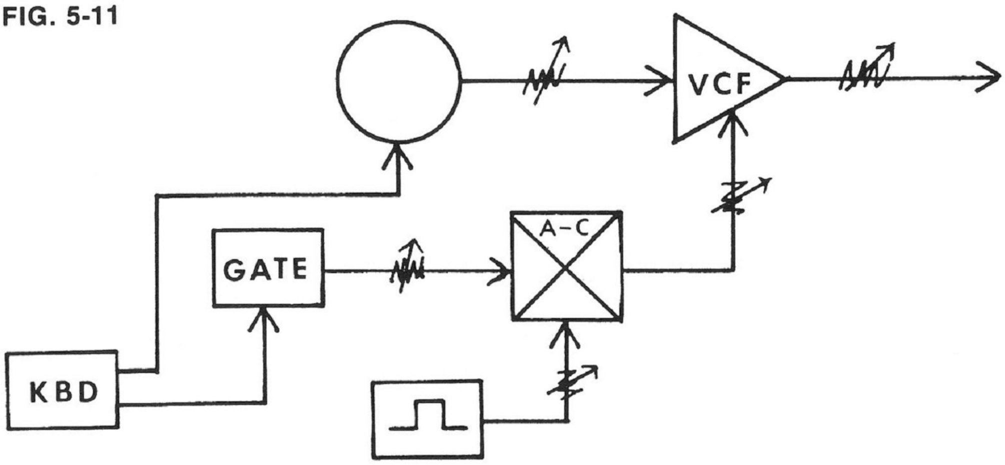
First input the AF sawtooth into one input of the balanced mod and the gate to the other input. Play one note and listen to the decay of the note. It takes a certain time. Now reverse the inputs, so that the AF sawtooth is input into the other jack. Play one note. The decay of the second note may be different from the decay of the first, since the time constant of the capacitor behind one jack is frequently different from that of the capacitor behind the other jack.

Let's review what's happening in Experiment #72. A keyboard gate signal looks just like an LF pulse wave (see, for example, Figure 3-20). When you depressed a key in Experiment #72, the keyboard gate was sent to one input of a balanced modulator. The keyboard gate is DC (Figure 5-10a); the first thing the balanced mod circuitry does is remove the DC component of the gate, making it balanced AC (Figure 5-10b). However, as you hold the key down, the balanced mod still detects DC, the steady + or − 5 volts being held (shown in Figure 5-10b). At that point the capacitors remove the DC at a rate determined by their time constants, so the keyboard gate envelope becomes like that in Figure 5-10c.

Did you notice that the balanced mod output not only when you depressed a key but also when you released the key? If not, go back and observe that this is true. The reason for this is that once the LF pulse (the keyboard gate) becomes AC-balanced, the balanced mod treats both positive and negative halves equally, *each* instantaneous voltage change (*each* depression *and* release of a key) yielding a voltage that because of the time constant, has an instantaneous attack and a slight decay. Thus it is possible to use the balanced mod as a quasi-envelope generator. Think of the envelope created by this "envelope generator" as having a fixed attack and decay—the time constant of the capacitor behind whichever input you are using.

Of course, a balanced mod may be used with two LF waves as inputs to serve as a further processor for a control voltage (see Figure 5-11).

FIG. 5-11



When using a ring mod, it's a good idea to experiment with the switch (if yours has one) in both the AC and DC positions, as well as to reverse the inputs. You may get four different effects, and one may be more suitable for your needs than the others.

We have seen that the ring mod can be a modifier whose output is clangorous tones; it can function as a VCA, an inverter, an electronic switch, and a quasi-envelope generator. You will probably find other uses for it as well. It is possibly the most versatile module on a standard synthesizer.

III. CONTROLLERS

A. DC Voltage[7]

Any voltage that does not alternate between two voltage levels (for example, a steady +4 volts or a steady −6 volts) is DC. Many patchable synthesizers make available sources of DC voltage, typically with an associated attenuator, so that the user has available sources of either positive or negative DC voltage from 0 to ±10 volts (see section VIII in Appendix A).[8] The most common sources of DC voltage are the coarse frequency attenuators associated with VCOs and VCFs, and the gain attenuator associated with a VCA.

Since the output of voltage-controlled modules on a synthesizer varies in some proportion to the input control voltage, having these positive and negative voltages available widens your possibilities of control. For example, in Experiment #6 you listened to the harmonics of a sawtooth wave. At the end of that experiment you applied negative DC voltage to the LFO, which had the effect of substantially slowing down the wave being generated by the LFO. Since it was already generating "a very low-frequency sawtooth wave," the negative voltage allowed you the flexibility of slowing the wave down even further. Experiment #10 is another example of this use of DC voltage.

An extremely important function of DC voltage sources is *offsetting* waves, raising or lowering a given parameter of a wave. In Experiment #28 you added +10 volts to an audio input to the VCF, which already had a 20 volt p-p sine wave input. The effect was to change the position of the wave in the VCF, causing clipping at the filter and an attendant timbral change. In Experiment #38 you added a +5 volt offset to the VCA so that the VCA would not shut down when negative voltage was being input by the controlling sine wave. The offset came from the gain attenuator of the VCA. In Experiment #29 you used a negative offset to abate clipping.

Just as amplitude (Experiment #38) and timbre (Experiment #28) can be changed by offsetting, so can frequency be offset with very useful results for live performance.

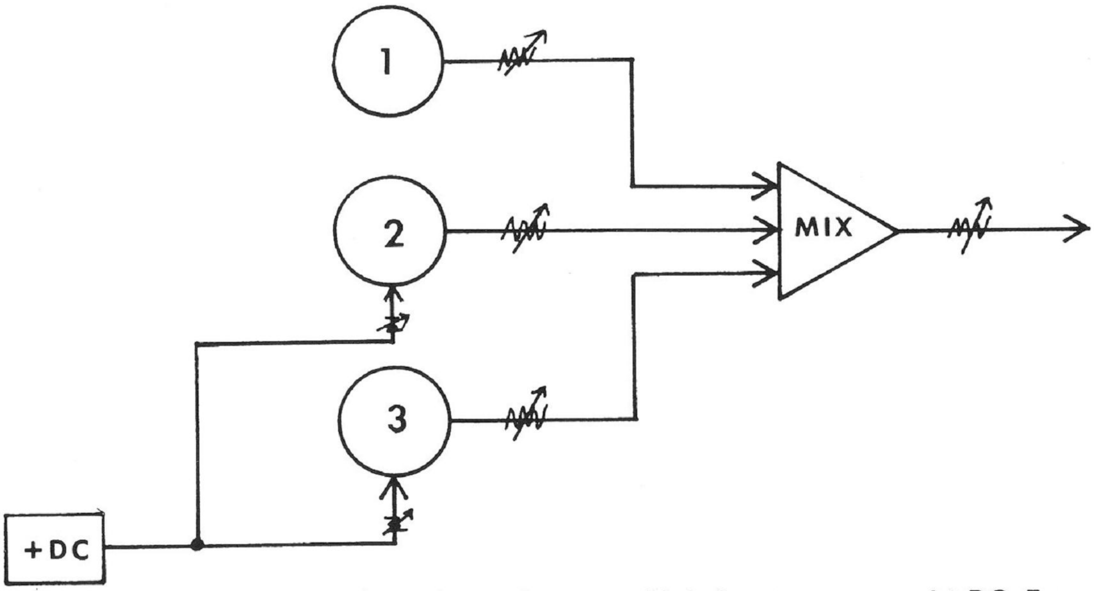
EXPERIMENT #73: Real-time chord changes with a one-voice synthesizer

In this experiment we will set up a patch that will allow you to change from any chord to any other chord almost instantaneously. The chords we'll use will be C major to F major, but they could be any chords.

Tune three VCOs to C, E, and G respectively. We're going to set up frequency offsets such that the C will remain the same, the E will go to F, and the G will go to A simultaneously upon moving the attenuator, allowing +DC to control two VCOs.

[7] Do not confuse DC (direct-current) voltage with a DC (direct-coupled) four-quadrant multiplier.

[8] Other sources of DC voltage include the gate signal from a keyboard, sample and hold, and 0 or 100% duty cycle pulse waves. Any horizontal line in the diagram of a voltage indicates the presence of DC.

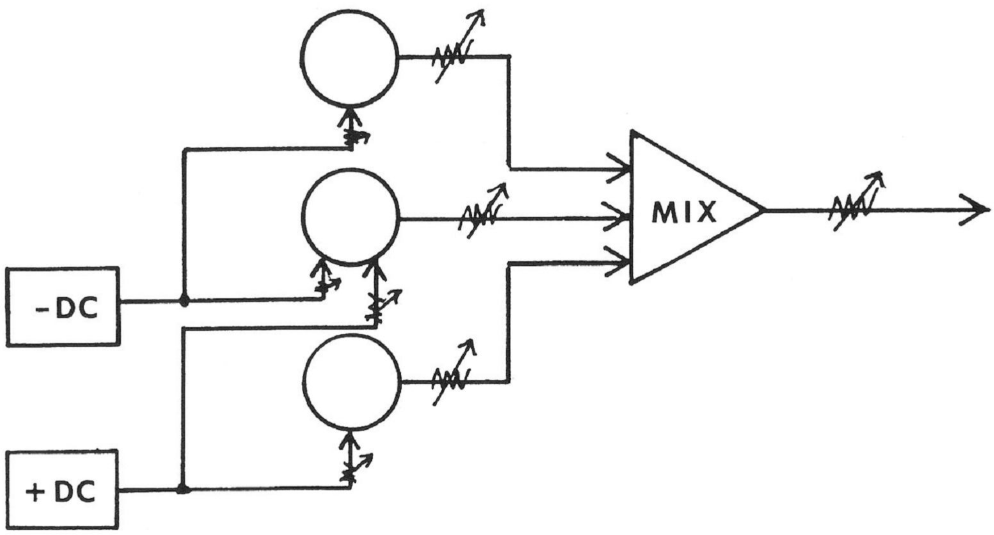


In the first diagram, a patchcord goes into a multiple from a source of +DC. From the multiple come two patchcords going to attenuated CV inputs of VCO-2 and VCO-3. If there is one, open the attenuator associated with the +DC all the way, so that +10 volts DC are available. Listen now only to VCO-2. Slowly raise the attenuator over the CV input to VCO-2. The frequency of VCO-2 slowly rises because you're inputting positive control voltage. Since that input control voltage does not vary, the frequency of VCO-2 will not vary except when you raise the attenuator into VCO-2. Stop raising that attenuator when the pitch of VCO-2 has risen from an E to an F.

Now listen only to VCO-3. Allow positive control voltage to be input until the pitch has risen from a G to an A. Leave both control attenuators where they are. Close the attenuator associated with the +DC. Now raise the three VCO attenuators into the mixer. You hear the C major chord. Open the attenuator associated with the +DC all the way. You hear an F major chord.

Naturally, you could use this method on any two chords where the second is generally higher than the first (because you only have access to positive control voltage). If the second chord varies lower (say from C–E–G to B–D–G), you would follow the same procedure but use negative DC voltage. Try that.

If you have access to two multiples, you can set up the patch shown in the following diagram, which will allow you to shift both up and down: up by opening the attenuator associated with the +DC voltage; down by opening the attenuator associated with the −DC voltage. Of course, opening both would give you still another chord.



Perhaps the most important use of offsetting is in relation to LF control voltages. A wave is AC if the sum of all its instantaneous changes through one cycle equals 0 volts. Thus a sine and triangle wave as generated by many VCOs (varying from +5 to −5 volts) is AC. A wave is DC when there is no variation in voltage as, for example, a constant +5 or −4 volts. *All other waves can be viewed as AC waves with an added DC component and called offset AC waves.* In Experiment 38 you added a DC component (a steady +5 volts) to an AC sine wave to create an offset sine wave. Since the DC voltage was +5 volts, the wave became a *positively* offset sine wave (Figure 5-12a). Five volts was added to every portion of the wave. Had the added voltage been −5 volts, the wave would have been a *negatively* offset sine wave (Figure 5-12b).

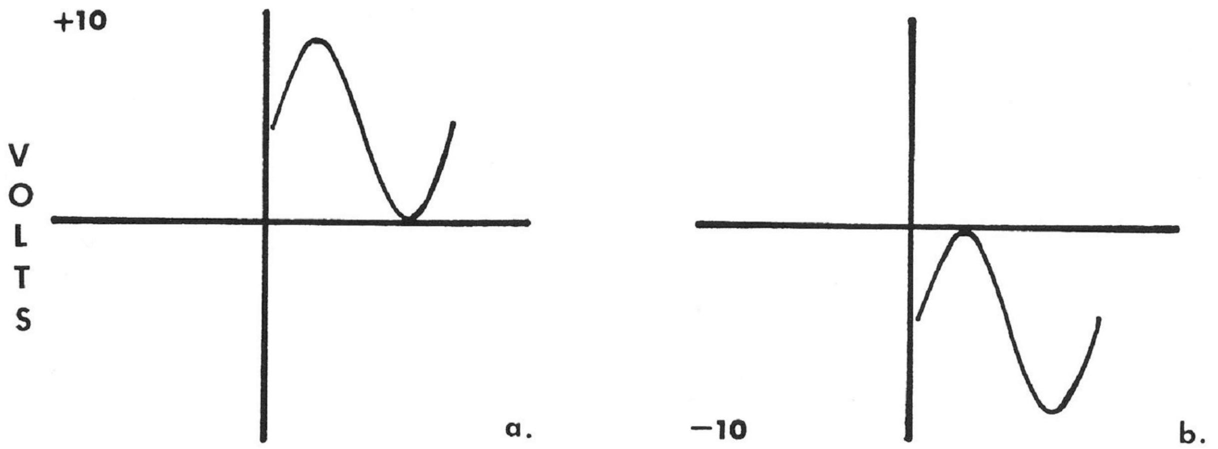
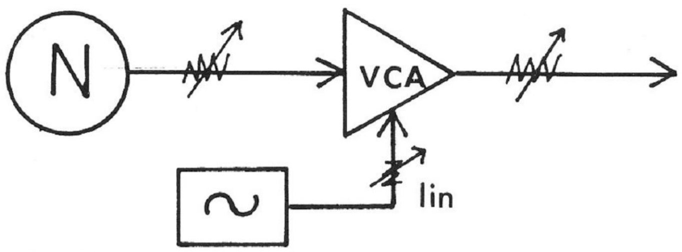


FIG. 5-12

EXPERIMENT #74: Creation of a positive offset

If you were to control a VCA with an AC sine wave, the VCA would have no output for half the time—the time the sine wave was negative. However, if the sine wave fluctuated from just above 0 volts to +10 volts, the VCA would output the audio signal all the time.

First set up the following patch and verify for yourself that in this configuration the VCA outputs signal only half the time.



Now open the gain attenuator associated with the VCA halfway. Typically this attenuator makes available a maximum of +10 volts, so opening it halfway makes available +5 VDC. As Figure 3-10 shows, the voltage from the gain attenuator is mixed with any other control voltage input to the VCA. Thus the LF sine wave voltage is being mixed with a steady +5 VDC. You will hear the audio signal input to the VCA more and more as the LF sine wave goes from its original −5 to +5 volts to a more absolutely positive voltage sweep (i.e., 0 to +10 volts). When you have mixed the LF sine wave with +5 volts DC, the wave will fluctuate from a bit more than 0 volts to +10 volts and there will be no silent moments from the VCA. You have offset the LF sine wave with +5 volts DC, creating an LF wave such as that in Figure 5-12a.

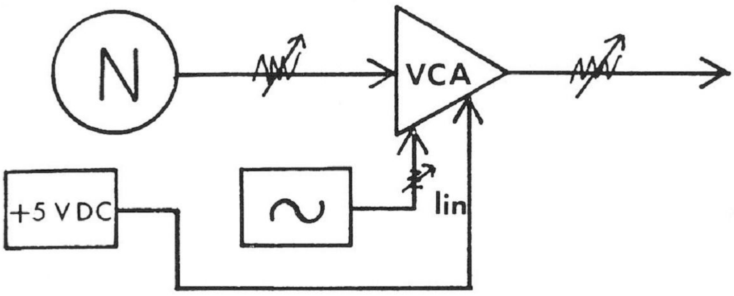
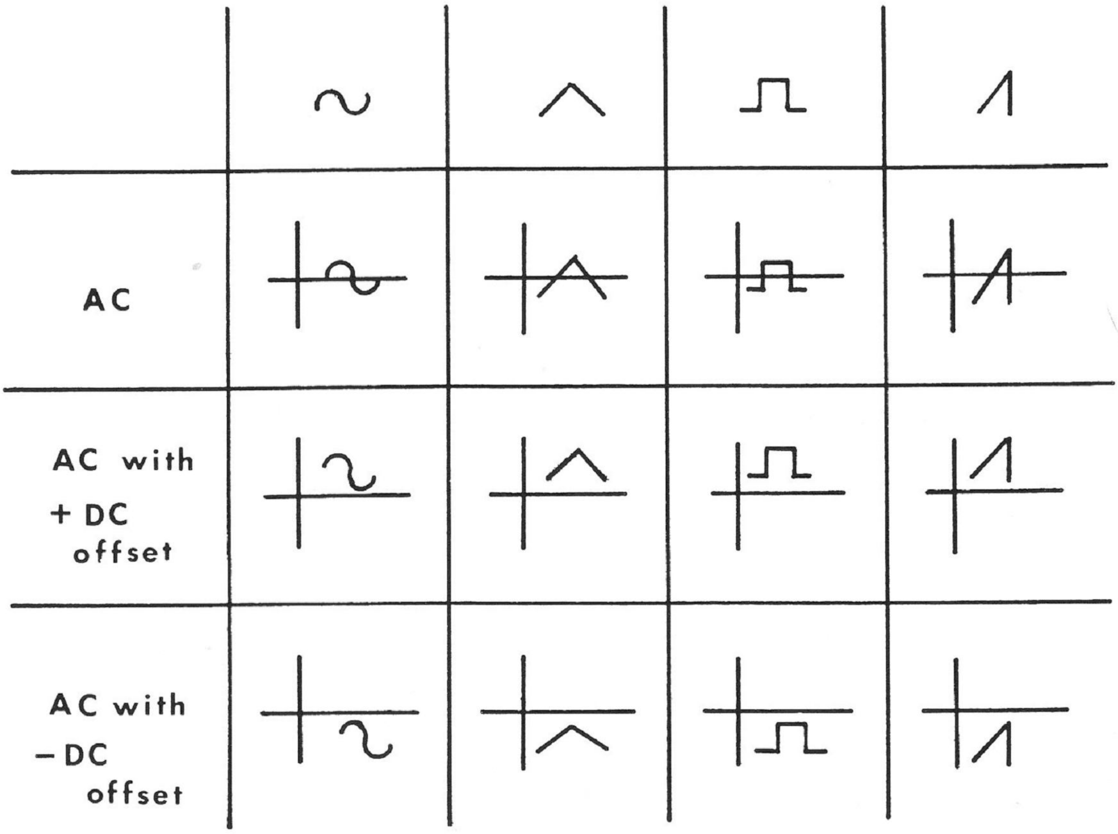


Table 5.1 shows some possibilities for offsetting waves. You might try creating some of these and using them to control other voltage-controllable modules.

TABLE 5-1



B. The Keyboard

A keyboard is a separate module, just as a VCO or VCF is a separate module. Virtually all of its associated controls make live performance in real time easier and more exciting.

The first thing to be said about synthesizer keyboards is that they are voltage-controllers; as such they can control *any* voltage-controllable module. Of course the most common use is to control a VCO at a rate of 1 volt/octave, but a keyboard could also control the opening of a VCA, pulse width modulation, or anything else.

One-voice synthesizers (like a Minimoog, ARP AXXE, Yamaha CS-5, and KORG MS-10) have keyboards that have either low- or high-note priority. If the keyboard is low-note priority, the lowest note played determines what voltage gets from the keyboard to the module being controlled. If the keyboard controls a VCO and you play a C major triad (C–E–G), only the C will be sounded by the VCO, because only one control voltage—that from the lowest note being played—is output by the keyboard. If your keyboard is high-note priority, the VCO would sound only the G. This is why one-voice synthesizers play only one note at a time; they output only one control voltage at any given instant. (Of course, that control voltage may be routed to many VCOs, sounding many pitches simultaneously.)

Typically one or more keyboard control voltages are pre-patched to VCOs and the VCF; in addition the keyboard control voltage may be available at an output jack on the synthesizer and may be patched anywhere. The keyboard also outputs timing signals, a gate, and (for all synthesizers except Moog) a trigger, each time a key is depressed; these are typically pre-patched to the envelope generators and are available at gate and trigger output jacks on quasi-modular and modular synthesizers.

With *portamento* (sometimes called "glide") you can make notes glide to one another instead of going in discrete steps. Typically when you go from playing one key to another and another, the instantaneous frequencies created look like those in Figure 5-14a; when portamento is used they become like those in Figure 5-14b. Portamento is like a lag processor for the keyboard. If your portamento has an off–on switch, it must be in the "on" position to function. The portamento attenuator then determines how much glide there will be

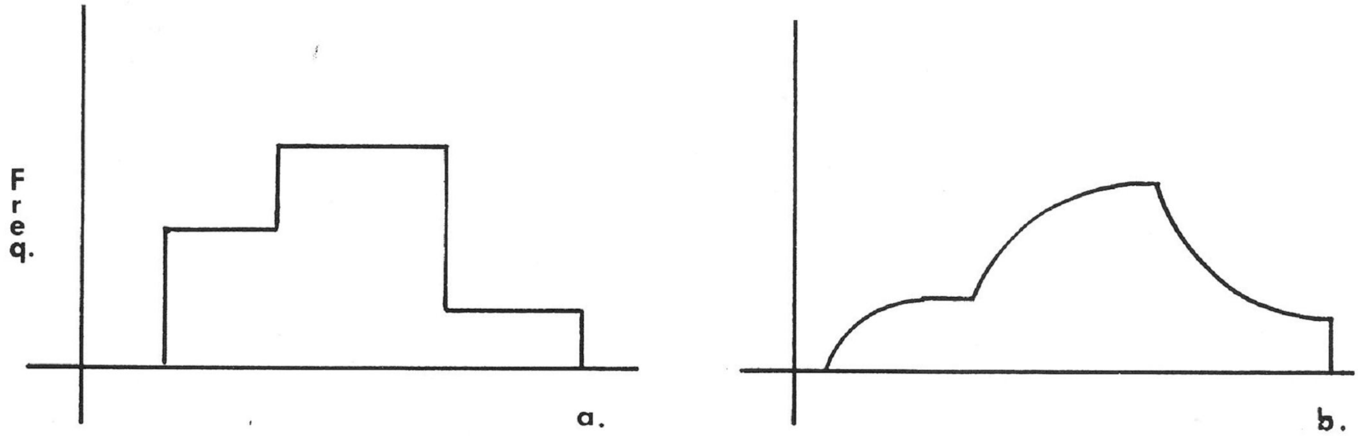
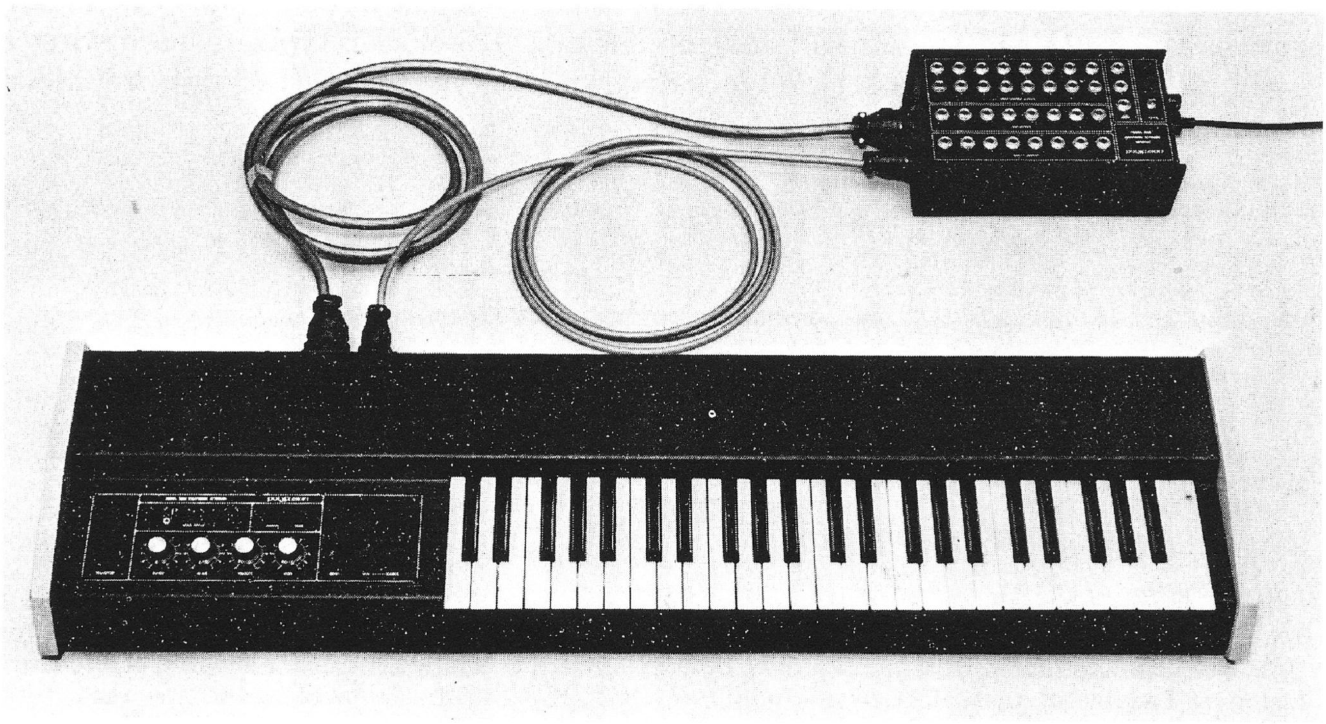


FIG. 5-14



Polyfusion 2058 Keyboard

(how much time it will take to reach the frequency of the key depressed). If you have the attenuator in a certain position and the portamento switch is off, depressing a "momentary" button (should your keyboard have one) will allow you to have portamento at the rate determined by the setting of the portamento attenuator just for the time you are depressing the button.

A *transpose* switch allows you to instantly make the keyboard go either one or two octaves up or down by flipping the switch in the direction indicated. Engaging this switch either adds or subtracts (that is, offsets) 1 or 2 volts to the keyboard control voltage; if +2 volts are added to any note, it will sound two octaves higher (at a 1 volt/octave rate); −2 volts will cause a pitch to sound two octaves lower.

Pitch bend is arguably the single most important device by which synthesists express themselves. A guitarist can bend a string; a trombonist has a slide. Now a keyboard player can play "between the cracks," bend a string (if a guitar-sounding patch is set up), create a slide trombone, and obtain similar effects.

There are many kinds of pitch-bend devices standard on synthesizer keyboards. The Mini-Moog, Moog Prodigy, KORG MS-10 and MS-20, and Sequential Circuits Prophet 5 and Prophet 10 have wheels; the PolyMoog and Yamaha CS- series have ribbon controllers; the Oberheim OBX series and OB-1 lead synthesizer use spring-loaded levers; ARP uses both an exponential rotary knob (perhaps the least effective of these devices) and a pressure-sensitive pad called PPC; KORG's Sigma offers two joysticks with the possibility of controlling two parameters (like pitch bend and modulation depth) simultaneously.

How to use a pitch-bend device effectively is not the subject of this book (but see Chapter 4, section I). To play a lead synthesizer you *must* practice again and again until you learn to use pitch bend to its greatest potential, (see section IX in Appendix A.)[9]

C. Sample and Hold (S & H)

In a typical S & H module, samples are taken of an input signal at a rate determined by a clock (an LF square wave) and are then made available at an output. The module derives a control signal from the original input signal. The clock is typically hard-wired as a sample command to the S & H. A block diagram of a typical S & H module would look like Figure 5-15.

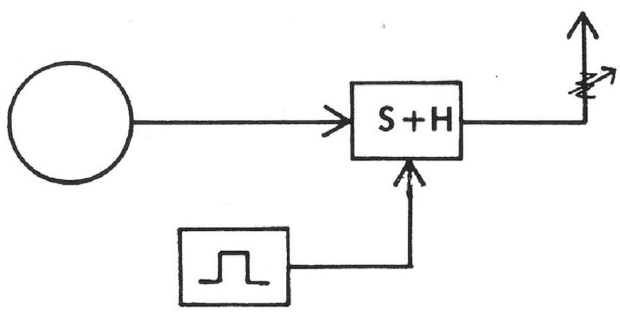
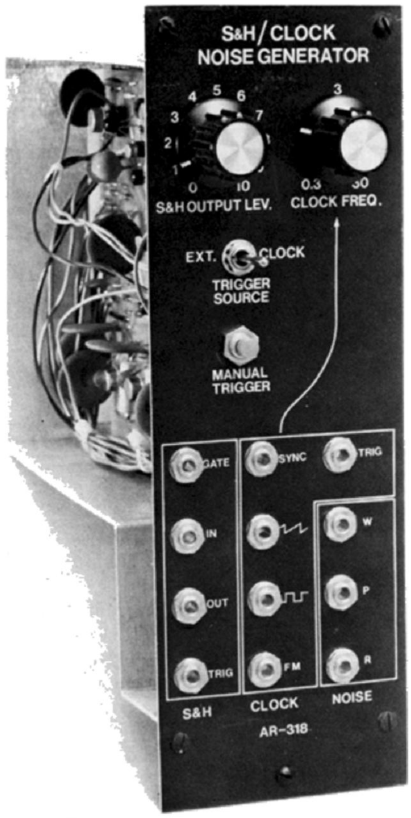


FIG. 5-15

[9] For a good introduction to pitch bending see the following columns in *Contemporary Keyboard:* Roger Powell, January 1977 and May 1978; Tom Coster, May, June, and July 1979 and *Synthesizer Basics,* May and June 1979. Even more helpful is to listen to anything by Jan Hammer, generally agreed to be a master of pitch bend. Listen to "Led Boots" from Jeff Beck's album *Wired* (Epic, PE 33849) and follow along with the transcription on page 46 of the February 1977 *Contemporary Keyboard.* You might also listen to "Festival" from George Duke's album *Follow the Rainbow* (Epic, JE 35701) and follow the transcription on page 92 of the October 1979 *Contemporary Keyboard.*

Noise (a typical pre-patch) is a signal that consists of all voltages occurring at random amplitudes. If it is the input being sampled, then at one instant a given voltage will be sampled and at the next instant another voltage. Figure 5-16a represents a graph of pink noise, 5-16b the sampling edge of the clock, and 5-16c the result when the clock samples noise. Let's assume that the voltages in Figure 5-16 are being used to control a VCO. For example, at point 1 in Figure 5-16c the clock has *sampled* a voltage that causes a VCO to output a frequency of about 100 Hz. *and holds* it until the next pulse of the clock, at which time a voltage that causes the VCO to output a frequency of 1000 Hz. is sampled and held until the next clock pulse, at which time a voltage that causes the VCO to output a frequency of 10,000 Hz. is sampled and held, and so on.

If noise is the input being sampled, the level attenuator of the noise generator must be at least somewhat open. The level attenuator of the S & H module will have to be at least partially open, as will the clock rate attenuator (unless you want the sampling rate to be *very* slow).



Aries AR-318 Sample and Hold, Clock and Noise Generator

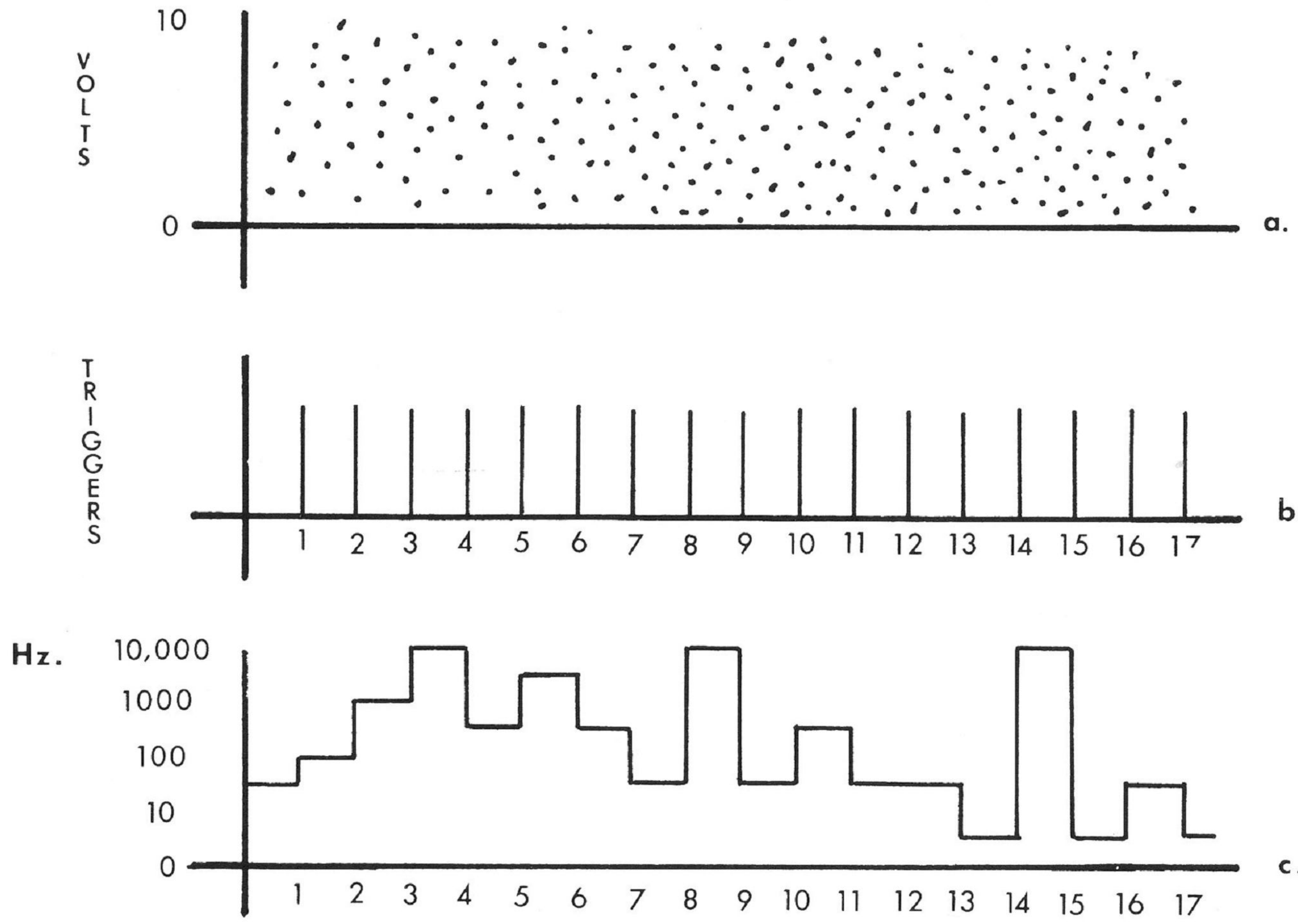
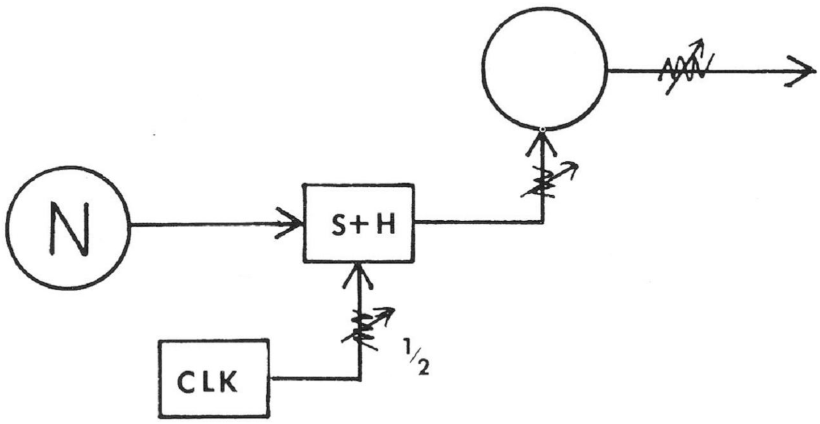


FIG. 5-16

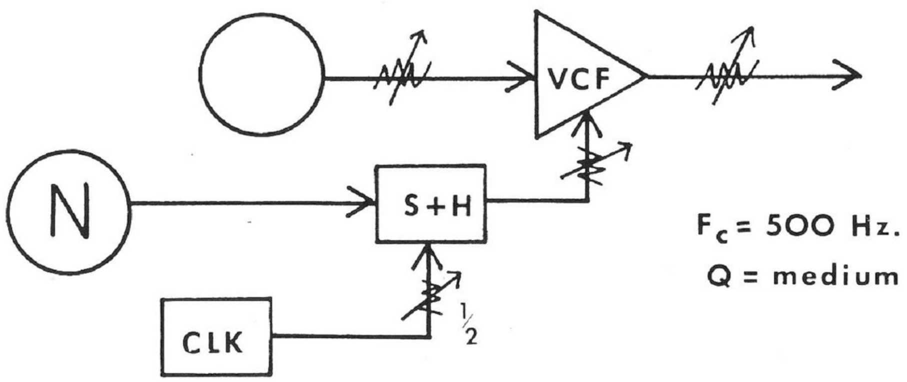
EXPERIMENT #75: Sample and Hold



Since noise is the signal being sampled, the NG attenuators should be open. The rate attenuator of the clock should be at level ½.

The noise input to the S & H is sampled and a control voltage derived from that by the S & H module. If, say, a voltage that would cause a VCO to output a frequency of 100 Hz. is sampled, that voltage is input into the CV input of the VCO, resulting in the appropriate pitch output. If a voltage four times the first voltage is next sampled, the subsequent pitch rise from the VCO would be two octaves higher than the first pitch.

A more useful, if typical, application of the noise input to the sample and hold module is to have the derived voltage control the filter.

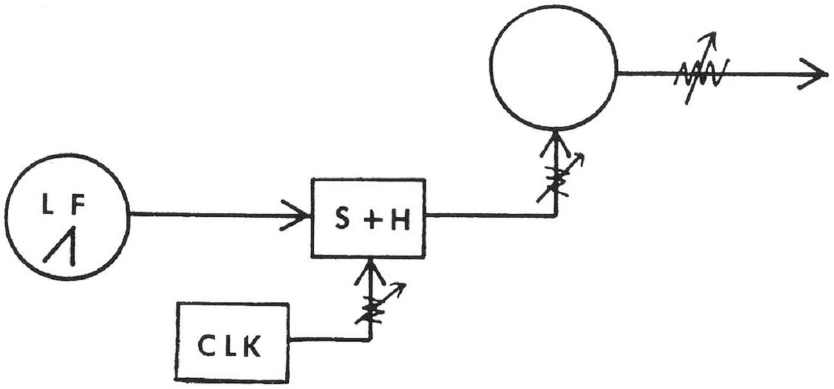


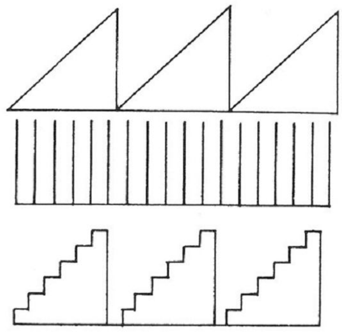
Set up the second patch shown here and hear the effect of the filter moving instantaneously from one F_C to another in a random manner.

You are not limited to having noise as the sampled input. Virtually any signal can be input, as you can see in Experiment #76.

EXPERIMENT #76: Voltages other than noise as sampled inputs

In this patch an LF sawtooth wave is sampled; the clock is running fast enough that it catches many points of that LF sawtooth wave, and the derived control voltage is used to control a VCO. You will need to experiment with the frequencies of both the VCO and the clock to get the right combination so that the "staircase" effect can be achieved.





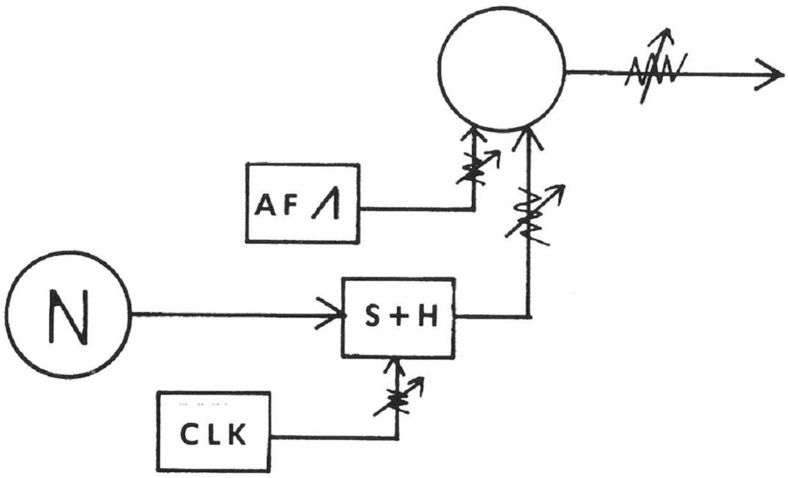
Try inputting other LF waves into the S & H module. Predict what the resulting sound will be like (if you are using the S & H output to control a VCO) before you listen to it.

Try inputting AF waves into the S & H module.

The level attenuator in the sample and hold module allows you to attenuate the range, or width, of the input being sampled, causing a smaller variation of control voltage to any given parameter.

EXPERIMENT #77: Attenuated sample and hold

Just to make the timbre more interesting, note that the audio signal is controlled by an AF sawtooth wave. This type of frequency modulation will be discussed in some depth in Chapter 6.



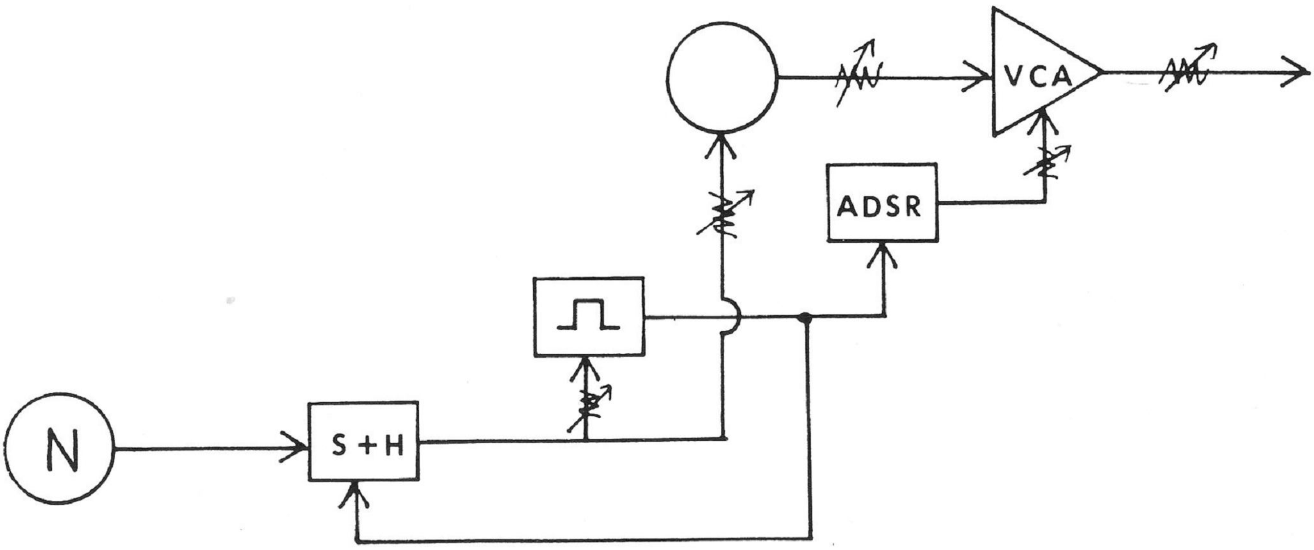
Start with the S & H level attenuator all the way closed. In this position no sampling takes place. Slowly open that attenuator and note that sampling takes place within a narrow range of voltages. As you raise the attenuator further and further the band of frequencies becomes wider and wider.

With the S & H level attenuator all the way open, slowly close the S & H CV attenuator into the VCO. The effect will be the same as lowering the level attenuator in the S & H module. You have two opportunities to attenuate these control voltages: either at the attenuator into the module being controlled or at the S & H module itself. If you attenuate at the S & H, then all modules controlled by the S & H will be similarly attenuated.

The output from sample and hold can be used to control any voltage-controllable parameter. It can also be further processed before controlling any such parameter. For example, since it exhibits sudden changes (represented by vertical lines when being graphed) it can be lagged; it can be inverted; it can be mixed with other signals. The modules on all synthesizers are part of a total gestalt and can and should be conceptualized as interdependent.

EXPERIMENT #78: Recursive randomness

In the patch shown here we create random pitches and random envelopes; however, the higher the pitch the shorter the event.



Noise is the input being sampled. The LF pulse supplies timing signals to the ADSR and functions as a clock to the sample and hold module itself; therefore its output must go to a multiple and then to the envelope generator input and the clock input to the sample and hold. Set up some attenuator settings on the ADSR and the rest of the patch is straightforward.

If you listen you will hear that indeed the higher the pitch the shorter the event. Can you describe why this is so?

The answer is that random voltages are controlling the clock so the LF square wave is going to output random triggers which in turn cause the sample and hold module to trigger itself randomly. Pitch of the VCO is random and envelopes are triggered randomly, controlling the VCA. When the pitch is higher, that means a relatively higher voltage has been generated by the LFO; the LF pulse moves more rapidly at a higher voltage and thus the time of the event itself will be shorter.

D. Clock

A clock may be an LF pulse or sawtooth wave with a general frequency range of approximately .03 Hz. to 20 Hz. It is most typically a 50% pulse (i.e., square) wave. It may be voltage-controllable. Its frequency is controlled by an attenuator. The clock output is available on quasi-modular and modular synthesizers, and a patch from this jack directly out will enable you to "hear" the clicks of the clock. There is typically an internal patch between the clock output and the sample and hold module; in addition it is sometimes pre-patched elsewhere (e.g., envelope generators). (See section X in Appendix A.)

E. Envelope Follower (EF)

An EF is a relatively simple device that derives a control voltage proportional to the changing amplitude of an input signal. This envelope voltage accurately follows the dynamic contour of the original audio signal. For example, the cymbal in Figure 5-17a, when struck, will have an amplitude envelope that looks approximately like that in Figure 5-17b.

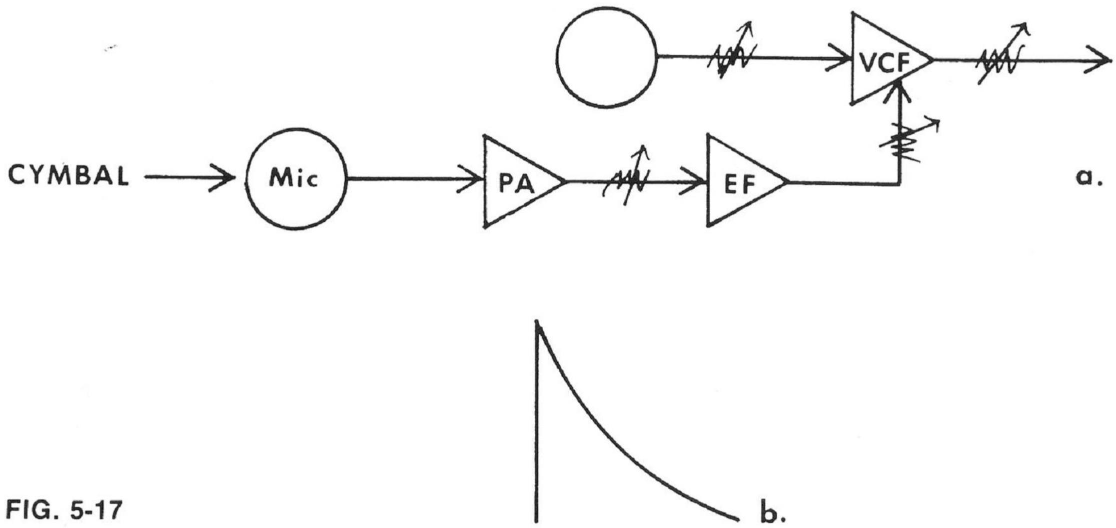


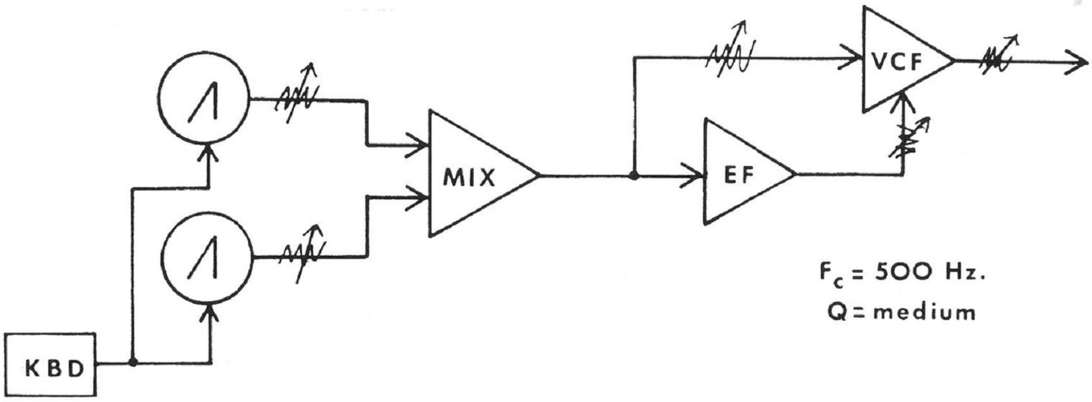
FIG. 5-17

The circuitry in the EF will track that envelope, allowing a control voltage to open the filter in proportion to the amplitude envelope of the cymbal. As soon as the cymbal is struck, the filter will open wide, allowing the VCO signal to pass; the filter will then gradually close as the cymbal's amplitude envelope decays.

A pre-amp is frequently pre-patched to an EF input because the EF is typically used with external signals. As you will now hear, it can also be used with a synthesizer's own signals.

EXPERIMENT #79: The sound of a "Kluged phase shifter"

In this patch both VCOs are generating AF sawtooth waves that are closely, but not exactly, tuned—perhaps a beat frequency of 1 Hz. To get the mixer output to both an audio and a control input to the filter, you will need to use a multiple.

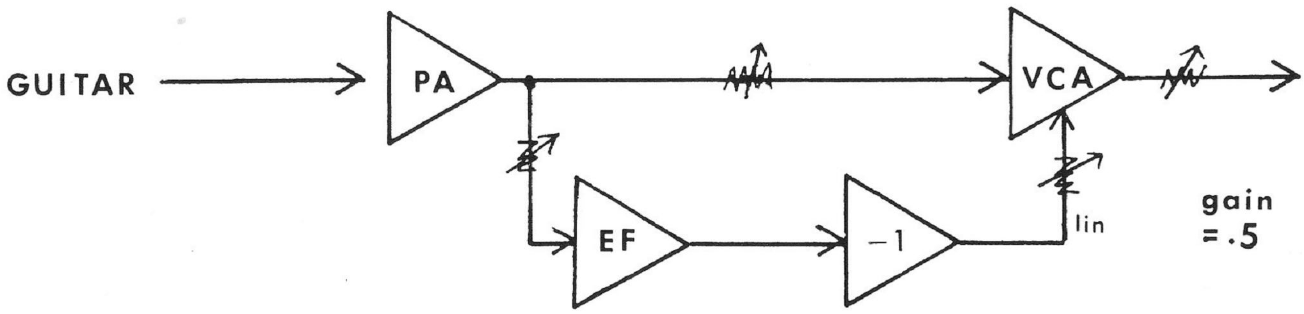


The beat frequency of 1 Hz. is a sine wave, with fluctuating amplitude. The filter is quite resonant, so any control voltage that sweeps the F_C will produce a sweep through the harmonics. Here the timing of the sweep is proportionate to *amplitude changes in the harmonics themselves, by the beat frequency. The harmonic sweep is not arbitrary, as it would be if you simply input the sawtooth waves into the filter and swept the filter with an unrelated LF sine wave. In effect this patch reinforces the entire spectral phenomenon of the beating sawtooths. A given harmonic will be reinforced at the same time that it lies within a resonant peak, exaggerating the effect of "phasing."*

You should be aware of at least one other use of an envelope follower—in making a quasi-limiter, ensuring that the amplitude of an external signal does not exceed a certain amount.

EXPERIMENT #80: A "quasi-limiter"

In the patch shown here, the EF detects the increases in the guitar's volume (amplitude). The EF output is inverted *and then used to control the VCA, whose initial gain is .5. An increase in volume will mean a proportionate* decrease *in control voltage reaching the VCA, which ensures that volume will not exceed a given point (the point being the initial gain control setting).*



F. Sequencers

All sequencers output control voltages and timing signals upon demand. In this respect sequencers are like keyboards and can control voltage-controllable modules and envelope generators. Unlike keyboards, however, sequencers *store* the information and output it upon receipt of a trigger. This trigger is typically, but not necessarily, an internal clock, generally voltage-controllable, which determines the rate at which the voltage and timing signals will be output. Sequencers have been used to create some of the most interesting and some of the most banal of synthesizer effects. Most synthesizer manufacturers offer either analog or digital sequencers.

1. Analog Sequencers

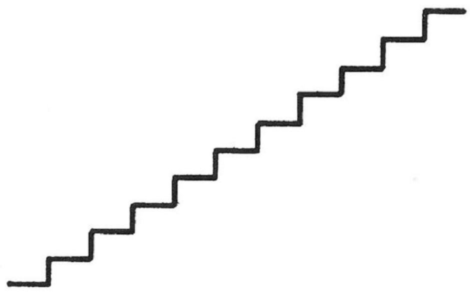
On analog sequencers the voltages to be output are determined by the settings of the many rotary knobs or vertical attenuators. Their voltage range is typically a minimum of 0 volts and a maximum of 2 to 5 volts. Zero volts is defined as the amount of initial DC voltage input into the module being controlled by the sequencer.

EXAMPLE:

Assume the sequencer is going to control a VCO, whose coarse frequency attenuator is set such that the VCO generates a pitch of 300 Hz.; 300 Hz. is then defined as 0 volts, and the sequencer inputs as many volts in order from that point

Korg Analog Sequencer SQ-10

as are set up on its controls. If the sliders or knobs were set such that each succeeding one was a bit more to the right (or up), then each step would output a bit more voltage than the previous step. The sequencer would generate a "staircase wave" (see the diagram). If that wave were used as a control voltage to the VCO, it would generate successively higher pitches until the end of the sequence, at which point it would either repeat or stop, depending on whether the sequencer was in the Run or Stop mode.



At the same time as the control voltage is output, a gate is available at a "gate out" jack on the sequencer. Using that to control an envelope generator allows sound shaping (Figure 5-18).

Analog sequencers generally have two control voltage outputs, which may be used in series or in parallel. If used in series, the length of the sequence may be twice as long as if

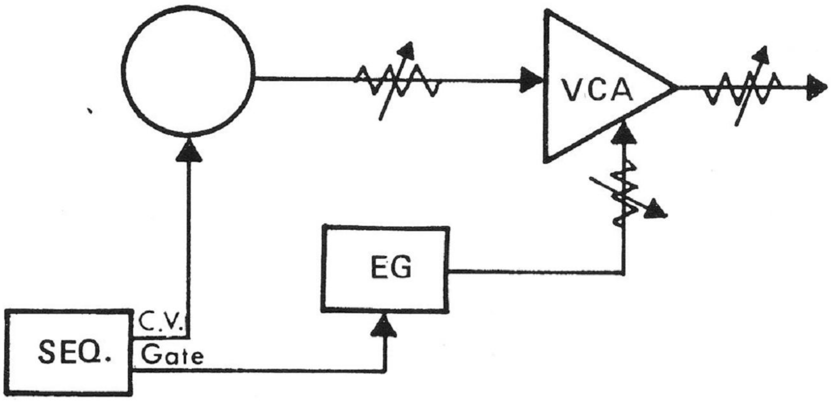
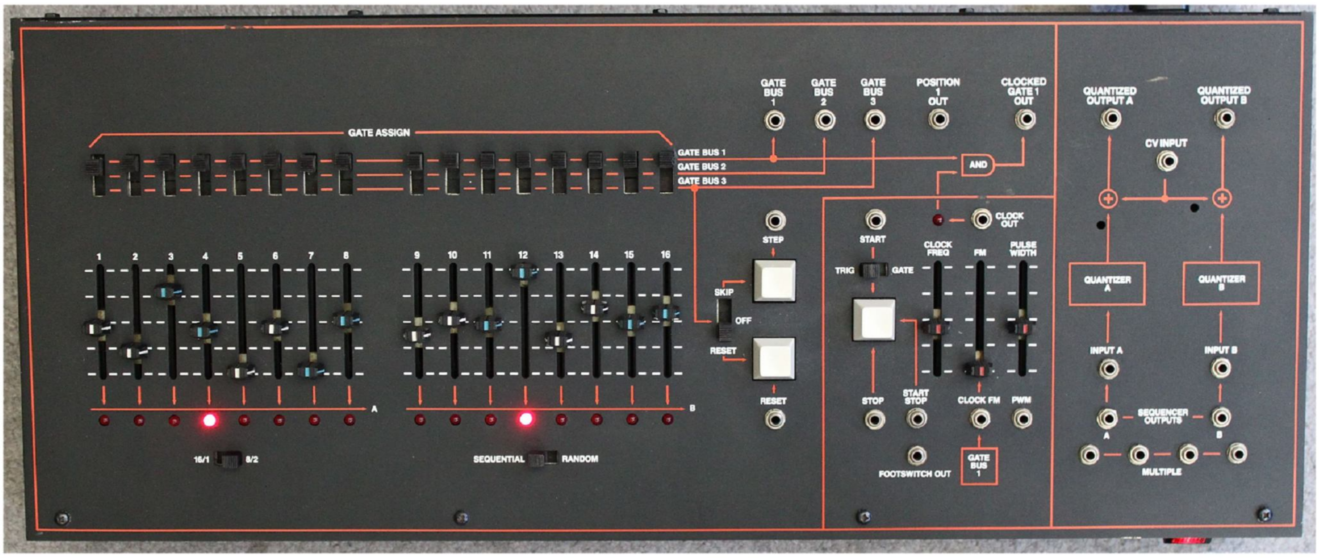


FIG. 5-18



ARP Analog Sequencer

used in parallel, but only one continuous control voltage is available. If used in parallel, two modules may be simultaneously controlled with different control voltages. For example, in the patch shown in Figure 5-19, if the controls of row A were set to increase slowly (that is to generate a staircase wave) while the controls of row B were set to decrease slowly, the pitch would rise as the filter closed.

Analog sequencers frequently have "position gate" outputs: a gate goes high when the sequencer arrives at the particular position, and is otherwise low. This allows for various effects, one of which is to accent a beat. If the sequencer in Figure 5-20 were generating an eight-step sequence, at the second, fourth, and seventh steps (positions) the filter would open more than its initial setting, causing a perceived accentuation at those steps.

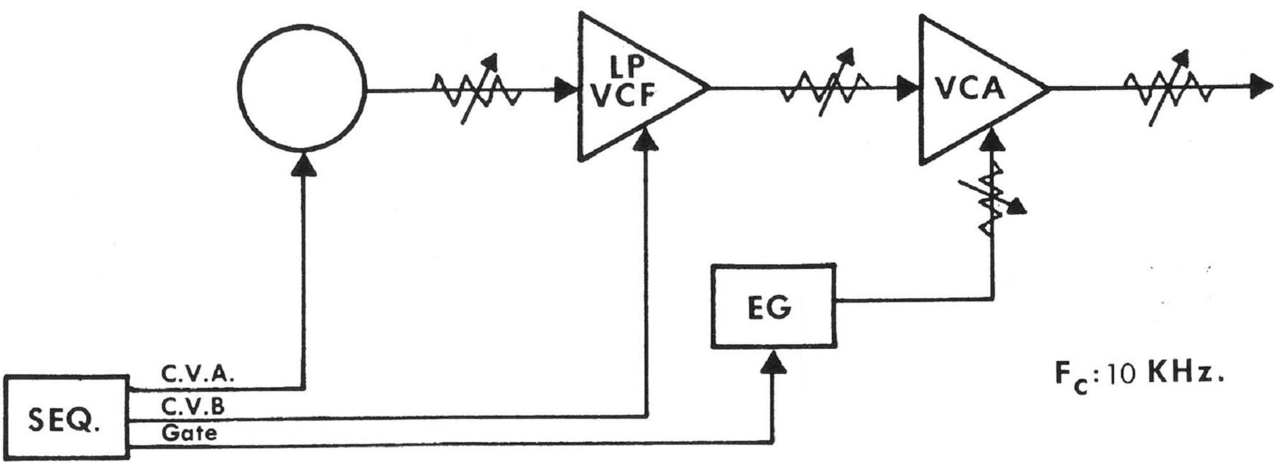


FIG. 5-19

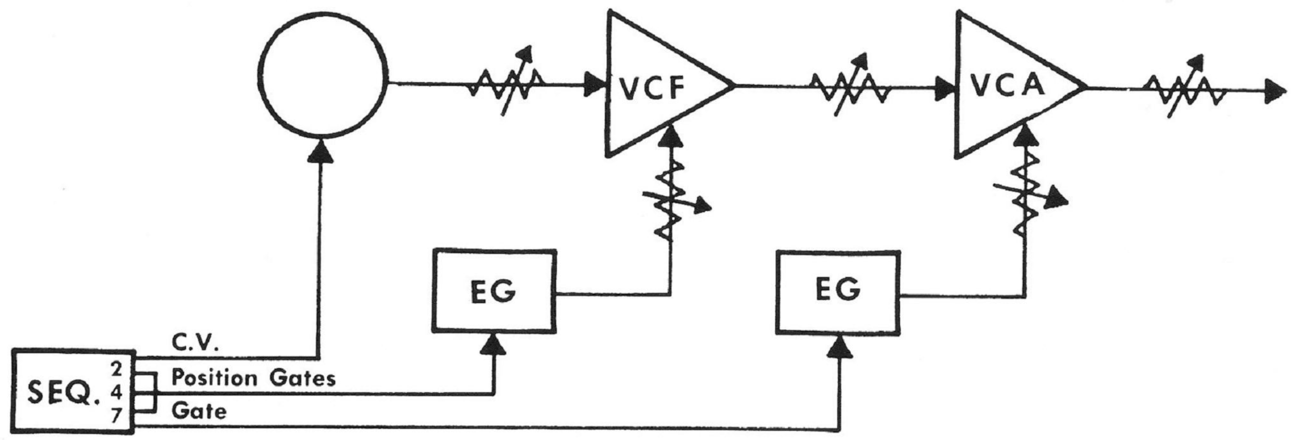
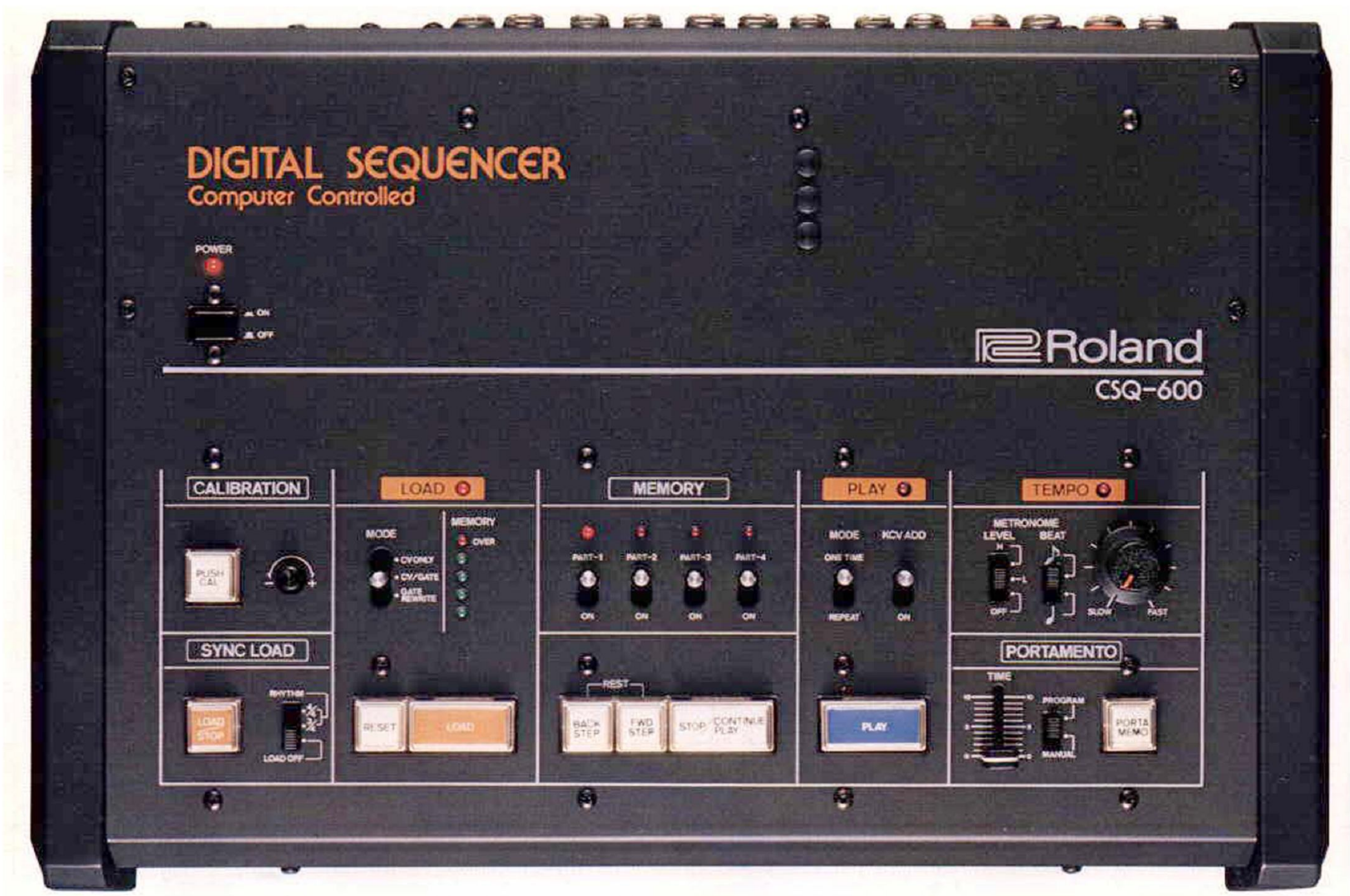


FIG. 5-20

2. Digital Sequencers

The memory of a digital sequencer is a microprocessor rather than the mechanical positioning of knobs or sliders; typically the memory has much greater capacity than that of analog sequencers, up to thousands of steps in the sequencers built into the digital synthesizers discussed in Chapter 7. The voltages may be fed into the digital sequencer in real time directly from a keyboard. The sequencer remembers those voltages as numbers and, using a digital-to-analog converter, outputs them as voltages upon demand. Digital sequencers remember not only the voltages but also the exact *rhythm* in which they are entered.



Roland Digital Sequencer CSQ-600

Sequential Circuits Model 800 Digital Sequencer

Do not think that digital sequencers are "better" than analog sequencers; they each have advantages. Digital sequencers frequently have a volatile memory, which means that they may "forget" everything as soon as the power is shut off; analog sequencers, whose "memory" is the mechanical positioning of a knob or slider, "remember" with or without power. Analog sequencers typically have many more inputs and outputs, and thus more flexibility, than digital sequencers.

The Roland MicroComposer and Emu Model 4060 Keyboard (and Model 4070 floppy disk memory unit) are essentially very powerful *polyphonic* sequencers, capable of controlling 8 and 16 simultaneous voices respectively. Sequential Circuits' Model 1005 is a 5-voice polyphonic sequencer that is intended to control a Prophet 5 synthesizer. Oberheim's DSX digital polyphonic sequencer can record up to 3000 notes polyphonically and in real time, with independently controllable gates and control voltages. Microcomputers (Chapter 7, Section II) may also serve as sources of polyphonic control voltages and timing signals and thus become powerful digital sequencers.[10]

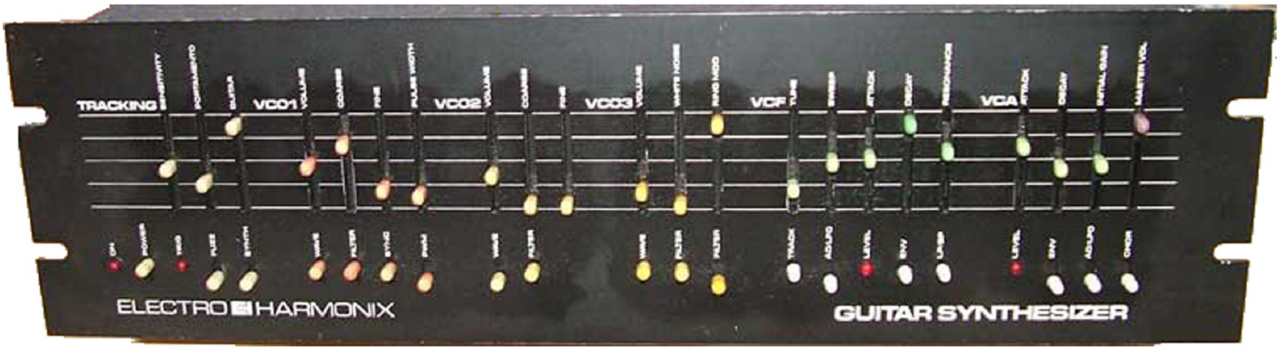
It is easy to fall into noncreative use of the sequencer because it is so easy to set up and, at first, so impressive in its effect. One such cliche is simply to use the sequencer to control a VCO. The sequence repeats itself rapidly and quickly becomes tiresome. Another such use is as a "rhythm machine" that has no rhythm, using only an LF square wave as a clock. Like a synthesizer keyboard, the sequencer requires considerable practice and experimentation before it is thoroughly understood and put to best use.

G. Traditional Instrument-Type Controllers

Although most people tend to think of synthesizers as keyboard instruments, a keyboard is only one type of voltage-controller. In the past several years controllers have been invented so that musicians other than keyboardists may use synthesizers. These in-

[10] See Roger Powell's columns in the February, June, July and Sept., 1981 issues of Contemporary Keyboard.

strumentalists require a synthesizer but don't need to pay for a keyboard. The Oberheim Expander Module, Steiner-Parker Microcon, Korg MS-50 and SMS Voice 400 are such synthesizers and are frequently interfaced with nonkeyboard controllers (although they may of course be interfaced with keyboards). The synthesizer from 360 Systems, the ARP Avatar, Electro-Harmonix guitar synthesizer, Roland GR-500, and the KORG X-911 are all monophonic guitar synthesizers that use a pitch-to-voltage converter to allow the guitarist to control a synthesizer voice. Of the guitar synthesizers, only the Zetaphon from HEAR and the Roland GR-300 are polyphonic.



Electro-Harmonix Guitar Synthesizer

Korg X-911

SMS Voice 400

(Photo by Helios Studios)

The Lyricon from Computone is a windlike controller and looks much like a clarinet. Computone has also recently introduced the "Humanizer," a device that allows the synthesist's *mouth* to create envelopes for various parameters. The Electronic Valve Instrument from Steiner-Parker has a brasslike control mechanism. The Synare 3 is a percussive controller from Star Instruments. The Syndrum is available from Duraline (11300 Rush St., South El Monte, CA 91733).

Lyricon II (from Computone)

Synare III

IV. INTERFACING SYNTHESIZERS

You may have the opportunity to control two or more synthesizers from one keyboard. For example, if one synthesizer has two VCOs, a VCF, a VCA, and two envelope generators, you would be able to control four VCOs, two VCFs, two VCAs and four envelope generators from one keyboard. The critical element is to have control voltage and timing signals from the keyboard get to all appropriate voltage-controllable parameters.

If one keyboard is to control two synthesizers, the keyboard's control voltage and timing signals must be available at jacks. Patch the KBD CV output to a multiple; patch the multiple outputs to (typically) the unattenuated control inputs of the "slave" synthesizer's VCOs and VCFs, (see section XI in Appendix A). Similarly, patch the keyboard's gate and trigger signals to the gate and trigger inputs of the "slave" synthesizer's envelope generator inputs.

chapter six

Basic Amplitude and Frequency Modulation

I. INTRODUCTION

A wave is *amplitude-modulated* if its amplitude (or the amplitude of any of its harmonics) is subjected to control voltage changes. You heard an example of amplitude modulation in Experiment #4, when an AF wave was subjected to periodic control voltage changes in amplitude by an LF sine wave. As the LF sine wave voltage varied, the amplitude of the AF wave varied proportionately.

A wave is *frequency-modulated* if its frequency is subjected to control voltage changes. You heard examples of frequency modulation in Experiments #16 and 17, when AF waves were subjected to periodic voltage changes by an LF sine and an LF square wave respectively, resulting in a vibrato and a trill. As the LF control voltage varied, the frequency of the AF wave varied (at a rate of 1 volt/octave).

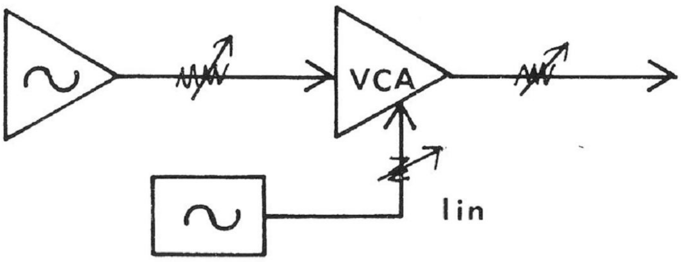
Although it is technically correct to say that a wave is amplitude- or frequency-modulated when it is controlled by an LF wave, henceforth when we refer to AM and FM we will mean the control of an AF wave by another AF wave. When it is AF, the ear can no longer distinguish each individual cycle of the controlling wave. The result is an aural illusion—the creation of *sidebands*[1], which have the possibility of creating richly textured timbres. Typically these sidebands will be additional frequency components which will not have a harmonic relationship to one another (although when used in small amounts they may increase the interest in harmonic music). However, there *are* certain ways to create harmonically related sidebands. Thus AM and FM are important ways in which a synthesist may create new sounds, frequently atonal, which may be further processed.

[1] You know that a moving picture is really a series of still pictures shown so quickly that the eye cannot detect each individual frame. This creates the illusion of motion. Analogously, sidebands give the illusion of the creation of new tones, because the controlling voltage is moving too rapidly for the ear to distinguish each cycle.

II. AMPLITUDE MODULATION

A. Using a VCA

EXPERIMENT #81: AM effects



Set the VCF to oscillate, so that the audio input to the VCA is an AF sine wave.

Initially set a VCO, acting as an LFO, at about 3 Hz. As you listen you hear an audio output proportional to the positive half of the sine wave's voltage swing. There is no output as the sine wave goes negative, because a VCA will not output if the control voltage is negative.

Slowly *increase the frequency of the LFO. As you approach 15 Hz. it will become more and more difficult to distinguish individual oscillations from the LFO. Now change the VCO's frequency to a low audio frequency. The sine wave from the VCO now has a frequency of about 20 Hz.* Slowly *open the coarse frequency attenuator. As the sine wave from the VCO (which was formerly acting as an LFO) becomes AF, different sound qualities will be heard. Stop at different points as you open the attenuator and listen to the many varied possibilities, depending on the VCO's frequency. You are hearing the sidebands created by the modulation of the oscillating filter by an AF sine wave from the VCO, together with the original frequency of the VCF's sine wave.*

In AM and FM the signal whose amplitude is being modulated is called the *carrier* (C), the modulating signal is called the *program* (P), and the resulting output is called the *modulated carrier.* In Experiment #81 the signal from the oscillating filter was the carrier and the signal from the VCO was the program, and the output from the VCA was the modulated carrier.

While it may have seemed to you that the strange assortment of sounds you heard performing Experiment #81 were random, they were in fact very specific and ascertainable. When a pure signal (one with no harmonics—that is, a sine wave) amplitude-modulates another pure signal, the modulated carrier has three components, whose frequencies are C + P, C − P, and C. An experiment will make this more clear.

EXPERIMENT #82: Creation of a harmonic timbre by amplitude modulation

Set up the same patch as in Experiment #81; however, this time tune the oscillating filter to any easily heard pitch and the sine wave from the VCO a fifth above it. Thus if you have tuned the filter to oscillate at middle C, the VCO is tuned to G above middle C.

Middle C (which is the carrier, C) has a frequency of about 261 Hz. G above middle C (the program, P) has a frequency of about 392 Hz. Thus if P amplitude-modulates C, the frequency components of the resulting modulated carrier will be C + P (261 + 392 = 653 Hz.), C − P (261 − 392 = − 131 Hz.) and C (261 Hz.). In these calculations a negative frequency (like C − P) indicates that frequency component is 180° out of phase with the other components. You hear it just as audibly as if it were a positive frequency (in this case of 131 Hz.) but if there were a positive frequency component of 131 Hz. in the modulated carrier, the two components would cancel.

We now have a modulated carrier which has frequency components of 131 Hz., 261 Hz., and 653 Hz.; 131 Hz. is an octave below 261 Hz. (it is half the frequency); 653 Hz. is an interval of a tenth above middle C. (See page 94.) Thus you have a timbre with frequency components of low C, middle C, and high E, a harmonic timbre.

The level of the control attenuator allowing P to control C affects only the amplitude (in this case, the volume) of the modulated carrier.

Here's another example to solidify what has been said so far. What will the frequency components of the modulated carrier be if you amplitude-modulate an 800 Hz. signal with a 320 Hz. signal and both signals are sine waves? See if you can figure out the answer before going on.

Answer: Since the 800 Hz. signal is being amplitude-modulated, it is C; since the 320 Hz. signal is doing the modulating, it is P. C + P = 800 + 320 = 1120. C − P = 800 − 320 = 480. You would have a timbre consisting of three frequency components at 480 Hz., 800 Hz., and 1120 Hz.

Experiment #82 illustrates another important rule in timbral construction by amplitude modulation: Whenever C and P are tuned to a ratio of integers, the frequency components of the modulated carrier will be harmonic; conversely, if the ratio of C and P is not one of integers, the frequency components will be inharmonic. Here are some examples to clarify this point:

EXAMPLE 1:

In Experiment 82 two oscillators were tuned a fifth apart. Their specific frequencies were 392 Hz. and 261 Hz. The fraction 392/261 reduces to 1.5, which is 3/2, which is a ratio of 3:2, a ratio of integers. The resulting modulated carrier had frequency components such that the pitches heard were low C, middle C, and high E—a harmonic chord.

If the frequencies of the carrier and program reduce to a ratio of integers, the resulting chord will always be harmonic.

Many harmonic intervals create a ratio of integers. Here are some examples:

Interval	Ratio
octave	2:1
perfect fourth	4:3
perfect fifth	3:2
major third	5:4
minor third	6:5
minor sixth	8:5
major sixth	5:3

This means that if C and P (which were a fifth in Experiment 82) were a fourth apart, or a major third, or any of these intervals apart, the frequencies would always reduce to a ratio of integers and the resulting frequency components of the modulated carrier would be harmonic.[2]

Try repeating Experiment #81 with different intervals for C and P as above.

EXAMPLE 2:

Suppose that C and P had frequencies of 217 Hz. and 86 Hz. respectively. The fraction 217/86 reduces to about 2.52, which does not create a ratio of simple integers. Therefore the frequency components of the modulated carrier will be inharmonic. In fact they will be 303 Hz., 217 Hz., and 131 Hz. 303 Hz. is between D sharp and D natural above middle C; 217 Hz. is slightly lowr than A below middle C; and 131 Hz. is low C. The chord is inharmonic.

The simple amplitude modulation of one sine wave by another creates an extraordinary number of possibilities for both harmonic and inharmonic timbral construction. However, the plot (and sound) thickens when either C or P are waves that have harmonics (higher-frequency components)—that is, any wave but a sine wave.

Let us suppose that in Experiment #82, P (the program, which had a frequency of 392 Hz.—G above middle C) had been a sawtooth instead of a sine wave. It would have had a fundamental (first harmonic) of 392 Hz. and additional harmonics (of diminishing amplitude) of 784 Hz., 1176 Hz., 1568 Hz., etc. In such a case the frequency spectrum of the modulated carrier is not only C, C + P, and C − P; *it would have also included C $\pm$ each harmonic frequency component of P.* Continuing with our example, the frequency components of the modulated carrier would be:

C	-	261 Hz.
C + P	-	563 Hz.
C − P	-	−131 Hz.
C + P(2)	-	1045 Hz.
C − P(2)	-	−523 Hz.
C + P(3)	-	1435 Hz.
C − P(3)	-	−1174 Hz.
C + P(4)	-	1827 Hz.
C − P(4)	-	−1305 Hz.

etc.[3]

The amplitudes of the additional upper and lower sidebands would diminish, because the amplitudes of the harmonics themselves diminish, but their effect would surely be perceived. Try substituting a sawtooth wave from the VCO instead of a sine wave in Experiment #82 and hear the difference in the modulated carrier with all those additional sidebands.

Of course there's no reason that C can't also be a sawtooth wave.

[2] Recall that some companies manufacture VCOs whose control inputs have linear, rather than exponential, sensitivity (Chapter 3, footnote 1). John Chowning, in an article entitled "The Synthesis of Complex Audio Spectra by Means of Frequency Modulation" (*Journal of Audio Engineering Society*, vol. 21, no. 7, September 1973, p. 526) discusses methods of obtaining harmonic spectra by means of frequency modulation with oscillators whose control inputs are linear.

[3] P(2) means the second harmonic of the program frequency. Since that frequency in our example was 392 Hz., P(2) would be 784 Hz. Similarly P(3) would be 1176 Hz., etc.

In that case you would not only have all the frequency components shown in the table above, but you would also have frequency components as follows:

C(2) + P	C(3) + P	C(4) + P	
C(2) − P	C(3) − P	C(4) − P	
C(2) + P(2)	C(3) + P(2)	C(4) + P(2)	
C(2) − P(2)	C(3) − P(2)	C(4) − P(2)	etc.
C(2) + P(3)	C(3) + P(3)	C(4) + P(3)	
C(2) − P(3)	C(3) − P(3)	C(4) − P(3)	
C(2) + P(4)	C(3) + P(4)	C(4) + P(4)	
C(2) − P(4)	C(3) − P(4)	C(4) − P(4)	
etc.	etc.	etc.	

These frequency components would go on for all the significant harmonics of both C and P. However, the amplitudes of the upper harmonics become so low that the additional energy they add to the modulated carrier becomes negligible.

Redo Experiment #82, substituting various other waves for C and P. You will have to use another VCO rather than the VCF (which only provides a sine wave to work with). Become familiar with the broad palette of timbres available through amplitude modulation.

B. Ring Modulation

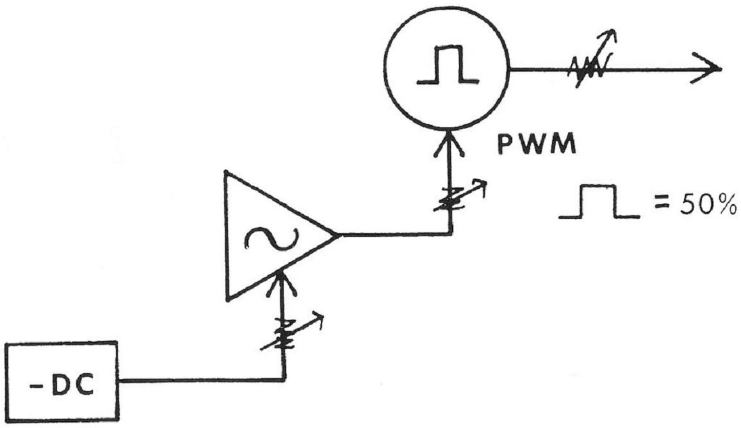
Ring modulation is a particular type of amplitude modulation. The frequency components of the modulated carrier depend not only on the nature of the input waves but also on whether the modulator is AC or DC. If it is DC *and* if C and P have DC components[4] in them, then the modulated carrier will consist of the sidebands and *both* C and P (remember that with a VCA only C was a component of the modulated carrier). If either C or P is AC and the other is DC, then the modulated carrier will contain sidebands *and* the original frequency of the wave that has a DC component. If both C and P are AC, the modulated carrier will have only sidebands; neither C nor P will be present in the modulated carrier. It follows that the modulated carrier will always consist of only sidebands when the modulator is balanced, since when in that position it eliminates any DC component (makes a balanced AC wave) of any wave input into it.

C. Pulse Width Modulation

Pulse width modulation is a form of amplitude modulation; specifically, as pulse width varies, the amplitude of different harmonics of the pulse become more or less attenuated. Picture what happens as you manually sweep a pulse width attenuator from 10% to 50%; the pulse starts out with every tenth harmonic missing. As you go toward 20% every fifth harmonic becomes attenuated; when you reach 20% every fifth harmonic is gone. You continue on towards 25% and every fifth harmonic returns but every fourth harmonic becomes attenuated. As you go toward 33% every fourth harmonic returns but every third harmonic is gone. Continuing on toward 50% every third harmonic returns but every even harmonic is gone. Thus as you change pulse width you change the amplitude of each harmonic of the pulse wave. If the changing of pulse width is done at audio-frequency speeds, amplitude modulation timbres will be created.

[4] See page 120.

EXPERIMENT #83: AF pulse width modulation



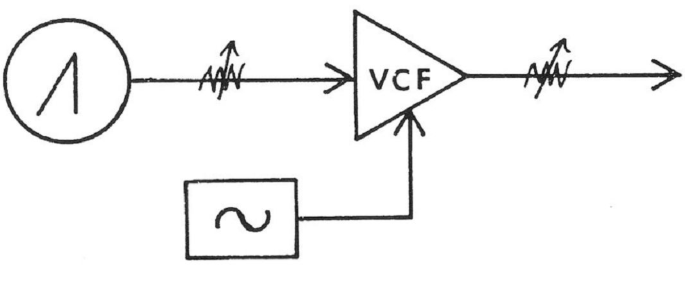
To set up the patch shown here, first get the filter to oscillate a sine wave. Listen to that sine wave and, while listening, raise the control attenuator into the filter so that the frequency of the sine wave goes lower and lower. It will go below audibility. At this point raise the pulse width modulation attenuator into the VCO (which is generating a 50% pulse—i.e., square—wave). If what you hear sounds garbled, raise the CV attenuator into the filter even more, further slowing down the LF sine wave, until you can hear the LF sine wave voltage-controlling the pulse width of the VCO, slowly sweeping the pulse width from one end to the other.

Now slowly begin lowering the CV attenuator into the filter. The LF sine wave will get faster and faster; at one point it will start creating sidebands. As you continue lowering it, you will hear the same type of amplitude-modulation effects you have been hearing using the VCA or the ring mod. The pulse is swept through its entire width so rapidly (at an audio frequency) that audible sidebands are created.

D. Timbral Modulation

You heard in Experiment #23 that you could voltage-control the F_C of a VCF with an LF sine wave. As the F_C varies, the amplitudes of different harmonics of input waves vary. If, for example, a 100 Hz. sawtooth were input into a typical low-pass VCF with an F_C of 200 Hz., harmonic attenuation would begin at 200 Hz. and the harmonics would be down 24 db at 400 Hz. However, if the F_C were changed to 1600 Hz., harmonic attenuation would not begin until that point. Thus filter modulation is selective amplitude modulation of the harmonics of any input waves; if filter modulation occurs at audiofrequency rates, then amplitude-modulation timbres can be created.

EXPERIMENT #84: AF timbral modulation



Start with the frequency of the VCO and the F_C both at about 2 KHz. and a VCO in its LF range. Slowly increase the LFO's frequency; when it has reached its highest *frequency in the LF range, put the range switch to audio frequency and move its coarse frequency attenuator from lower to higher frequencies. A wide variety of modulation timbres are created.*

As before, the program signal need not be a sine wave. Try substituting more complex waves from the VCO and hear how dense the modulated carrier becomes.

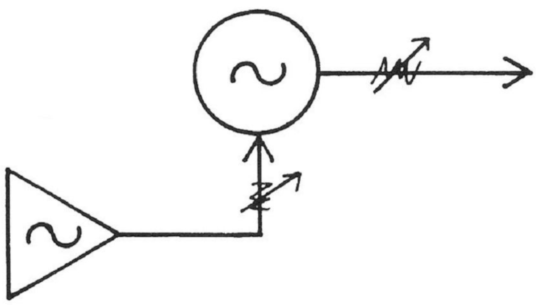
Note also that increasing the resonance of the filter has the effect of accentuating certain harmonics, creating still different textures.

If the filter were oscillating, then amplitude modulation would no longer be taking place. The filter would be generating a sine wave of frequency F_C and frequency modulation would occur.

III. FREQUENCY MODULATION

The modulated carrier that results when an AF wave frequency-modulates another AF wave is even more complex than when amplitude modulation occurs. Recall that with amplitude modulation the level of P that was allowed to modulate C affected only the *quantity* (volume) of the modulated carrier; it did not *qualitatively* change its nature. In frequency modulation the amount of P allowed to modulate C (the *modulation index*) qualitatively changes the nature (i.e., the frequency components) of the modulated carrier.

EXPERIMENT #85: Frequency modulation



Once again the VCF should be in oscillation so that one AF sine wave is frequency-modulating another AF sine wave.

As you slowly raise the attenuator into the VCO, you should hear two different things happening: (1) the overall pitch will begin to rise,[5] and (2) additional frequency components will begin to be added, creating a new modulation timbre. Note that it is not merely that the modulation timbre gets louder; it actually changes, the more program you allow to frequency-modulate the carrier.

Further experimentation will show you that the modulation timbre is affected by five variables:

1. C's initial frequency
2. P's initial frequency

[5] This pitch change does not occur if the VCO control input sensitivity is linear rather than exponential; see footnote 2.

3. C's harmonic content
4. P's harmonic content
5. The level of P allowed to control C (the modulation index)

In amplitude modulation only the first four variables will qualitatively change the nature of the modulation timbre. In frequency modulation the fifth variable, the level of P allowed to control C, creates even more complex timbres.

You will determine what musical value amplitude and frequency modulation have for you. A modulated carrier can be and often is further processed. These timbres frequently have no discernible pitch and are good source material for percussion instruments, bells, airplanes, machinery, and other sounds whose waveshapes are nonperiodic. Remember also that you are not limited to using VCOs for P and C; you could use your voice, an instrument, a sequencer cycling at an audio frequency, noise, etc. As always, experiment.

chapter seven

An Overview of Synthesizers

Most synthesizers are completely built by manufacturers.[1] However, you should be aware of two other possibilities if you have a lot of time and a small amount of money: You can build a synthesizer from scratch, if you understand some electronics theory, using circuits from *Electronotes* or *The Engineer's Notebook* from Radio Shack; or you can build a kit, available from Aries, EML, Paia, and Serge-Modular. In addition, E-mu offers submodules and voice cards (the electronic components of a synthesizer) for the experimenter. Godbout Electronics[2] and Blacet Music Research[3] also offer a variety of kits of interest to the synthesist.

A word of caution: Do not judge synthesizers by the presence or absence of special features. All other things being equal, the more "bells and whistles" a synthesizer has the more interesting it may be; but this does *not* mean it sounds good, and *that* is the most important feature any synthesizer can have. Always *listen* to any synthesizer you are interested in purchasing; do not be seduced by fancy features. The sound comes first.

I. ANALOG SYNTHESIZERS

Analog synthesizers consist of modules that generate continuous electrical signals that are modified or controlled by other electrical circuits or signals to yield a desired sound. They are contrasted with digital synthesizers, which store and return information

[1] Manufacturers' addresses are listed in Appendix B.

[2] Building 725, Oakland Airport, CA 94614.

[3] 18405 Old Monte Rio Road, Gurneville, CA 95446.

through extremely fast manipulation of numbers. Four types of analog synthesizer are available:

A. Preset Synthesizers

This class of synthesizer is generally used by the multi-keyboardist who wishes to be able to switch easily from one familiar instrumental sound (preprogrammed at the factory) to another. It offers little flexibility but great convenience at a modest price. Typically it will have buttons that say "Clarinet," "Banjo," "Tuba," and the like. It has few traditional synthesizer controls. Examples are the ARP PRO/DGX, Minitmoog, KORG KM500 MicroPreset, and various "string synthesizers" (which are also polyphonic).

B. Hard-Wired Synthesizers

This group has the largest variety in that almost all manufacturers (except those that offer only modular equipment) have several such nonpatchable synthesizers in their product line. Hard-wired synthesizers are characterized by signal and control paths that offer a variety of possibilities (that is, you may use one of several control paths offered) but limit the user to those possibilities that the manufacturer has chosen to include in the synthesizer. Hard-wired monophonic synthesizers typically will have a signal path that cannot be changed: one or two VCOs fed into a low-pass (and sometimes a high-pass) filter, which goes to a VCA and then out. A keyboard, LFOs, and envelope generators are available as controllers. These synthesizers may include some extra modules like sample and hold, a noise source, or a ring modulator (which has two VCOs, or a VCO and noise, as permanent inputs). Monophonic examples of hard-wired synthesizers include the ARP Axxe and Solus, Electro-Harmonix Mini-Synthesizer, Korg Sigma, Minimoog, Multimoog, Moog Prodigy, Moog Source,[4] Oberheim OB-1, Roland SH-09, Sequential Circuits Pro One, Yamaha CS-5 and CS-15, and Wasp.

ARP Solus

[4] The Moog Source is the first of a new generation of low-cost microprocessor *generated* (as well as controlled) waveforms. There are no attenuators on the face of the synthesizer—only touch switches that function together with an incremental control at the left of the keyboard. Like the Sequential Circuits Pro One, the Source has two built-in sequencers.

Crumar Stratus

Electro-Harmonix Mini-Synthesizer

Korg Sigma
(Unicord exclusive U.S.A. distributor)

Minimoog

Moog Opus 3

Moog Prodigy

Moog Source

Oberheim OB-1

Roland SH-09

Sequential Circuits Pro One

Some synthesizers (like the ARP 2600 and Octave Cat SRM) are capable of sounding two distinct notes played simultaneously, but the two notes are routed through one signal path. These are still one-voice synthesizers. ARP calls its 2600 "duophonic." The Oberheim two-voice (which usually includes an eight-step analog "mini-sequencer" capable of controlling one voice) and Yamaha CS 40M, which are capable of playing two notes simultaneously and routing them through two separate synthesizers slaved to one keyboard, are true two-voice (as opposed to two-note) synthesizers.

The current vogue in synthesizers is programmability—the ability of the synthesizer to memorize all or part of a desired patch and recall it at the push of a button. Some of the aforementioned synthesizers have built-in programmers. The Oberheim OB-1 and Yamaha CS-20M can each store eight programs; Paia's Proteus and Moog Source store sixteen programs, while the Yamaha CS-40M can store twenty programs. Excluding the Proteus, all these synthesizers may be interfaced with a cassette deck onto which programs can be "dumped" and stored; then new programs may be placed in the programmer's memory and the old ones retrieved from the cassette tape.

Sequential Circuits offers its Model 700 outboard programmer, a device that can store 64 programs and be interfaced with any otherwise nonprogrammable one-voice synthesizer. With this device any such synthesizer can be made programmable.

Excluding totally digital synthesizers, there are generally four types of hard-wired *polyphonic* synthesizers presently available: those that use one or more master oscillators and that have "divide-down" circuitry; those that use several modular-type synthesizers "slaved" to one keyboard; those that use several "dedicated" synthesizers slaved to one keyboard and that produce a "layer" of sound; and poly-voice synthesizers that use a microprocessor for greatest programming flexibility.

Polyphonic synthesizers first became available to the general public in 1975 with the introduction of the Polymoog. It achieves polyphony by a process known as "top-octave division": the division of the frequency of a supersonic oscillator into all the frequencies appropriate for the keys of a keyboard. The Polymoog has two master supersonic oscillators which are tunable so the user can set them at any interval. If they are set a fifth apart, depressing one key will sound two pitches a fifth apart (one keyboard control voltage controlling two oscillators). Depressing three keys would sound six pitches. The two VCOs can be tuned to unison or to a slight beat frequency for more typical keyboard playing. Each of these pitches is separately processed through "synthesizer on a chip" technology. This means that each key has its own separate synthesizer filter, VCA, and envelope generators, so it will have its own separate articulation. Even if you press a key on the Polymoog that causes that key's filter to open completely, you can still press another key and have that key's filter open in the same way; it won't encounter an already open filter (and thus give no impression of filter movement). However, all keys have the same filter and amplitude envelopes. The Polymoog is truly polyphonic; you can play as many of its 71 velocity-sensitive keys as you desire simultaneously.

The top-octave division principle is used by the ARP Quartet and Omni II, KORG's Lambda, and the Crumar Stratus. It is a powerful method of achieving total polyphony.

Oberheim marketed a radically different synthesizer, the four-voice with programmer, at about the same time that the Polymoog appeared. The four-voice has four separate synthesizers (Oberheim SEM's), each of which has two VCOs, a multimode VCF, VCA, two envelope generators, and an LFO. Since the Oberheim keyboard outputs four (or more) discrete control voltages and timing signals, all voices can be played simultaneously and still sound *completely* different from one another. This is the unique advantage that this approach has over all other approaches to polyphony. However, the pièce de résistance that made this instrument unique in its time was the development of the first polyphonic programmer, capable of remembering many (but not all) of the user-programmed parameters of each separate synthesizer. It can remember sixteen programs simultaneously. The keyboard has circuitry that allows it to control up to *eight* separate voices, which the Oberheim programmer can remember.

Polymoog

Oberheim 4-Voice with Programmer

Oberheim Synthesizer Expander Module

Polyfusion has recently marketed its Model 2058 keyboard, which is also capable of controlling up to eight separate voices. It has more advanced control mechanisms than Oberheim's keyboard, including a polyphonic *glissando* (something the Yamaha CS-80 and Oberheim OBX-a also offer).

Roland's Jupiter-4 uses the same general scheme of four voices and a programmer. It has one VCO per voice, offers eight user-programmable programs plus ten factory presets, and a self-contained Chorus Ensemble that helps to create a fuller sound.

The disadvantages of this approach to polyphony are that you can play from only four to eight notes simultaneously, not all parameters on the face of the synthesizers are programmable, and the instrument requires considerable discipline to learn.

The third (and presently most popular) approach to polyphony is the micro-

Sequential Circuits Prophet 5

Roland Jupiter 4

processor-assisted total programmability offered by Sequential Circuits in 1978 on its Prophet 5. Its programmer can remember *every* parameter that appears on the face of the Prophet, which makes it a very fast and easy live performance instrument. A keyboard player can literally plug it in and play it, since it comes with *40* factory programmed patches. Of course, all those programs can be completely redone by the synthesist.

Shortly after the Prophet 5 appeared, Oberheim brought out its OBX. The Prophet and OBX are remarkably similar: They both have first-rate sound, are about the same size and weight, use microprocessor technology and a cassette interface to store programs, have automatic tuning, a five-octave keyboard, an edit mode for easy editing of programs, quantized tuning (in semitone intervals) of one VCO, and a synch mode for the two VCOs. In early 1981 it introduced the OBX-a, an improved version of the OBX.

Oberheim OBX-a

There are important differences between the Prophet-5 and OBX-a, which are shown in Table 7-1.

The primary disadvantage of this approach to polyphony is that all voices are homogeneous—they sound the same, having the same filter and amplitude envelopes. A second disadvantage is that you can play only as many notes as there are voices—as few as four on the four-voice version of the OBX-a.

Oberheim has introduced the OB-SX, a "baby brother" to the OBX-a, which is less expensive and emphasizes factory presets.

ARP took a very different approach to polyphony with its Quadra. It contains four separate but dedicated synthesizers: one is for "bass," one for "strings," one for "polyphonic synthesizer," and one for two-note lead lines. With the different synthesizers playing simultaneously, a "layered," very lush sound is heard. The programmer stores sixteen programs (there is no cassette interface) but in a simplified way: where the Prophet 5 and OBX-a store the level of a control parameter, the Quadra programmer stores only information about whether a parameter is on or off, instead of how much. This provides less nuance than either the Prophet or the OBX-a programmers. However, included in the Quadra is a built-in stereo phase shifter through which any combination of the four synthesizers can be routed, and a unique sequencer-type effect that plays in sequence all notes being held down. The sequence can be memorized and played while the keyboardist plays additional string and polyphonic lines on top of it, for a very full sound. Roland's Jupiter-4 also offers this sequencer-like feature.

Korg's entry in this class of synthesizers is the Trident, an 8-voice polyphonic synthesizer that combines the "layer of sounds" approach of the Quadra with the full-information programming approach of the Prophet. The Trident's 16 VCO's are routed to three separate sections, which are called "polyphonic synthesizer," "brass," and "strings." Any oscillators not used in one section are available for use at another. The "synthesizer" section has two VCO's, one of which makes available only a sawtooth wave.

TABLE 7-1

	S.C. Prophet	Oberheim OBX-a	Korg Trident	Roland Jupiter 8	ARP Quadra
Type of filters	4-pole	switchable 2- or 4-pole	4-pole	switchable 2- or 4-pole	4-pole
Number of voices	5/10	4, 6, or 8	8	8	8
Modulation hardware	wheels	spring-loaded lever	2-axis joystick	spring-leaded lever and touch pad	pressure-sensitive keys
"Layers of sound"	No	No	Yes	No	Yes
Sample & Hold	No	Yes	No	Yes	No
Number of patches stored in real time	40/64	32	16	64	16
Cassette interface	Yes	Yes	No	Yes	No
Ability to play two or more different sounds simultaneously	No	Yes	Yes	Yes	Yes
Sequencer/Arpeggiator	No/*	No	No	Yes	Yes
Keyboard split	No/*	Yes	Yes	Yes	Yes
Oscillator Synch	Yes	Yes	No	Yes	No
Additional	*Prophet 10 has 2 keyboards; OSC scaling; polyphonic sequencer option; filters oscillate	2-range transpose switch; polyphonic portamento	Built-in flanger; no edit mode or method of storing patches externally	See text	Built-in V-C phase shifter; limited programming (see text)

Arp Quadra

Sequential Circuits Prophet 10

Its one ADSR envelope generator serves as a control voltage to both its VCF and VCA. The "brass" section of the Trident has its own separate ADSR envelope generator; thus this section is not limited to the generation of only brass sounds. The "strings" section has an AR envelope generator, 6 db/octave filter and equalizer. The Trident has a built-in flanger, a keyboard split facility that allows the user to have one sound in the lower two octaves and a completely different one in the upper three octaves, a two-axis joystick for pitch bending and modulation effects, and three pre-sets: piano 1, piano 2, and clav. Its programmer works like the Prophet's, remembering all the parameters of a particular patch; however, it remembers only 16 patches, has no edit mode and no cassette interface.

In addition to the features shown in Table 7-1, Roland's Jupiter-8 (also called the JP-8) offers eight different "Patch Preset" pairs, which enable the performer to call up a different program for each end of a split keyboard by touching only one button. It has four keyboard assignment modes: solo mode turns it into a monophonic keyboard for lead line playing; unison mode puts all eight voices (16 VCOs) under one key, four each under two

Korg Trident (Unicord exclusive U.S.A. distributor)

Roland Jupiter 8

lines played, and so on; poly 1 mode is the normal one-voice-per-key polyphonic; and poly 2 mode is like poly 1 except that it allows only the last note or notes struck to release the full length of time programmed; all other notes have a shorter release to promote mid-line clarity. The LFO generates sine, square, sawtooth, and sample-and-hold control voltages, and the VCOs generate triangle, sawtooth, square, pulse, and noise waveforms.

The Yamaha CS-50, CS-60 and CS-80 polyphonic synthesizers, which offered velocity-sensitive keyboards and limited, analog programmers, were replaced in 1981 by the SK- series. Billed as "symphonic ensembles," they are hybrid electronic organ/synthesizer combinations. However, the CS-70M is a true six-voice polyphonic synthesizer that uses magnetic data cards (instead of cassettes) to store patches externally, and that offers a built-in limited polyphonic sequencer.

A recent innovation in chic keyboard display has been the advent of keyboards worn around the neck, freeing the keyboardist from the confines of staying behind an instrument, allowing movement with all the flexibility and appeal of a guitarist. Jan Hammer, George Duke, Gary Wright, and Roger Powell have been using keyboards like these for some time,[5] but only in 1980 did such keyboards become available to the general public. These keyboards include the Clavitar,[6] the Syntar,[7] and the Liberation from Moog.

The Clavitar

Moog Liberation

[5] See the covers of *Contemporary Keyboard* for May 1978, October 1978, October 1979, and July 1980.

[6] The Clavitar is available from Davis Music Electronics Co., 1816 North Harvard Boulevard, Hollywood, CA 90027. It is a controller only, requiring a separate synthesizer to control.

[7] The Syntar is available from Performance Music Systems, P.O. Box 6028, Bend, OR 97701.

The Syntar

C. "Quasi-modular" Synthesizers

This group of synthesizers typically has available inputs and outputs to modules and uses patchcords, together with hard-wired pre-patches that are defeatable. Although this group is generally more flexible than hard-wired synthesizers, you must buy those modules that the manufacturer has chosen to include with the synthesizer; hence the name "quasi-modular." The ARP 2600 is quasi-modular, as are the Electrocomp 101, Korg MS-10, MS-20, and PS- series, the Roland System 700, the Synthi VCS-3 (Mark II), Wavemaker 6, and the recently discontinued ARP 2500.

Of the quasi-modular synthesizers only the Korg PS- series is polyphonic. Each of these three synthesizers has twelve master oscillators which, like the Polymoog, use divide-down circuitry to provide the appropriate pitch for each key. Because there are

Korg MS-20 (Unicord exclusive U.S.A. distributor)

Wavemaker 6

Synthi VCS-3

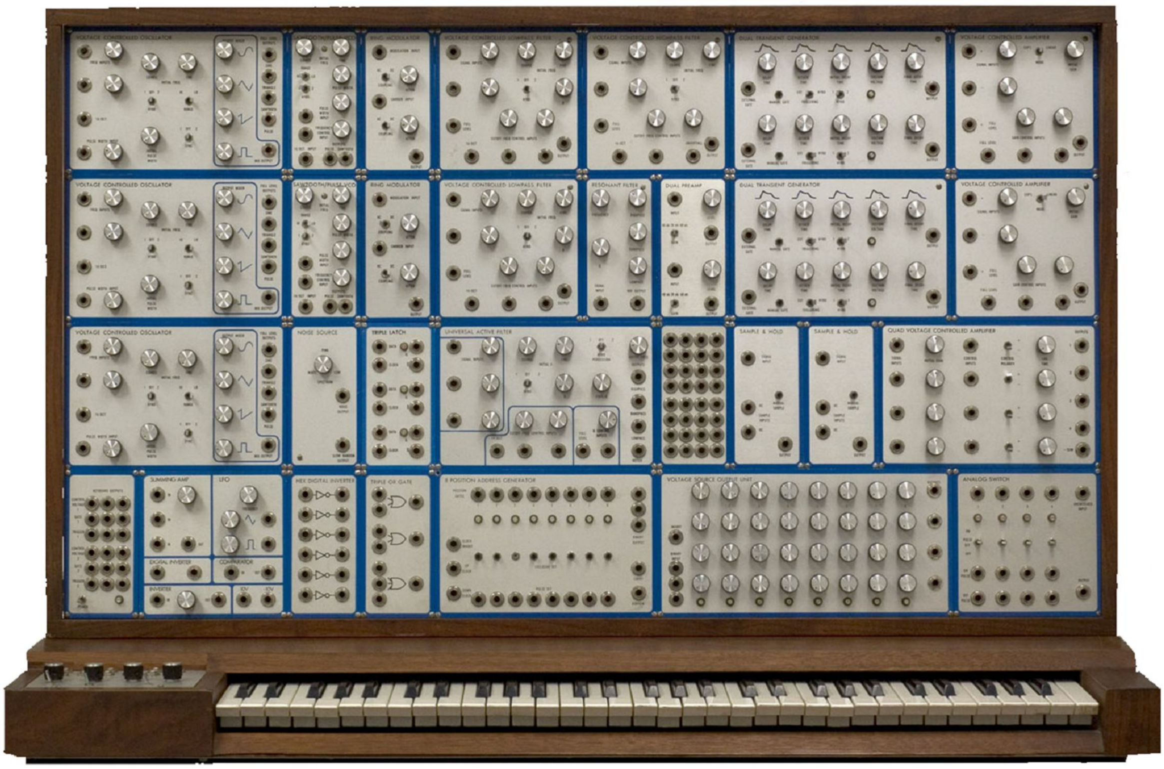
twelve (instead of two) master oscillators, the synthesizers can and do provide a "scaling" feature (as does the Prophet 5), that allows each note in the chromatic scale to be tuned plus or minus a semitone. This allows for just, Pythagorean, and other types of unusual tunings.

The PS-3100 offers one VCO for each key of the chromatic scale, a three-peak voltage-controllable "resonator" which functions as a complex filter, and a "chorus" device. The

Korg PS-3200
(Unicord exclusive
U.S.A. distributor)

PS-3200, which offers two VCOs per key, is the only quasi-modular polyphonic programmable synthesizer commercially available. The PS-3300, with three VCOs per key, has been described by Bob Moog as the best of the polyphonics "for sheer versatility in sound shaping and potential for animated 'fat' sound."[8]

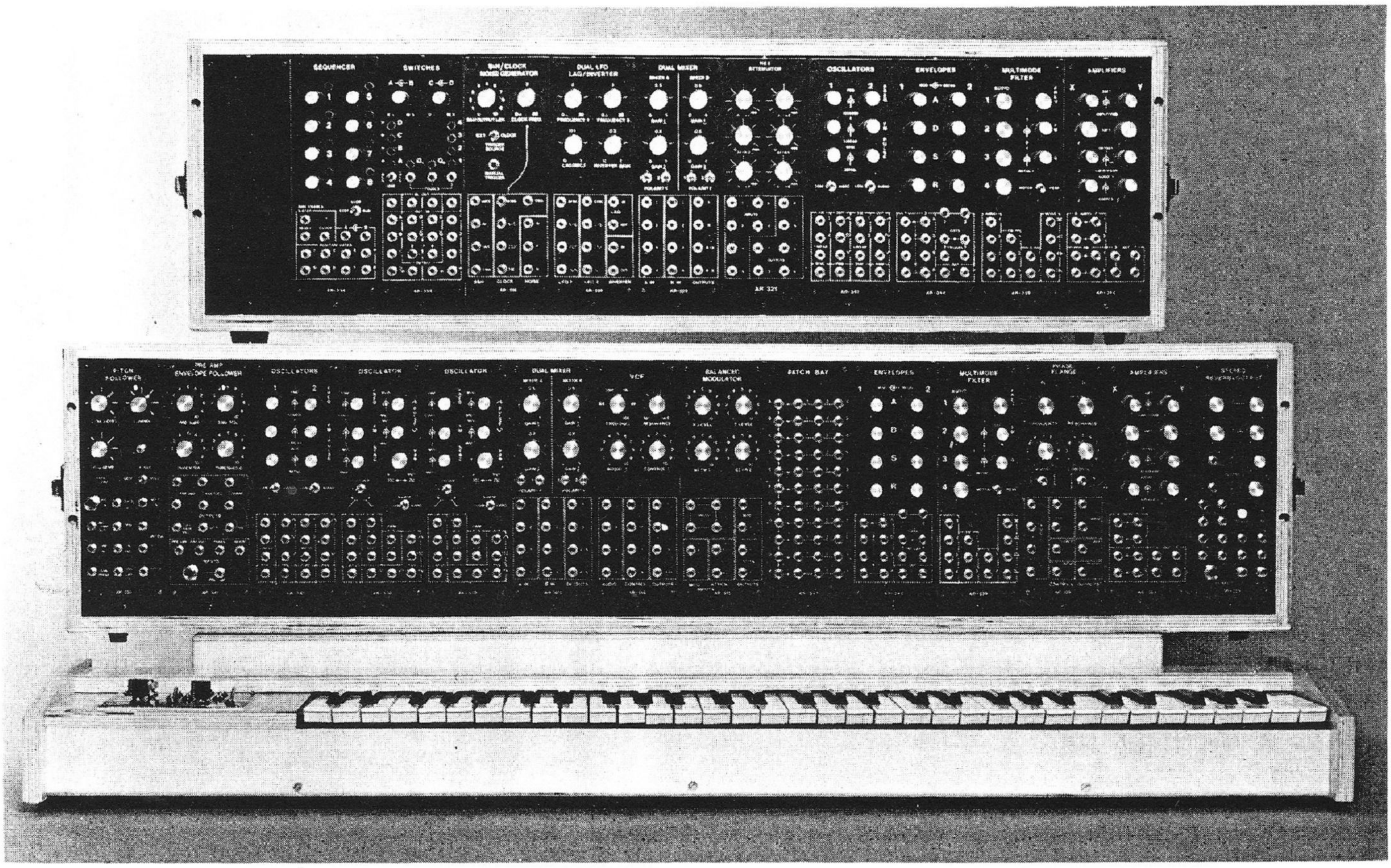
D. Modular Synthesizers



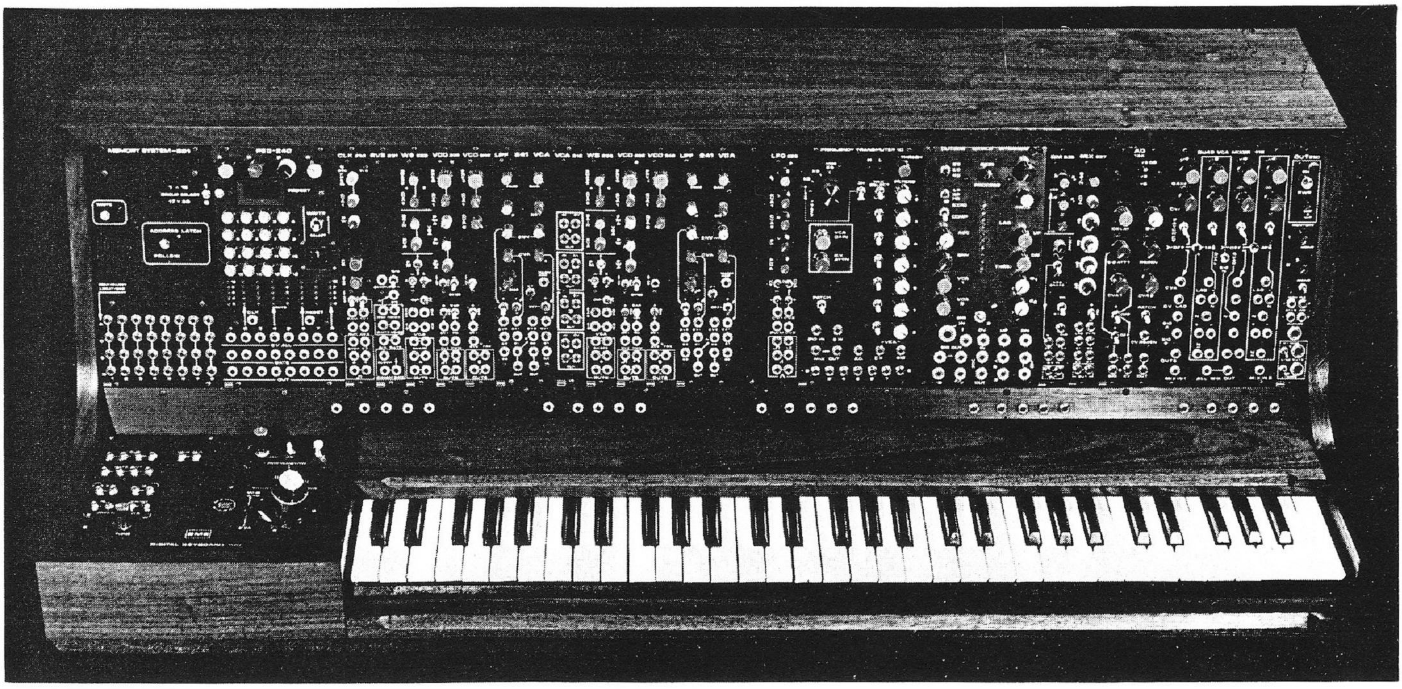
E-mu

[8] *Contemporary Keyboard*, vol. 6, no. 7 (July 1980), p. 64.

Aries, EMS, E-mu, Moog, Polyfusion, Roland, Serge-Modular, SMS, and Wavemakers offer modular systems. Each module may be purchased separately, so the user tailors the system to her or his specific desires. These systems always use patchcords, although typical interconnections (e.g., VCO–VCF–VCA) may be pre-patched by the user. These are the most flexible analog synthesizers because they have just the modules the synthesist needs and have all inputs and outputs accessible.

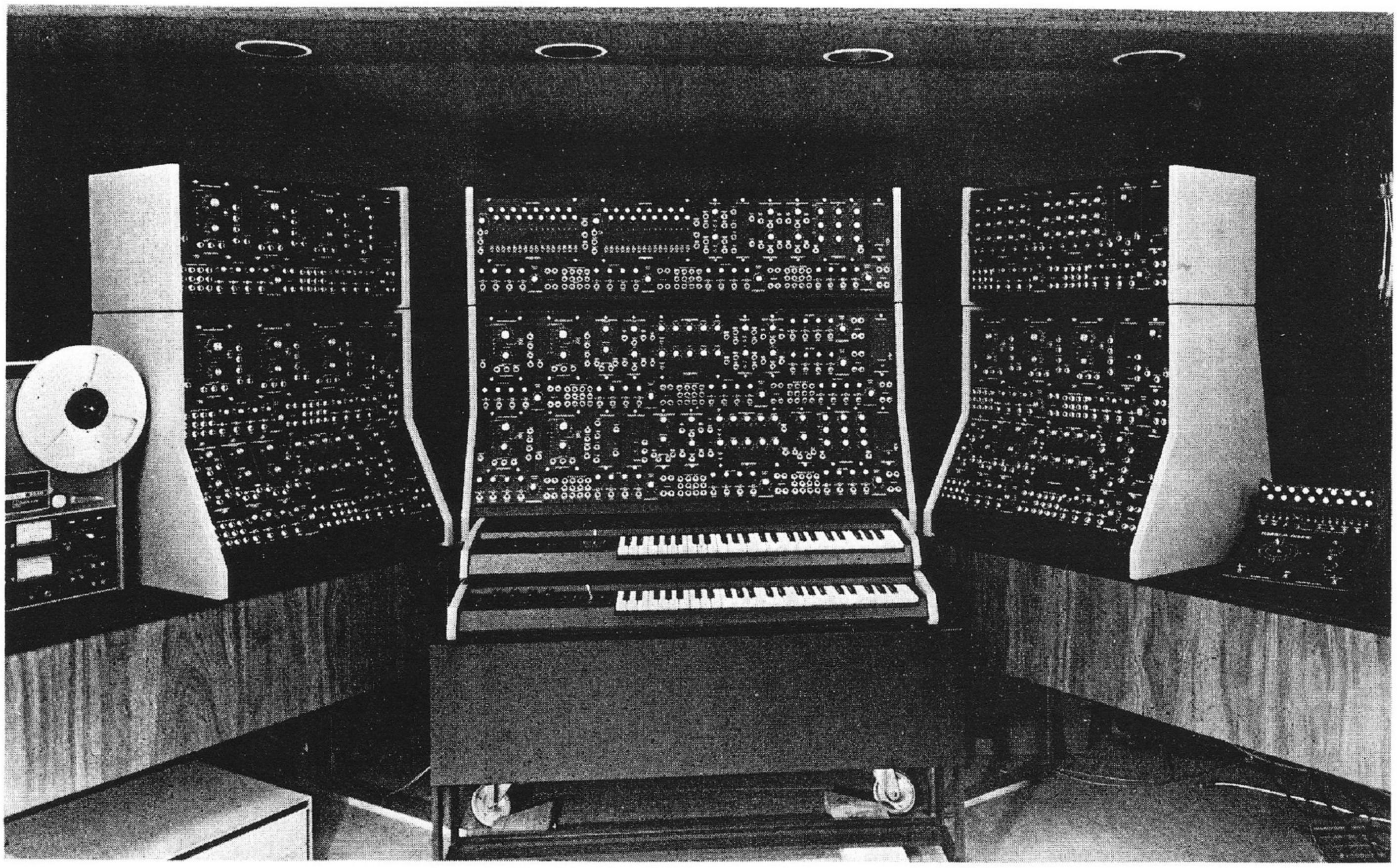


Aries Keyboard System III



SMS Modular Synthesizer

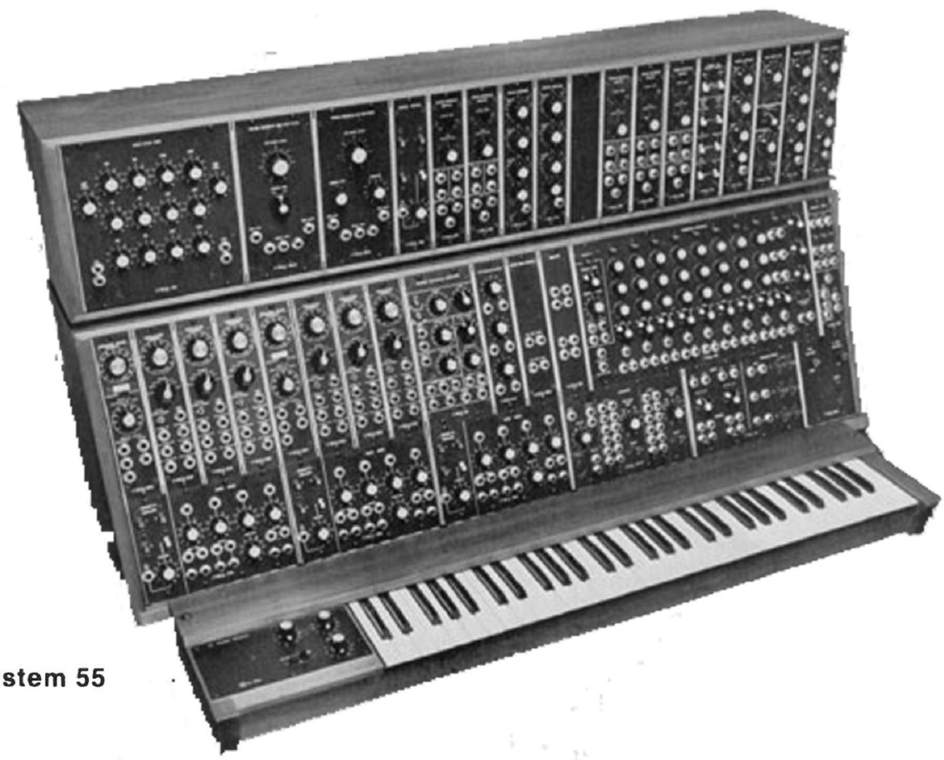
(Photo by Helios Studios)



Polyfusion

They are high-priced and of highest quality in their technical specifications. In addition to selling each module separately, virtually all of these companies offer suggested packages of modules which they sell at some discounted price.

Although we have seen many brands of synthesizer in these pages, the ways in which these instruments are similar are much more profound than the ways in which they differ. All the instruments have signal paths that are essentially VCO–VCF–VCA; they all use controllers like envelope generators and LFOs; they all use a keyboard as a primary controller.



Moog System 55

EMS Synthi 100

The reason these instruments have such basic similarities is that they all follow the design implemented in the mid-1960s by Robert Moog. Bob Moog is to the voltage-controlled synthesizer as Thomas Edison is to the electric light bulb. It was Moog who, in conjunction with others,[9] designed the first voltage-controlled modules, analyzed musical events and developed ADSR envelope generators, designed filters that acted upon certain waveshapes already prevalent in radio technology, and incorporated the keyboard as a primary control mechanism. More than any other person, he has defined what a synthesizer is. Sadly, very few people have expanded upon his early designs. The Prophet 5 may use microprocessor control to remember patches, but all the sound generation is analog and not very different from having "five Minimoogs in one box."[10]

One person who did approach synthesis differently is Serge Tcherepnin, developer of Serge-Modular Music Systems. Serge's aesthetic is to bring the *electronics* to the foreground, making everything available. He approaches the synthesizer as an instrument that *can* produce tonal, typically keyboard-oriented music, but that is unnecessarily limited by an adherence to that structure. Thus he produces a wide variety of unusual modules that are commercially available nowhere else, like the new timbral oscillator, filters that have voltage-controlled slope, band width, and resonance (in addition to cutoff frequency), wave multipliers, and analog shift register. All of these help to produce a richer palette of sounds than those available with more conventional analog synthesizers. In addition, he has developed the concept of "patch programmability," whereby one module may serve many different functions depending upon how it is used in a patch. As an exam-

[9] Some of those who worked with Moog in early years include Vladimir Ussachevsky, Walter Carlos, Myron Schaeffer, Gustav Ciamaga, and Herbert Deutsch (now director of marketing at Moog Music, Inc.).

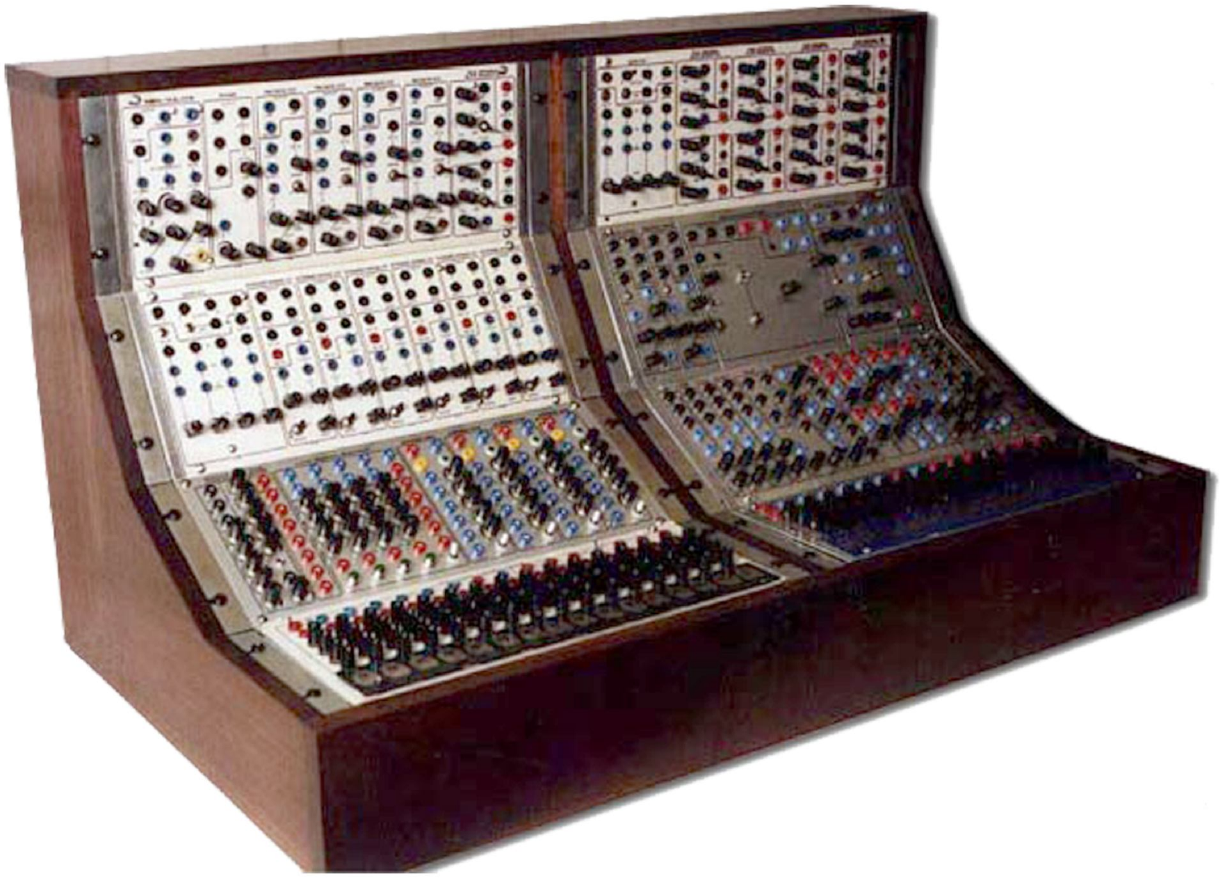
[10] Dave Smith, designer of the Prophet 5, quoted in *Contemporary Keyboard*, vol. 5, no. 11 (November 1979), p. 61.

ple, his Dual Universal Slope Generator may be used as a voltage-controlled envelope generator, LFO that is voltage-controllable from a sawtooth to a triangle wave, portamento device, envelope follower, subharmonic series generator, VCO, or nonlinear low-pass filter. His modules come with a minimum of documentation, Serge's philosophy being that telling you what the module does limits your own experimentation; the person who is willing to work at it makes a Serge her or his own instrument.

Anyone who has come this far must realize the monumental debt synthesists owe to Bob Moog, but no overview of synthesizers can be complete without an introduction to Don Buchla, the person whose approach to the creation of electronic music is most different from Moog's. Moog and his associates analyzed musical events and created an instrument well suited to their production; Buchla put together certain electrical components with no prior idea of what the aural result should be. Moog's reasoning was deductive, Buchla's was inductive. Moog worked with Walter Carlos, Buchla with Morton Subotnick. Moog's design was accepted almost universally, Buchla's at a few universities. If Bob Moog is the Thomas Edison of voltage-controlled synthesis, Don Buchla is surely Nikola Tesla.

The generality of Buchla's system makes it difficult to specifically contrast it with Moog's design. His instrument is not a "synthesizer" at all (that is, the VCO–VCF–VCA design of Moog), but rather an "electronic music box." Buchla's VCOs are digital and deal with things like low- or high-ordered harmonic content, degree of harmonic content, and symmetry of harmonics. Control voltages and audio voltages are isolated and treated as completely different elements of the instrument. There are no VCFs: the instrument is additive rather than subtractive in terms of timbral construction. Envelope generators are not of the AR or ADSR type, but have 256 individual "segments."

Buchla first produced the 100 series in the 1960s; the 200 series was designed in 1971 and refined since then; the 300 series is basically modules of the 200 series coupled with general-purpose computers programmed to aid the performer/composer.



Serge-Modular System Series 1979

The first synthesizer record to achieve widespread popularity was *Switched on Bach* (Walter Carlos, Columbia MS-7194), a transcription of seventeenth-century music, using a Moog modular system. Moog's design is best implemented producing tonal and melodic music of the past 350 years. Buchla's design is best implemented producing the so-called "new music" of the past 25 years. Contrast *Switched on Bach* with *Silver Apples of the Moon,* Morton Subotnick (Nonesuch H-71174), which is performed on Buchla's instrument.

II. DIGITAL SYNTHESIZERS

Some of the analog synthesizers already discussed are actually analog/digital hybrids, essentially analog but with some digital technology. The Crumar DS-2 and Buchla's synthesizers use digital oscillators to ensure extreme pitch reliability. Various synthesizers that are otherwise analog use microprocessor technology to store patch memory information. The method used in the Oberheim four-voice and Octave Cat SRM II to scan the keyboard is digital.

When people speak of digital synthesizers, they generally mean the very expensive ones about to be briefly discussed. However, since the mass acceptance of microcomputers, other less expensive approaches to digital synthesis have become available: the control by a microcomputer of an analog synthesizer, and the creation of both hardware and software that allows the microcomputer itself to generate user-programmable sounds. *Polyphony* (see Bibliography) has had an ongoing discussion about the first of these for several years.[11] Familiarity with the magazine *Creative Computing*[12] would also be very helpful in this area. Software and hardware for creating digital synthesizers is offered for the Apple II,[13] Radio Shack TRS-80[14] and PET[15], microcomputers. Texas Instruments and Atari each manufacture a cartridge containing software for entering and playing music.

Recall from Figure 4-1 how the first twenty harmonics of a violin string varied over 3/10 second and how difficult that would be to duplicate with analog equipment. Digital synthesizers sometimes have over 100 wave generators and even more envelope generators; they possess the ability to create each harmonic individually and then synchronize them in real time for great precision in instrumental simulation. However, some synthesizer functions that are continuous (rather than discrete) by their very nature (like a sine wave or portamento) lend themselves more easily to production by analog equipment.

The first (and still the least expensive) digital synthesizer to become commercially available was the RMI KC-II, which has neither oscillators nor filters; it generates waveshapes directly in its computer by assembling the contours in small digital segments. The digital computer drives a stereo digital-to-analog converter. There are 38 individual

[11] See especially vol. 5, no. 6 (March-April 1980).

[12] P.O. Box 789-M, Morristown, NJ 07960. See especially vol. 6, no. 6 (June 1980).

[13] Information from Syntauri, Ltd., 3506 Waverly Street, Palo Alto, CA 94306; Mountain Hardware, 300 Harvey West Boulevard, Santa Cruz, CA 95060; American Micro Products, Inc., 705 North Bowser, MS 107, Richardson, TX 75080; ALF Music Products, 1448 Estes, Denver, CO 80215; and Passport Designs, 785 Main Street, Suite E, Half Moon Bay, CA 94019.

[14] Information from Software Affair, 473 Sapena Court, #1, Santa Clara, CA 95051; and Newtech Computer Systems, Inc., 230 Clinton Street, Brooklyn, NY 11201.

[15] Information from Electronic Music Systems, 45 Livingston Road, Suite 501, West Hill, Ontario, Canada MIE IK8. See also Grokett, "PET/Muse: "Interfacing PAIA Synthesizers with PET Computers", Polyphony, Sept./Oct., 1978, p. 40; and Grokett, "PET's Built-In Synthesizer", Polyphony, May/June, 1979, p. 10.

RMI KC-II

and permanent waveforms stored in the system; computer cards may be used to supplement the permanent waveforms. There are 12 presets.

The Synclavier® II is an eight-voice digital synthesizer, which can be expanded up to 128(!) voices. Its unique "partial timbre" method of synthesis allows extremely realistic creation of standard instrumental sounds, as well as a variety of "other" sounds. A partial timbre consists of 24 separately adjustable harmonics, a volume envelope generator, a harmonic envelope generator, a completely adjustable vibrato control and portamento rate, and other special effects. There are as many partial timbres as there are voices (e.g., 16 on a 16-voice machine). Up to four separately adjustable partial timbres (96 harmonics) can be triggered from just one key on the keyboard. The Synclavier® II also includes a 16-track digital memory recorder.

Synclavier® II

The Fairlight C.M.I. can output up to eight separate voices. It comes with a high-resolution video graphic monitor (like a television screen) on which can be diagrammed in great detail each envelope of any parameter of a given sound. Then the user can alter that parameter with a "light pen," a device that is held like a pencil next to the video graphic monitor and that can actually change the shape of the envelope (and thus the sound created) by the user's moving the pen next to the screen in the shape of the envelope desired. It's like an oscilloscope in reverse; the musician draws the waveform desired and the Fairlight C.M.I. produces the appropriate sound. Any acoustic sound can be input to this instrument through a microphone; the Fairlight's "waveform memory" instantly analyzes the sound and does a very credible job of reproducing it and allowing it to be modified and controlled digitally.

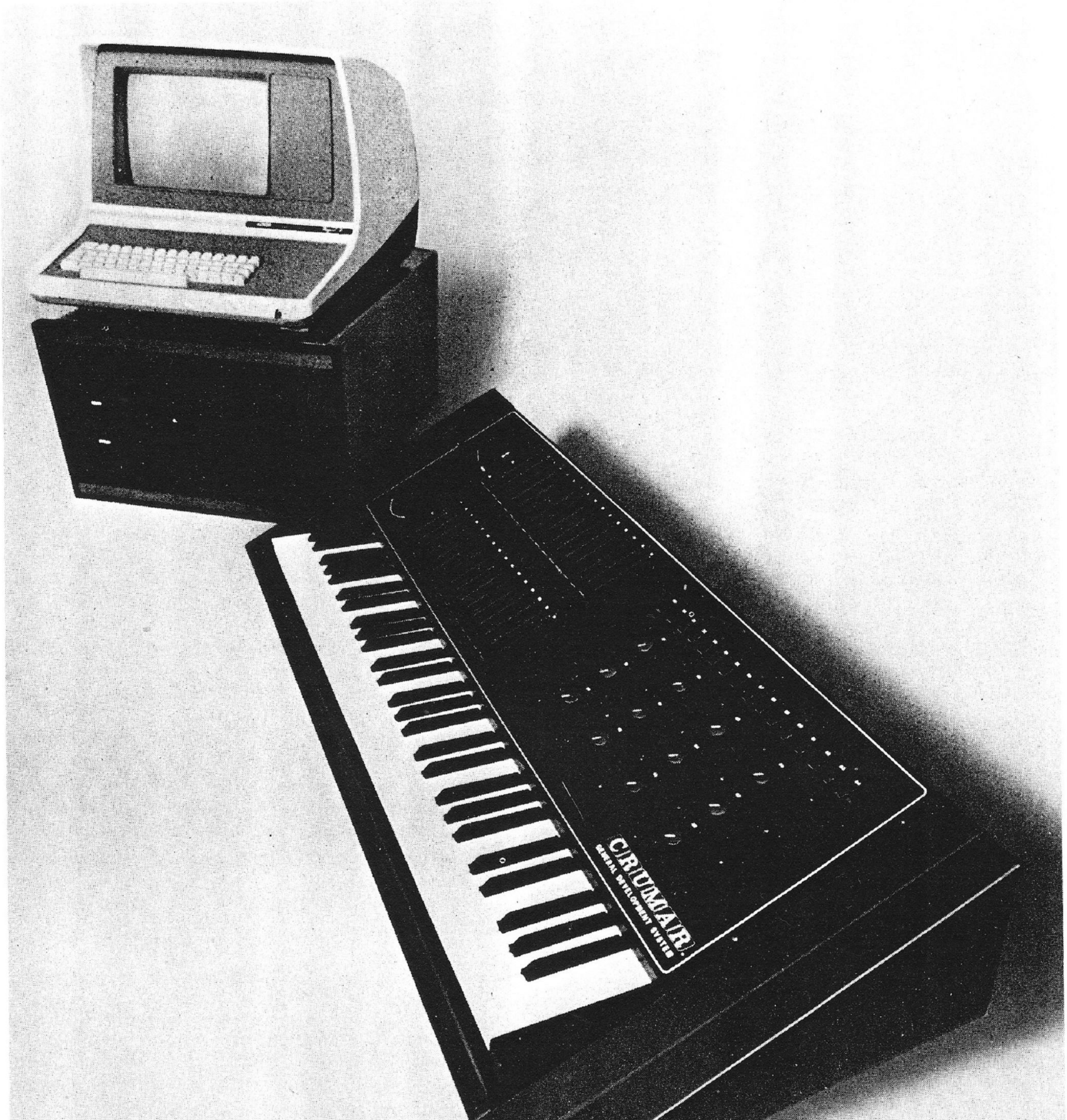
The Con Brio ADS 200 has 64 multi-waveform digital oscillators, 128 envelope generators (each of which has 16 stages), and a video screen that presents a visual representation of individual envelopes that change in real time as the musician changes the sound. Its 8″ floppy disk allows the storage of any number of fully polyphonic tracks, up to four of which may be played back simultaneously.

Fairlight C.M.I.

Con Brio ADS 200 Digital Command Console

The General Development System (GDS) has 32 digital oscillators, up to 16 of which may be used in the creation of each voice; it outputs a maximum of eight voices. Its digital memory is eight channels. An interesting concept incorporated in the GDS is that of "timbral interpolation." The unit can interpolate a variety of timbres within a single voice; which timbre is heard depends upon how hard the velocity-sensitive keyboard is struck. Rather than frequency and amplitude envelopes being specified as single curves, they are defined as "regions" limited by "upper-bound" and "lower-bound" envelopes. When a key is struck, values are selected between the upper and lower bounds proportionate to the key velocity. Up to 32 separate timbres may be heard between the two boundaries, depending on how hard the key is struck.

E-mu's Audity offers complete computer control of 16 totally separate analog synthesizers. Each of these analog synthesizers contains two VCOs, voltage-controlled low and high-pass filters as well as a multimode resonant filter, four ADSR envelope generators, noise source, LFO, and four independent modulation paths. Because the Audity's computer controls each synthesizer separately, it is capable of producing completely different voices simultaneously, as did the Oberheim 4-voice with programmer of an earlier generation. The Audity also has a 16 channel polyphonic keyboard and sequencer.

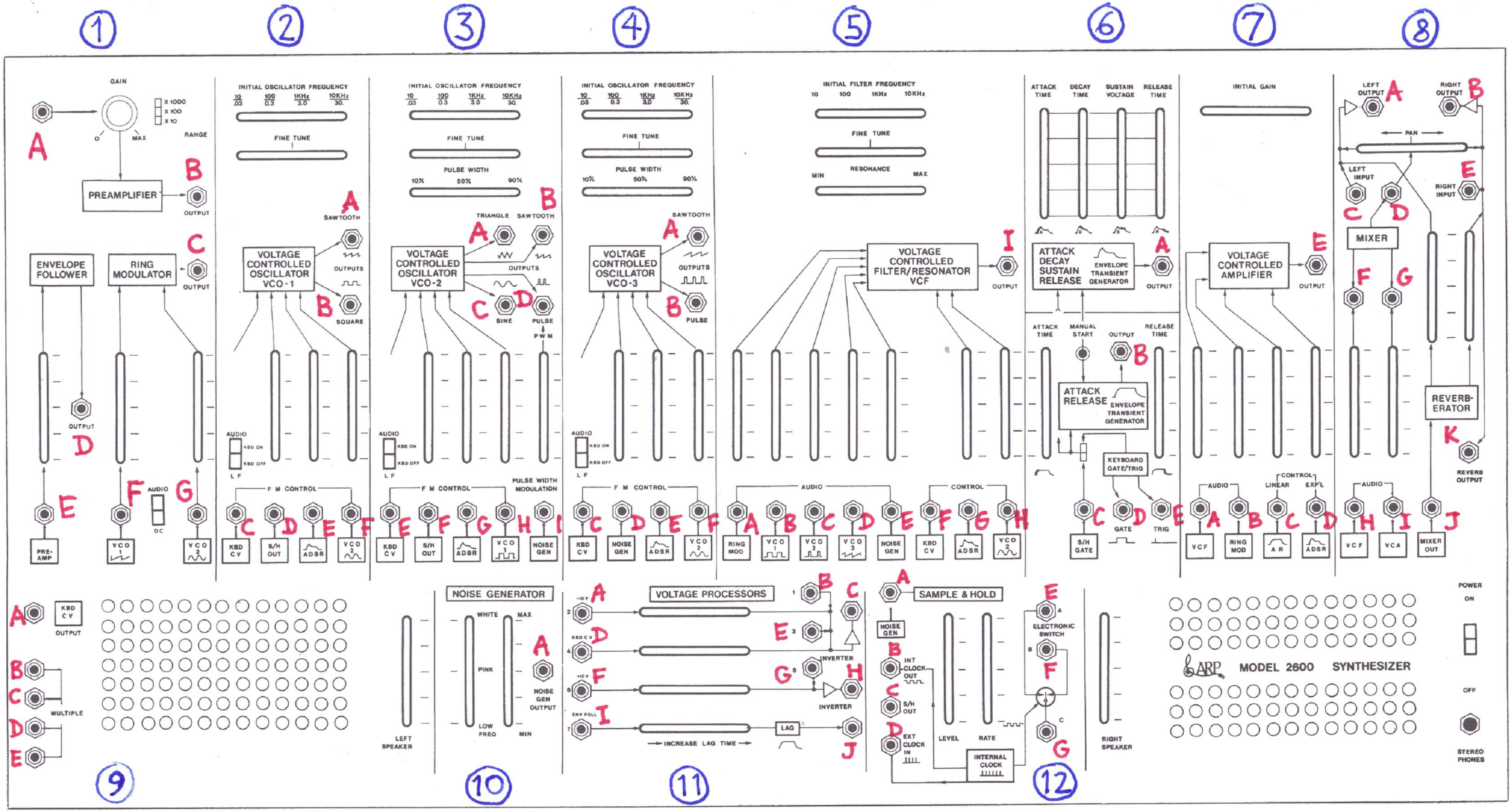


GDS

Like the Audity, Buchla's Touché combines both analog and digital circuitry. Its 24 digital oscillators are combined into eight voices that are playable in a variety of polyphonic, split keyboard, and multi-instrument modes. Additionally, the Touché contains a specialized processor that directs the progress of 64 acoustic parameters, each with a time resolution of 1/1000 of a second. Up to 32 labeled instrument definitions are instantly accessible, with additional definitions being stored on tape for subsequent retrieval.

E-mu Audity

Buchla Touché



This is a diagram of the face panel of the ARP 2600 synthesizer. Notice that twelve separate sections are noted. The first eight of these go from left to right at the top of the diagram and there are four more sections in the bottom third of the face panel. Jacks within a section are given a letter, beginning with A, at the top of each section and going down to the last jack in the section. Jack 11-J, for example, would refer to the jack at the bottom middle of the synthesizer to the right of the word LAG.

In this appendix references will frequently be made to jack numbers, to facilitate understanding. You should not think of these jacks by number (as, for example, jack 5-F), but rather by function (the unattenuated control input to the filter).

appendix a

The ARP 2600

I. BLOCK DIAGRAMS ON THE 2600 FACE PANEL

Look at the face panel of the 2600, specifically the VCF module. You will notice a series of arrows. Those arrows that go from an attenuator to the left side of the block representing the VCF indicate that the inputs are audio inputs; those that go to the bottom of the block representing the VCF indicate that they are control inputs.

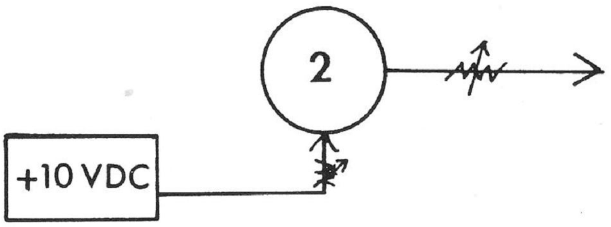
The Arp 2600

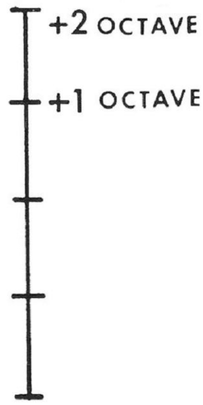
You will notice that many of the modules have arrows similar to those in the VCF. All arrows going into the left side of a module symbol indicate an audio path; those going into the bottom indicate a control path; and those exiting on the right indicate an output. Thus the face panel of the 2600 is itself set up as a large block diagram showing *signal flow* throughout the synthesizer.

EXPERIMENT #A-1: VCO's control attenuators

One source of DC voltage on the 2600 is from the Voltage Processor section (section 11). When the attenuator between jacks 11-A and 11-C is all the way to the right, +10 volts DC will appear at jack 11-C.

Put the attenuator between jacks 11-A and 11-C all the way to the right. Patch a 100 Hz. sawtooth wave from VCO-2 (section 3) directly out. Now put one end of a patchcord in jack 11-C and the other in jack 3-H. +10 volts is appearing at jack 11-C so you might expect that raising the attenuator over jack 3-H into VCO-2 would have the effect of inputting 10 control volts into VCO-2, raising the frequency by ten octaves, well above audible hearing. Slowly raise the attenuator over jack 3-H.





The maximum frequency deviation is, however, two octaves; that is, fully raising the attenuator over jack 3-H does not allow 10 control volts into VCO-2, but only 2 volts. Note also that you have raised the attenuator about three-quarters of the way when you reach the first octave above 100 Hz. These control voltage attenuators are exponential, which means that there will be less deviation from your input signal in the lower ranges of the attenuator than in the upper. This gives you an opportunity for fine tuning, since you have a lot of room to maneuver. If you wanted to input a control voltage that varied by 2 volts or less, this would be the control input to use.

Keeping everything else the same, lower the attenuator over jack 3-H. Take the plug out of jack 3-H and insert it in jack 3-G. Slowly raise the attenuator over jack 3-G. Notice that now the frequency is raised a total of about four octaves, indicating that the attenuator allows a maximum of about 4 control volts to pass through to the CV mixer.

Do the same procedure once more with jack 3-F and its associated attenuator. Now the frequency is raised beyond the range of hearing. That attenuator allows almost 10 volts to pass.

Finally insert the plug in jack 3-E, the unattenuated control input. The frequency immediately goes beyond audibility. That input allows the full 10 control volts to pass.

II. RECOMMENDED MODIFICATION

A serious design limitation of the ARP 2600 is that if one envelope generator is gated then the other is *necessarily* gated simultaneously. An inexpensive and recommended modification is to give the AR a separate gate input.

III. ENVELOPE GENERATOR INPUTS AND OUTPUTS

Patchable synthesizers generally have accessible gate and trigger inputs to and outputs from the envelope generators. On the 2600 there is a pre-patch from the keyboard to the input of both envelope generators. Unlike the case with other pre-patches, however, inserting a patchcord into jack 6-D (hereafter referred to as the gate input) or jack 6-E (the trigger input) does *not* defeat the pre-patch from the keyboard.

Since inputting a pulse into the gate input will not supply a trigger, doing so will cause the ADSR to function as a three-stage envelope generator.

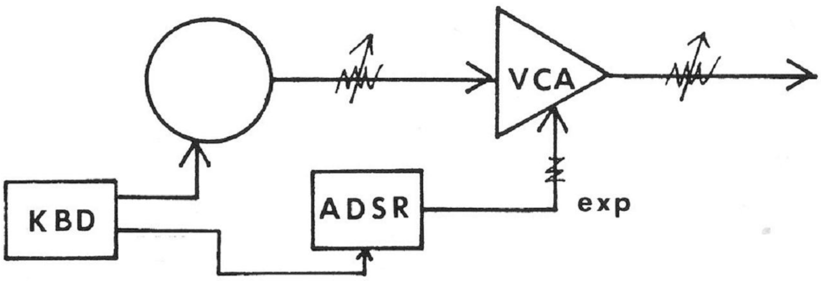
On the 2600 the gate and trigger jacks are also outputs for the keyboard gate and trigger. You can trigger other functions from the keyboard by patching these jacks, as outputs, to other inputs.

The function of jack 6-C (which is unique to the 2600) is, in a sense, determined by the switch about an inch above it. In typical live performance settings the switch would be up; when the switch is up the envelope generators are controlled by the keyboard (as has been the case in all previous examples). When the switch is down, however, the keyboard no longer controls when the envelope generators fire; the internal pre-patches from the keyboard timing signals to the envelope generators are defeated. Instead, the envelope generators fire at a rate determined by the rate attenuator of the internal clock. In other words, the internal clock is pre-patched to jack 6-C but does not become available until the switch is down.

Circuitry behind jack 6-C is such that both a gate and a trigger can be derived from any LF pulse wave. Since the internal clock is nothing more than an LF square wave, it can serve as a source of gates and triggers to the envelope generators when the switch is down.

The internal clock pre-patch can itself be defeated by insertion of a patchcord into jack 6-C. An LF pulse can be input into jack 6-C and the frequency of that oscillator will determine how fast the envelope generators fire.

EXPERIMENT #A-2: The function of jack 6-C on the 2600



Set up the patch shown here which is a typical straightforward way to get discrete notes (events) when the keyboard is played. The keyboard timing signals tell the ADSR when and for how long to fire.

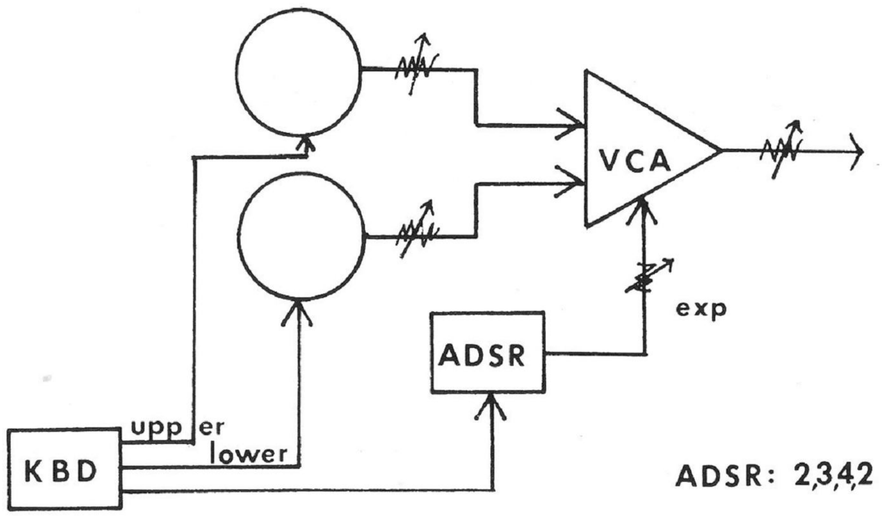
Flip the switch above jack 6-C down. Wait about 30 seconds and then listen. You will hear the VCO sound and then silence, followed by sound again. It's a typical LF-square-wave-controlling-the-VCA effect. If you try playing the keyboard, you will find the keyboard timing signals no longer control firing of the envelope generators. However, if you raise the rate attenuator of the internal clock, you will be able to change the frequency of the LF square wave.

Now patch an LF pulse from any available VCO into jack 6-C. The internal clock has been defeated (as moving the rate attenuator will prove to you), and the frequency of the LFO patched into jack 6-C will determine the frequency of the ADSR firing.

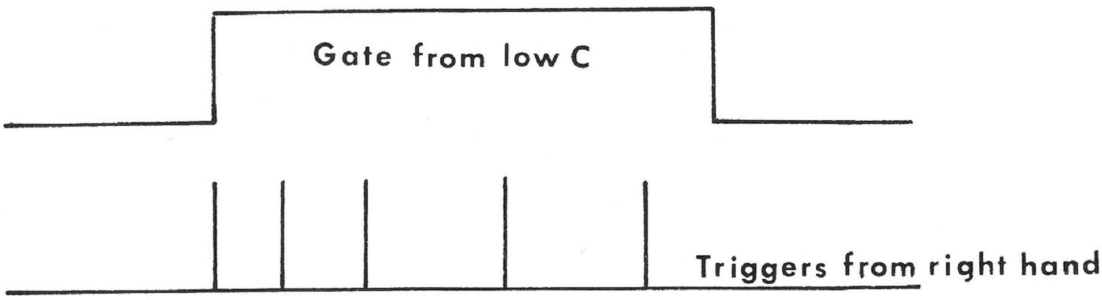
You can reinstitute voltage control of the LF pulse by inserting a patchcord between the KBD CV output and the KBD CV input of the LF VCO. Play the keyboard with a voltage-controlled LFO. Experiment with the difference that duty cycle of the pulse makes in the firing of the ADSR. Voltage-control the pulse width if the LFO has that capability.

EXPERIMENT #A-3: "Duophonic" keyboards

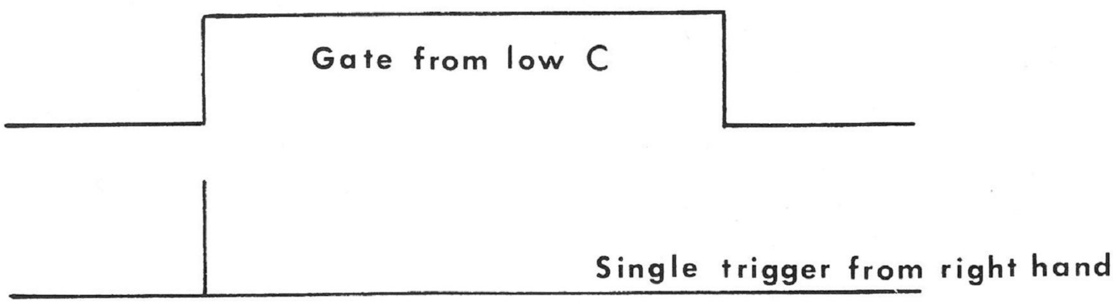
If your keyboard is "duophonic," set up the typical performance patch shown in this diagram:



With the trigger mode switch in the Multiple position, play low C with your left hand, and while keeping that key depressed, play a lead line with your right hand.



Flip the trigger mode switch to Single and repeat the same exercise. Hear the slur effect in the single trigger mode.



IV. CONTROL INPUTS TO VCOs

If you are controlling a 2600 VCO with an envelope generator, the effect will depend upon which control input you are using (see Experiment #A-1).

V. MIXERS ON THE ARP 2600

The Voltage Processors in section 11 have several capabilities, one of which is mixing. Jacks 11-A, 11-B, 11-D, and 11-E are four inputs of a 4 × 1 mixer. The output is jack 11-C. The inputs at jacks 11-A and 11-D are attenuated, the attenuators being the horizontal attenuators just to the right of these two jacks. Normally if you use the Voltage Processors to mix *audio* signals, use the attenuated inputs; if you are mixing control voltages, you might want them attenuated or unattenuated. In the latter case use the inputs at jacks 11-B and 11-E. Of course you can also combine these mixing inputs in any way, as well as mixing AF and LF voltages.

There is a second, independent mixer in the Voltage Processor section. Jack 11-F is an attenuated input; jack 11-G is an unattenuated input; jack 11-H is the output.

The circuitry associated with both mixers includes an inverter, discussed in greater detail in section VII. This means the output voltage is exactly opposite at every point to the input voltage (Fig. 3-20). Depending on the signals inputs, this may or may not matter.[1] In general, inversion will make less of a difference mixing audio than mixing control signals.

EXPERIMENT #A-4: The mixing function of the voltage processors

Patch AF waves from two VCOs into the attenuated inputs of Mixer #1 (the 4 × 1 mixer) in the Voltage Processor section. Open the attenuators all the way. Patch the output (jack 11-C) out (i.e., to either jack 8-H or jack 8-I). Raise the attenuator over the jack.

You should be hearing both AF waves, completely full since the associated attenuators are wide open. Close each attenuator part-way and vary them, so that you can verify for yourself that you can allow more or less of one AF wave to be mixed with more or less of the other.

Now patch a third AF wave from the remaining VCO to jack 11-B or 11-E, one of the unattenuated inputs to the mixer.

Hear the effect of mixing an AF wave with no opportunity for attenuation; the third wave will probably overpower the other two. (You could use a floating attenuator and attenuate the third wave as well.)

Repeat this experiment using the second mixer in the Voltage Processor section.

We have often used the output mixer, specifically called the "Mixer" in section 8 of the diagram. Figure A-1 is a block diagram of this mixer and its associated circuitry. As you know, signals can be input to jacks 8-H and 8-I, be attenuated to your taste, mixed and output. These jacks respectively have the VCF and VCA outputs pre-patched to them. As usual, these outputs will be defeated by insertion of a patchcord from the output of any other module.

There are two jacks (8-F and 8-G) between the input attenuators and the mixer itself. These jacks are outputs from the attenuators and permit them to be used as "floating attenuators."[2] If these jacks are used to make floating attenuators, the pre-patch to the mixer is defeated.

[1] If it does matter, you can make your output signal uninverted, by *inverting the inverted output*.

[2] See section VI.

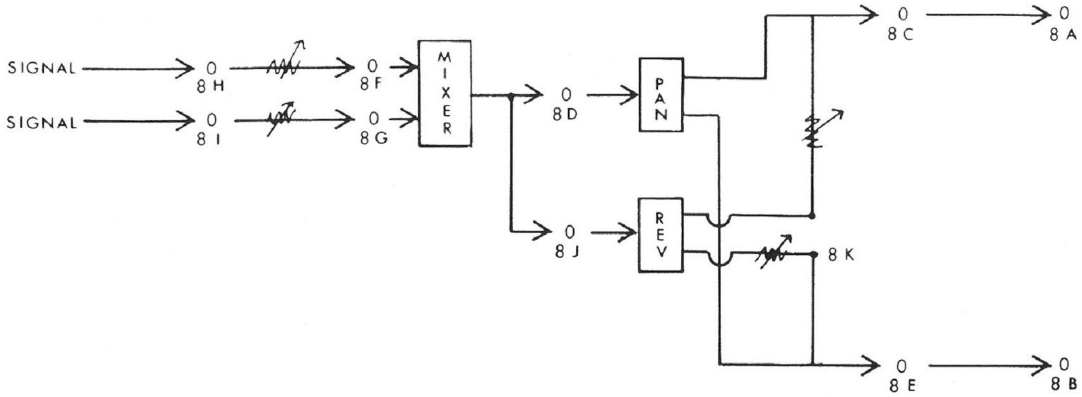


FIG. A-1

Assuming no plug is inserted in jacks 8-F or 8-G, the signals are mixed and then routed to the pan jack (8-D) and the reverb input (8-J). Any signal inserted into jack 8-D will defeat and supersede the signal coming from the mixer. What goes to the pan attenuator and the reverb input will be *either* the mixer output *or* the signal inserted into jack 8-D, but not both. Whatever signal has reached the pan attenuator can be distributed within the stereo field to taste by sliding the pan attenuator more to the left or right. If the attenuator is in the middle, the right-left distribution will be equal.

Jacks 8-C and 8-E provide both a mixing and a binaural capability after the signal has left the *panpot* but before it goes out. Signal patched into jack 8-C ("left input") will be mixed with the mixer output, if any. However, the signal patched into jack 8-C will appear only on the left side of the stereo field. The same is true of jack 8-E ("right input"), only that signal will appear only on the right side of the stereo field. Since neither jack 8-C nor 8-E has an associated attenuator, the signal will appear fully; to attenuate it to have the proper mix, use a floating attenuator.

Note also that you can bypass the mixer entirely and have true *binaural* (not stereo)[3] sound by inserting one audio signal into jack 8-C and another into jack 8-E.

Jacks 8-A and 8-B are the final outputs from the 2600. Although any output on the 2600 can be patched directly out, these are the typical outputs to use when using an external amplifier.

Recall that after leaving the mixer the signal is also internally routed to jack 8-J labeled "Mixer Out." This is more properly thought of as Reverb In, although the output from the mixer is pre-patched to this jack. As usual, insertion of any patchcord will defeat this pre-patch. Whatever signal goes through jack 8-J will get reverberated to whatever level the reverb attenuators are set. As the arrows on the face of the 2600 show, the signal going through the left attenuator will be mixed with the panpot output and anything that may be plugged into jack 8-C and then sent to the left output. Although the same is true of the right attenuator (except that its output goes to the right output), it gets routed to jack 8-K, the reverb output, as well. The signal from jack 8-K may be further processed. Inserting a plug in jack 8-K disables the signal path from going its normal route out; that is, if a plug is inserted in jack 8-K, the right reverberated signal will go only to where that patchcord takes it and will not also go out the right output.

VI. ATTENUATORS

The following inputs do not have associated attenuators: KBD CV on all VCOs and on the VCF; timing signal inputs to the envelope generators (jacks 6-C, 6-D, and 6-E);

[3] With stereo, one source is miked from two separate points; binaural involves two separate sources being heard from two separate points.

sample and hold (jack 12-A); external clock into the sample and hold module (jack 12-D); the electronic switch (jacks 12-E, 12-F, and 12-G); the panpot (jack 8-D); left and right inputs (jacks 8-C and 8-E); and the reverb input (jack 8-J). The only output attenuators on the 2600 are associated with the reverb and noise generator. Thus virtually all attenuators are associated with inputs.

There are five possible floating attenuators on the 2600; they are the attenuators between jacks 8-H and 8-F, jacks 8-I and 8-G, jacks 11-A and 11-C, jacks 11-D and 11-C, and jacks 11-F and 11-H. The three latter have inverted outputs.

VII. INVERTERS

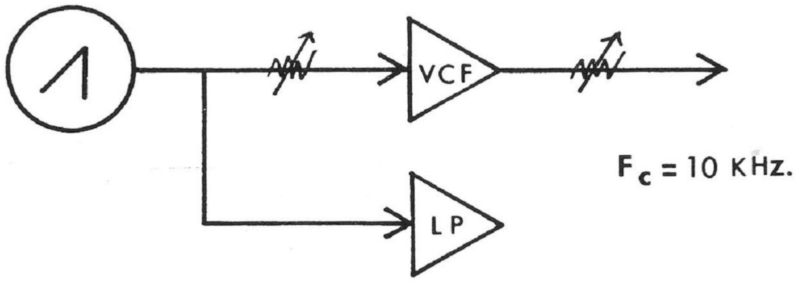
There are three inverters on the 2600. The two most often used as such are in the Voltage Processor section. Jacks 11-A, 11-B, 11-D, and 11-E input to jack 11-C, an inverting output. As you know, jacks 11-A and 11-D are attenuated. The attenuation of those inputs takes place *before* the inversion. Thus if, for example, you input a +10 volt p-p sawtooth wave into jack 11-A, you could attenuate it to, say, +5 volt p-p and then when you brought it out of jack 11-C it would be a 5 volt inverted sawtooth wave. That voltage could be mixed with as many as three other voltages and all would be inverted at the output, jack 11-C.

The second inverter in the voltage processor section is at jack 11-H, whose inputs are jacks 11-F and 11-G. The attenuation of jack 11-F's input takes place before the inversion.

The third inverter on the 2600 is the audio input to the VCA at jack 7-B.

At some time you might be using the Voltage Processor section as a mixer but not want the output inverted. Since inversion is equivalent to a multiplication by -1, then inverting the inversion would be the equivalent of multiplying -1×-1, giving you an output of $+1$, i.e., a non-inverted output. Patch the output of the second inverter anywhere you want it to go.

EXPERIMENT #A-5: *The 2600 lag processor filters with no output connection*



First patch VCO-3 through the open VCF and listen to the sound of an unfiltered sawtooth wave.

Still listening to that wave, patch VCO-3's sawtooth output to jack 11-I, the lag processor input. Doing nothing else, you will immediately hear that the sawtooth has been filtered, even though there is no lag processor output connected. This is a design peculiarity.

If you patch the lag processor out (that is, to jack 8-I), you will be able to further attenuate the harmonics of the sawtooth by sliding the attenuator between jacks 11-I and 11-J to the right. That attenuator, labeled "Increase Lag Time," is the F_C control for this -6 db/octave low-pass filter.

VIII. SOURCES OF DC VOLTAGE

There are two primary sources of DC voltage on the 2600; these appear at jacks 11-A and 11-F. Any voltage that does not vary is DC. The DC sources referred to here are non-varying, so that instead of going from 0 to +10 volts they remain steady at, for example, +5 volts, or −4 volts.

Notice that directly above jack 11-A is written "−10 volts." A DC source of −10 volts appears there and, just like any other pre-patch, will be defeated if a patchcord or dummy plug is inserted in jack 11-A. If it is not defeated, sliding the attenuator to the right of jack 11-A all the way to the right will allow −10 volts to appear at jack 11-C. However, at that point inversion takes place, so the final output at jack 11-C is +10 volts. Therefore if you want a source of positive DC voltage, use jack 11-C even though jack 11-A says "−10 volts" above it.

Similarly, although +10 volts appears at jack 11-F, by the time it appears at jack 11-H (when the attenuator to the right of jack 11-F has been opened) it becomes inverted and is finally output as −10 volts. If you want a source of negative DC voltage, use jack 11-H, even though jack 11-F says "+10 volts" above it.

IX. THE MODEL 3604 KEYBOARD CONTROLS

The three attenuators over the Portamento and Transpose sections are the controls for an LFO built into the keyboard and which is pre-patched to the VCOs and the VCF. Since this would typically be vibrato, the rightmost of these attenuators is labeled "Vibrato Depth." If not defeated as described below, this attenuator determines the amount of LF sine wave that will control the VCOs. If the Vibrato Delay attenuator is all the way down, then vibrato will begin as soon as you depress a key; raising that attenuator delays the time (to a maximum of about 2⅓ seconds) when vibrato takes effect. Most acoustic instruments have *some* delay before vibrato begins. The speed of vibrato (or other LF control) is determined by the LFO speed attenuator.

EXPERIMENT #A-6: The keyboard control attenuators

Set up any patch by which you can use the keyboard to create sound as in a live performance situation (see, e.g., Experiment 46). Initially set the LFO speed attenuator to level 2¾ and while playing with your right hand vary the Vibrato Depth attenuator. After getting the feel of that, slide the vibrato delay attenuator to level 1 or 2, and note that vibrato begins only after you have held a note down a moment or two. Further experimentation with this section will show you how easy it is to effectively add vibrato while playing in real time.

Directly above the three LFO control attenuators are four jacks: the ones labeled LFO triangle, square, and sine are outputs whose frequency is determined by the setting of the LFO speed slider. If, for example, you need an LF sine wave but VCO-2 is in use, you can patch the keyboard sine wave output to control any module needed. The same is of course true of the triangle and square wave outputs.

The jack marked Ext. Vibrato In allows any externally applied voltage to act as a control voltage to the VCOs and the VCF. It is *not* limited to an external vibrato, as marked.

EXPERIMENT #A-7: "External Vibrato In"

Patch an AF wave from VCO-3 out. Put VCO-2 in its LF and patch a sine or triangle wave from VCO-2 into the Ext. Vibrato In jack. As you play the keyboard, you will find that raising the Vibrato Depth attenuator allows more and more of VCO-2's wave to control VCO-3. You will also find the LFO speed attenuator has been defeated (raise it and you'll see that the vibrato frequency does not change; to change that frequency you would have to change the frequency attenuators on VCO-2). Now patch an LF pulse or sawtooth wave from VCO-2 into the Ext. Vibrato In jack. The LF pulse will cause a trill, the sawtooth its unique sound of controlling a VCO. Also experiment with putting VCO-2 in its AF and otherwise using the same patch. As you can see, the Ext. Vibrato in jack allows far more than merely the inputting of an external vibrato.

To the left of the four jacks just discussed are the two "Upper Voice" jacks. Either one of these jacks will provide the "upper" keyboard control voltage as shown in Experiment #A-8. The fact that there are two jacks in the "Upper Voice" section might lead you to believe that connecting each to the KBD CV of a VCO would give you a three-note capability. This is not true. In some filter settings when two voices are used the filter will not track the upper voice at 1 volt/octave. The effect would be that you would hear one VCO brighter than another. The remedy is to patch the second upper voice jack into the KBD CV input to the filter; *that* is why there is a second Upper Voice jack.

The upper voice may be used in combination with the Interval Latch jack just to the left of the Upper Voice section on the keyboard.

EXPERIMENT #A-8: "Upper Voice" jacks

Set up a typical performance patch. Patch an Upper Voice jack to the KBD CV input of VCO-1 and then tune VCO-1 to unison with VCO-3. Patch the other upper Voice Jack to the KBD CV input of the VCF. Plug the foot pedal into the interval latch jack and place the pedal conveniently near your foot.

Play a keyboard line; occasionally play two notes and note that, as patched, two notes will *play simultaneously. The keyboard is* duophonic. *Play any interval (say a fifth or an octave) and, while depressing both keys, step on the foot pedal. Keep the pedal depressed and now play the keyboard with one finger (i.e., play only one key). The keyboard will hold (latch on to) the interval selected as long as the pedal is depressed. This makes it much easier to play lead lines with two pitches at a given interval. As soon as you release the foot pedal, the VCOs will return to unison and you can then play a single note (two VCOs in unison) lead line.*

The use of the Single–Multiple switch was discussed in section D-6 of Chapter 3. Typically it will be in the Multiple position.

The Repeat switch allows the LF square wave within the keyboard section to be activated in one of two ways: when the switch is in the KYBD position, it will gate the keyboard at a rate determined by the setting of the LFO speed slider whenever a key is depressed. In the automatic position the keyboard is continuously gated whether or not a key is depressed. In order to appreciate this effect, the attack and release settings of the envelope generator controlling the filter or VCA should be at or near minimum.

EXPERIMENT #A-9: The repeat switch

Set up the performance patch shown in Fig. 2-5, with these exceptions: There should be no LFO control of the filter (it's not going to be used), and the ADSR settings should be 0,2,3,0.

Set the LFO Speed attenuator to level 3. Play the keyboard and at some point, while playing, flip the repeat switch to KYBD. Note that the sound you hear is a sound like a VCA being opened and closed by an LF square wave. The frequency of the gates is determined by the position of the LFO Speed attenuator.

Now flip the Repeat switch to Auto. The gating effect will continue whether you play the keyboard or not.

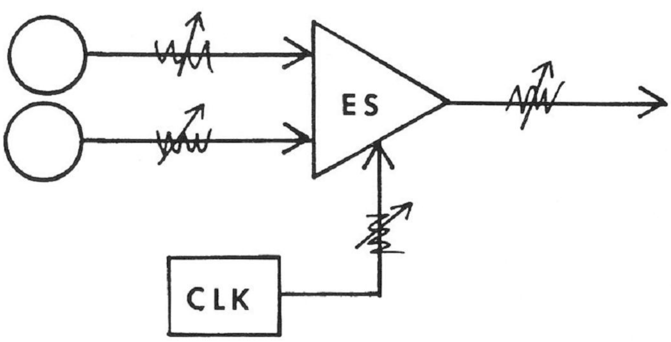
You must be careful not to have the attack and release times set too high or the repeats will come so often that the attack and release cycles haven't had a chance to complete; in that case you wouldn't hear the repeats. To experience this, set attack and release attenuators at level 3 each and hear what happens.

X. ELECTRONIC SWITCH (ES)

This module either (a) has two inputs and supplies one output, or (b) has one input and supplies two alternate outputs.[4]

EXPERIMENT #A-10: The electronic switch

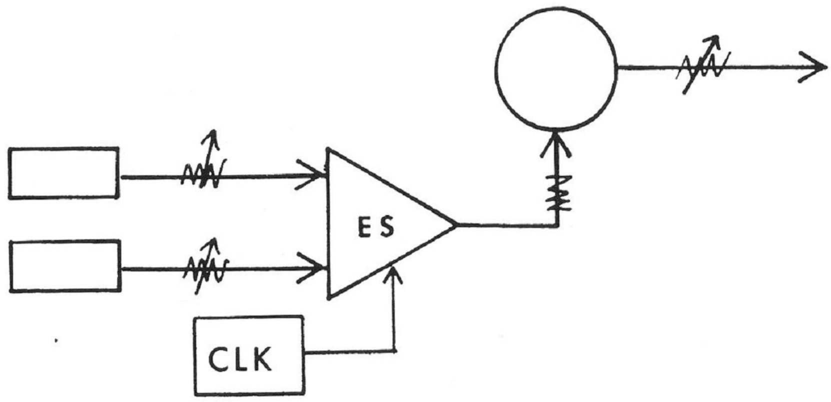
This patch will demonstrate that an ES can accept two inputs and alternate them to a given output. Tune two VCOs to an interval of a fifth. Remember that the rate of switching is controlled by the rate attenuator of the internal clock; set it to level 1 at first.



[4] However, there are different varieties of electronic switches. For example, the Aries AR-335 has four different switches: two are 2/1 as in the example here; a third is a 4/1 switch, and a fourth switch is controlled by a "window detector" which senses various threshold voltages and routes the switch output to different places depending on what that voltage is.

The jacks for the 2600 electronic switch are bi-directional; this means that they can function as either inputs or outputs. If you feed them outputs, they will function as inputs; if you feed them inputs, they will function as outputs.

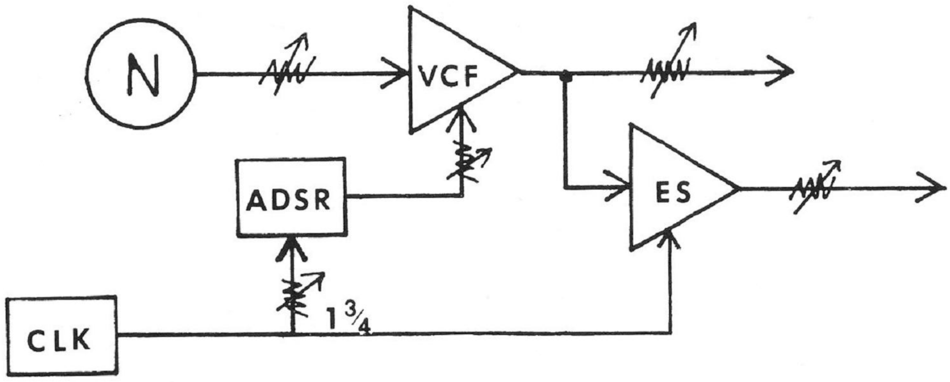
EXPERIMENT #A-11: Electronic switch as control voltage processor



In this patch the switch is routed to control a VCO so that you can best hear what effect the switch is having; any voltage-controllable parameter may be controlled. Naturally the LF waves might be further processed before being input to the switch, or the switch output further processed before controlling a parameter.

EXPERIMENT #A-12: A $\frac{2}{4}$ Meter

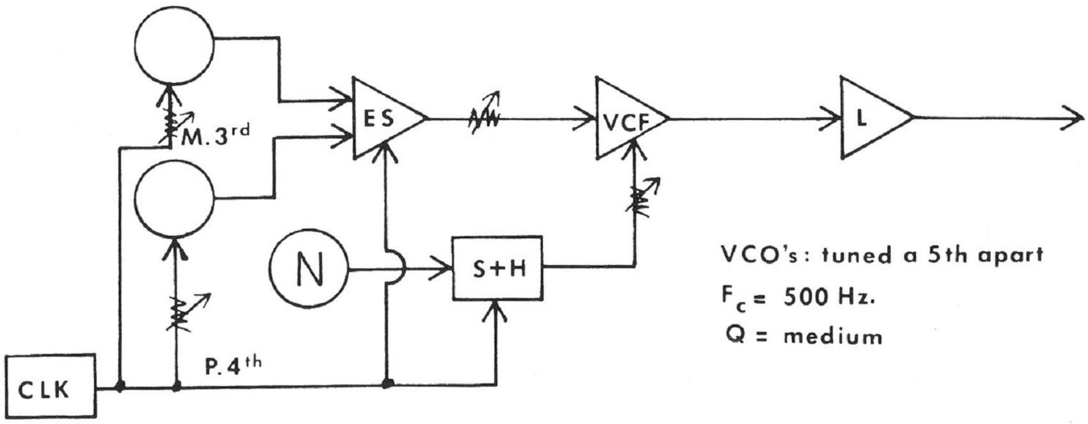
It is possible to patch only one output of the switch out and create a synchronized metrical relationship between the original signal and the signal input to the switch. This gives the possibility of a $\frac{2}{4}$ meter.

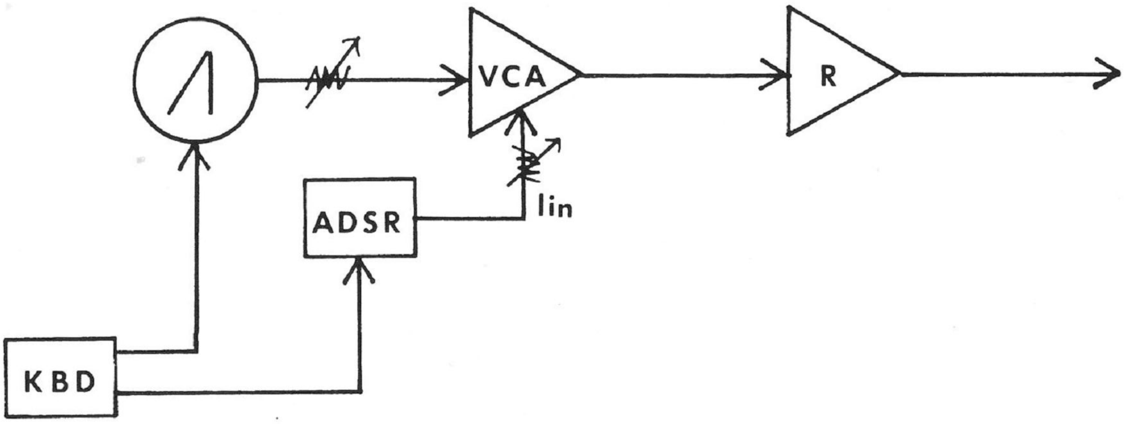


The VCF output is routed both out and to one input of the switch. The other input is left empty, and the ES output also goes out. Thus the switch alternately outputs a sound from the VCF and then nothing. Since this is attenuated into the output mixer, you can mix it with the regular VCF output and have every other beat accented. (If there are audible clicks from the switch as it goes back and forth between sound and nothing, just lag the switch output a bit before sending it out.)

EXPERIMENT #A-13: A four-note sequence and independent lead line

This experiment uses several of the applications you have already learned and is an example of the interdependence of the various modules.





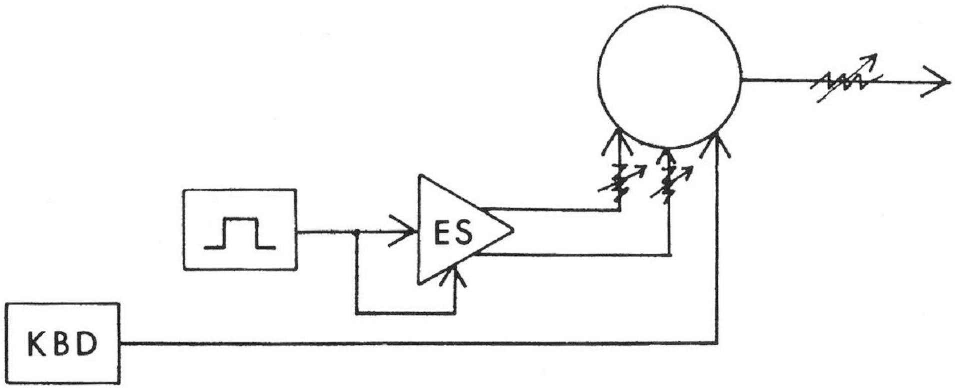
In the patch there are two entirely separate signal paths, something you saw before in Fig. 5-5. The upper block diagram is the four-note sequence, the lower is the patch for a lead line.

The first thing to be aware of is the written notes to the patch. Here they simply state that VCO-1 and VCO-2 are to be tuned a fifth apart. Since there is no KBD CV indicated, dummy out the pre-patched KBD CV inputs. Now *tune the VCOs a fifth apart. The clock is doing a lot of controlling; you will have to go through the multiple. VCO-1 is to be tuned such that the LF square wave from the internal clock creates a major third, e.g., from C to E; VCO-2 is to be a perfect fourth, e.g., from G to C. Thus the two VCOs play the notes C–E–G–C. They go to two inputs to the electronic switch, which itself goes to an audio input to the VCF. The VCF initial settings are shown in the block diagram. The switch will make the C–E from VCO-1 alternate with the G–C from VCO-2, giving the four-note sequence C–E–G–C. Some notes in the sequence will be randomly accented by the sample and hold module controlling the VCF, causing periodic changes in the* F_C*, in synch with the changing of the notes in the sequence and the switching (since they are all controlled by one LF square wave, the internal clock). Finally this is all routed to the left side of the listening field.*

The lead line patch is quite straightforward; tune the AF sawtooth wave to C, so it sounds good with the four-note sequence; use any ADSR settings that seem musically appropriate to you, patch the whole thing to the right output and play along. You can of course use the keyboard controls for expression and add anything else you like to the patch. (An example: use positive or negative DC *voltage to offset either or both VCO-1 and VCO-2 as shown in Experiment 73 and you will be able to change your sequence in real time.)*

EXPERIMENT #A-14: A three-note sequencer using one VCO

In the last experiment you constructed a four-note sequence and had to use two VCOs to do so. Here is a way to have a three-note sequence using only one VCO.



The LF square wave is both the clock and the input signal to the ES. If the control attenuators into the VCO are raised to different levels, then different LF square wave voltages will control the VCO when the switch outputs one or the other. For example, say the VCO is tuned to middle C; one control attenuator is raised such that the LF square wave voltage raises the pitch to G above middle C, and the other to E above high C. The sequence would be C-G-C-E. Note that a C is sounded for each time that either a G or an E is sounded, because the controlling square wave goes low twice as often as the time either one or the other output happens. Of course, playing the keyboard will transpose the entire sequence.

XI. BI-DIRECTIONAL JACKS

The KBD CV output of the 2600 (jack 9-A) is bi-directional: it can function as either an input or an output. It will be an input if an output voltage is plugged into it; it will function as an output jack if voltage is input to it.[5] Therefore to control one 2600 by another, patch jack 9-A of one (as a CV output) to jack 9-A of the other (as a CV input to all VCOs and the VCF). The gate and trigger jacks (6-D and 6-E) are also bi-directional; patch them as timing signal outputs from one synthesizer to the same jacks as timing signal inputs on the other.

[5] The following are the bi-directional jacks on the 2600: KBD CV (jack 9-A), gate and trigger (6-D and 6-E), and the electronic switch (12E, 12F, and 12G).

appendix b

Synthesizer Manufacturers

Aries Music, Inc.
see Rivera Music Services

Arp Instruments, Inc.
45 Hartwell Avenue
Lexington, MA 02174

Buchla
P.O. Box 5051
Berkeley, CA 94705

Computone, Inc. (Lyricon)
P.O. Box 433
Norwell, MA 02061

Conbrio
975 San Pasqual, #313
Pasadena, CA 91106

Crumar
105 Fifth Avenue
Garden City Park, NY 11040

Electro-Harmonix
27 West 23rd Street
New York, NY 10010

EML, Inc.
P.O. Box H
Vernon, CN 06066

EMS
460 West Street
Amherst MA 01002

E-mu Systems
417 Broadway
Santa Cruz, CA 90506

Fairlight C.M.I.
c/o Bruce Jackson
P.O. Box 401
Lititz, PA 17543

Gleeman Instrument Co.
97 Eldora Drive
Mountain View, CA 94041

HEAR (Zetaphon)
1122 University Avenue
Berkeley, CA 94702

Kinetic Sound Corporation
11 Mary Knoll Drive
Lockport, IL 60441

Korg/Unicord
75 Frost Street
Westbury, NY 11590

MOOG
7373 North Cicero Avenue
Lincolnwood, IL 60646

New England Digital Corp.
(Synclavier)
P.O. Box 305
Norwich, VT 05055

Oberheim Electronics
1549 Ninth Street
Santa Monica, CA 90401

Octave Electronics (CAT)
928 Broadway
New York, NY 10010

PAIA
1020 West Wilshire Boulevard
Oklahoma City, OK 73116

Performance Music Systems
P.O. Box 6028
Bend, OR 97701

Polyfusion
160 Sugg Road
Buffalo, NY 14225

Rivera Music Services
48 Brighton Avenue
Boston, MA 02134

Rocky Mountain Instruments, Inc.
Macungie, PA 18062

Roland Corp.
2401 Saybrook
Los Angeles, CA 90040

Sequential Circuits
3051 N. 1st St.
San Jose, CA 95134

Serge-Modular Music
572 Haight Street
San Francisco, CA 94117

SMS
P.O. Box 40267
San Francisco, CA 94140

Star Instruments (Syndrum)
Box 71
Stafford Springs, CT 06076

Steiner-Parker
2258 South, 2700 West
Salt Lake City, UT 84119

Strider Systems, Inc.
P.O. Box 2934
Norman, OK 73070

Synthi
c/o EMSA
269 Locust
Northampton, MA 01060

360 Systems
(guitar synthesizer)
18730 Oxnard Street, #215
Tarzana, CA 91356

WASP
Aim Ltd.
P.O. Box 424
Orange, MA 01364

Wavemakers
2316 1/2 Fairpark Avenue
Los Angeles, CA 90041

Yamaha International
Box 6600
Buena Park, CA 90620

Glossary

A–B comparison comparison of two things that are generally similar but that have some differences, like two television sets or two AF waves.

AC *alternating current* a fluctuating voltage the sum of whose instantaneous changes through one cycle equals zero volts.

additive synthesis a process by which specific timbres may be constructed by the mixing of sine waves of specific frequency and amplitude.

ADSR an envelope generator with variable timing (*A*ttack, *D*ecay and *R*elease) and voltage (*S*ustain) parameters.

AF *audio frequencies* those frequencies that are audible (capable of being heard) and that we perceive as sound; between 20 Hz. and 20 KHz.

alternating current see **AC.**

AM *amplitude modulation* changes in the strength of an audio wave or its harmonics caused by a control voltage. If the control voltage is of low frequency, the effect is tremolo; if it is audio frequency, then a unique modulation timbre, caused by the creation of sidebands, will be perceived.

amplitude the strength of a signal, generally expressed in volts. E.g., the amplitude of an unattenuated sawtooth wave generated by a typical VCO is either +5 volts or +10 volts.

amplitude modulation see **AM.**

analog synthesizer a synthesizer whose voltages vary continuously rather than in discrete steps. Distinguished from digital synthesizer.

aperiodic random, irregular.

AR an envelope generator with manually variable timing (*A*ttack and *R*elease) parameters, which always sustains at +10 volts.

attack the time of the initial stage of the envelope.

attenuator a signal modifier which determines how much of an input signal will be allowed to pass. Symbol: 

audio frequencies see **AF.**

audio input see **input.**

audio wave a wave whose frequency is between 20 Hz. and 20 KHz., i.e., one we can hear.

bi-directional jack a jack whose associated circuitry allows it to function as either an input or an output.

block diagram a map of audio and control signal flow in a particular patch.

carrier the name given to an audio-frequency signal that is amplitude- or frequency-modulated by another AF signal.

clipping the type of distortion which occurs when a module (like a VCF, VCA, or pre-amp) has more than 15 volts input. The result of clipping is a change in waveshape, thus a change in harmonics and timbre.

clock an LF square wave with manually variable or voltage-controllable rate, generally used as a control voltage.

coarse filter cutoff frequency attenuator a source of +10 DC control volts that determines the initial control voltage in the VCF's control voltage mixer, and thus the initial F_C of the filter.

coarse oscillator frequency attenuator an attenuator that determines the amount of constant DC control voltage in a VCO control voltage mixer and thus the initial frequency output from a VCO. It has a range of 10 volts (ten octaves).

contour generator see **envelope generator.**

control voltage determines the manner in which a voltage-controllable module will output.

cps see **cycles per second.**

cutoff frequency (F_C) the point at which a filter begins harmonic attenuation.

CV abbreviation for control voltage.

cycles per second *cps* the measure of frequency; how many times a periodic wave completes a full cycle each second; measured in Hertz (Hz.).

DC *direct current;* a nonfluctuating voltage.

db see **decibel.**

decay the second stage of an envelope created by the ADSR; the amount of time the voltage generated by the attack stage of the ADSR takes to fall to a preset sustain level.

decibel *db;* the logarithmic unit of amplitude.

digital synthesis the creation, modification, and control of sound by means of a computer.

direct current see **DC.**

distortion the creation or attenuation of harmonics which results when more audio signal is input to a module than it was designed to accept. See **clipping.**

drone (1) a steady tone which accompanies a melodic line; (2) a VCO that does not track the keyboard but sounds the same pitch no matter what key is depressed.

dummy plug the name given to a plug that is inserted into a jack solely in order to defeat the pre-patch at that jack.

duty cycle the ratio of the time a pulse is high to the time it is low. A duty cycle of ⅓ (1:3) means the pulse is high for the first one-third of its cycle and low for the remaining two-thirds.

dynamic range the difference between no signal and the greatest possible signal of a module; e.g., the dynamic range of a VCA is typically 80 to 100 db.

electronic switch a module that routes one input to two or more alternating outputs *or* two or more inputs alternately to one output.

envelope a graph of the voltage parameters of an event over time.

envelope follower a module that derives a control voltage from the fluctuations in amplitude of (generally) an external signal.

envelope generator a module that determines timing and voltage parameters of an LF control voltage which it generates; on the ADSR the voltage parameter is variable; on the AR it is generally +10 volts.

exponential a relationship that involves a continual doubling of a given parameter.

exponential converter a submodule within a VCO, VCF, and VCA. It causes the frequency of the VCO and the VCF to change exponentially (double or halve) each time a control volt is added or subtracted. It causes the VCA to open at a rate of 10 db/volt when a control voltage is input to the exponential input.

F_c abbreviation for *cutoff frequency.*

fine tuning attenuator source of DC voltage of about ¾ volt on VCOs and 1 volt on a VCF. This voltage may be mixed into the control voltage mixers of these modules. The fine tuning attenuator is particularly helpful for precision tuning.

floating attenuator an attenuator with a separate output as well as an input.

FM *frequency modulation;* the control of the frequency of a VCO (or oscillating VCF) caused by an input control voltage. If the control voltage is LF, the change may be perceived as vibrato or a trill; if the control voltage is AF, sidebands will be generated and a modulation timbre will be perceived.

formant a harmonic or range of harmonics emphasized by a particular instrument.

frequency the number of cycles a wave completes in a given unit of time (generally one second). Now called Hertz (Hz.); 1000 Hz. = 1 KHz. (Kilohertz).

frequency modulation see **FM.**

fundamental the first harmonic of a wave; the perceived pitch of a periodic wave.

gate a timing signal of instantaneous rise time, amplitude generally of +10 volts, and some duration. Any unattenuated LF pulse wave can be a gate. A gate is generated by the keyboard each time a key is depressed and typically hard-wired to envelope generators. It signals the envelope generators to begin.

harmonics the associated higher-frequency components of a wave. They are sine waves in whole number multiples of the frequency of the fundamental. See **sine, triangle, square, pulse,** and **sawtooth.**

harmonic series a series of higher-frequency, lower-amplitude sine waves which are whole number frequency multiples of a fundamental.

Hertz synonym for *cycles per second;* abbreviated **Hz.**

Hz. see **Hertz.**

input a signal entering a module. If the signal goes into an audio input, it will probably be modified in some way; if it goes into a control input, it will probably cause some change in an audio signal entering the module. Audio inputs are diagrammed as entering a module from the left; control inputs are diagrammed as entering a module from the bottom.

inverter a signal processor that multiplies any input voltage by a constant of −1.

keyboard the most common voltage controller. Typically, if unattenuated, it will cause changes in a VCO or the VCF at a rate of 1 volt per octave. It outputs at least one control voltage and two timing signals, a gate and a trigger, every time a key is depressed.

KHz. see Kilohertz.

Kilohertz a periodic electrical signal that recurs 1000 times each second.

kluge pronounced *kl-oo-j*; verb meaning to create a certain patch where you do not appear to have the modules necessary to be able to create that patch.

lag processor a signal modifier generally used with pulse and sawtooth waves; it causes the instantaneous portions of those waves to be slowed (lagged). When an audio signal is input, it functions as a non-voltage-controlled filter whose maximum F_c is about 1600 Hz. and whose slope is −6 db/octave.

LF *low frequency* those frequencies that are too low (below 20 Hz.) to be perceived as sound. They are recognized by their effect on other frequencies or modules.

linear a description of equal increments of control voltage change that will cause equal changes in a parameter. Generally used in connection with the keyboard (which has equal voltage changes per unit—1 volt for every twelve semitones) and the linear control input to the VCA.

low frequency see **LF.**

manual control changes of voltage being made by hand rather than by electrical signal.

master the controlling oscillator of two oscillators that are synchronized.

maximum stable resonance the point just beyond which a filter will begin to oscillate.

mixer a device that adds instantaneous voltages and that outputs their instantaneous sum. E.g., if the mixer in question is the CV mixer to a VCO, the sum of the voltages in that mixer will be output as frequency.

modulated carrier the resultant signal when one audio-frequency signal has been either amplitude- or frequency-modulated by another audio-frequency signal.

modulation the control of a module by a control voltage. See **frequency modulation, amplitude modulation,** and **pulse width modulation.**

modulation index the measure of how much of one AF signal (the program) modulates the frequency of another AF signal (the carrier).

module an independent voltage-generating or sound-processing section of a synthesizer.

multiple several jacks wired in parallel which will accept one input and make that signal available at many separate outputs.

noise the sound resulting from simultaneous voltages of random amplitude. See also **white noise** and **pink noise.**

offset a DC voltage applied to a signal in order to effect a steady-state change in that signal.

one-voice synthesizer a synthesizer that has only one independent signal path.

one volt per octave the typical way that control voltage causes changes in a VCO or VCF. The linear change of one control volt causes an exponential change of one octave.

oscilloscope a device whose cathode-ray tube (like a television picture tube) can, by being connected to the output of a synthesizer, display the total waveform present at that output.

output a signal leaving a module; on block diagrams it is represented at the right of the module.

overtones wave harmonics other than the fundamental. The second harmonic is the first overtone, the third harmonic is the second overtone, etc.

panpot the attenuator that determines how much of a synthesizer output will be distributed to either side of the stereo field.

parameter a major component of a patch, whose characteristics may be varied and manually or voltage-controlled. Examples: pitch, timbre, and amplitude.

patch (1) to create a connection between an output and an input; (2) several connected outputs and inputs which create an effect, as a "patch" of a violin.

peak-to-peak (p-p) amplitude the number of volts between the maximum positive and maximum negative amplitudes of a wave. Most waves have p-p amplitudes of 10 volts, but they can be attenuated.

periodic recurring at regular intervals.

phase A wave is said to go through 360° of phase each cycle. Waves are in phase if they have their point of 0° phase at the same instant. To the extent that they do not they are out of phase. They are completely out of phase if they are 180° apart, and will cancel those components they have in common.

pink noise the sound that occurs when noise has equal enegy per unit band width.

pitch the perceived frequency of a wave in the audio spectrum.

pitch bend perhaps the single most important expressive device for live performance. The performer can manually add any increment up to ±1 volt to or from the keyboard allowing changes ranging from less than a semitone to a full octave.

portamento a signal processor that enables one voltage (as frequency) to glide to another rather than move in instantaneous discrete steps.

pre-amp a signal processor that is generally (although not always) used in conjunction with an external signal whose output is too weak and must therefore be boosted before it can be interfaced with a synthesizer.

pre-patch internally hard-wired connection which may be defeated by inserting a dummy plug (one end of a patchcord) into a jack.

process to modify in some way. E.g., a VCF is a signal processor.

program the name given to an audio-frequency signal that is either amplitude- or frequency-modulating another audio-frequency signal.

pulse wave wave with two DC voltage levels, one instantaneously high and one instantaneously low; in AF its harmonics depend upon the pulse width.

pulse width modulation changes caused by voltage or manual control in the duty cycle of a pulse wave, with corresponding changes in harmonics or, if the pulse wave is LF, control voltage parameters.

Q symbol for *resonance.*

ramp wave a negative-going sawtooth wave.

red noise the sound that occurs when noise has more energy in the lower frequencies than the upper.

release the time it takes for an envelope generator voltage to go to 0 volts from the moment the envelope generator no longer receives a gate signal.

resonance a filter characteristic in which a particular frequency or band of frequencies become more and more pronounced, causing timbral changes. At or near maximum, one frequency becomes totally pronounced and the filter oscillates, generating a sine wave at that frequency.

retriggering takes place when a new envelope is triggered prior to the completion of the release stage of the old envelope.

reverb an electro-acoustic device which creates the illusion of spaciousness.

ribbon controller a modulation mechanism that generates a voltage proportional to where one touches a long, thin ribbon.

ring modulator a module that accepts two inputs and that outputs their sum and difference frequencies; can also be used as an inverter, VCA, electronic switch, and quasi-envelope generator.

sample and hold a module that derives a control voltage by sampling instantaneous voltages of an input signal at a user-determinable rate and holding those voltages until the next sampling takes place.

sawtooth wave a wave that, if positive-going, is characterized by a gradual rise in voltage and then an instantaneous drop to the starting point; if negative-going, is characterized by an instantaneous rise and a gradual fall; has all harmonics in the harmonic series.

semitone the smallest interval in the chromatic scale. 1/12 of an octave, half a whole tone. If a VCO tracks the keyboard at 1 volt/octave, a change of a semitone means that 1/12 volt has been either added to or subtracted from the VCO's control voltage mixer.

sidebands sum and difference frequencies generated by the modulation of one AF wave by another AF wave.

signal flow the route that an audio or control signal takes in a patch.

sine wave the basic wave in nature and acoustics; characterized by a gradual rise and fall of alternating current; has no harmonics, since all harmonics are themselves sine waves.

slave the controlled VCO of two VCOs in the synch mode.

slope the degree of steepness by which a filter attenuates harmonics. The slope of a typical low-pass VCF is −24 db/octave. When a lag processor acts as a filter its slope is −6 db/octave.

square wave a pulse wave whose duty cycle is ½; has only odd harmonics.

s-trigger the signal that "shorts" a +10 volt gate to ground; used with Moog envelope generators.

subtractive synthesis synonym for *filtering,* the process by which harmonics are attenuated or, using a resonance circuit, accentuated.

sustain the user-determined variable voltage level of the ADSR.

synch generally, the process whereby the timing of one event is determined by the timing of another event. When oscillators are "in synch" the waveform of the "slave" resets at a time determined by the frequency of the "master" waveform.

synthesizer wave generators, modifiers, and controllers which are able to create a multiplicity of sounds. A modern synthesizer has at least a VCO, VCF, VCA, and envelope generator.

timbre the unique characteristic sound of a particular wave, determined by its harmonic content. E.g., although the frequency and amplitude of a sine and sawtooth wave may be the same, their timbre is quite different.

timing signal the general name given to voltages that signal modules (typically envelope generators) to start or stop.

tone color see **timbre.**

transient generator see **envelope generator.**

transpose switch switch that adds or subtracts up to 2 volts to the keyboard, resulting in raising or lowering of a VCO's frequency, or VCF F_C, by two octaves.

tremolo a rapid periodic shift in amplitude of an audio wave.

triangle wave wave characterized by a linear voltage rise and fall of alternating current; has only odd harmonics, whose amplitudes decrease exponentially.

trigger a voltage of instantaneous rise time to an amplitude of some specific voltage. Typically used to begin the attack stage of the ADSR. Generated by the keyboard each time a key is depressed, a trigger is generally hard-wired to the ADSR.

trill a rapid, periodic, instantaneous shift in frequency of either a semitone or a whole tone of an audio wave.

unattenuated generally refers to an input that has no attenuator associated with it.

VCA—*voltage-controlled amplifier* a module that allows the instantaneous sum of the voltages in its CV mixer to determine the instantaneous output amplitude of the audio voltage in its audio mixer.

VCF—*voltage-controlled filter* a module that allows the user to determine harmonic attenuation of an input wave. A typical low-pass VCF has a slope of −24 db/octave, tracks the keyboard at 1 volt/octave, has audio and control inputs and a resonance circuit.

VCO—*voltage-controlled oscillator* a module that generates different periodic waveshapes at user-adjustable frequencies which may be changed by voltage control.

vibrato a low-amplitude rapid periodic smooth shift in the frequency of an audio wave, generally caused by an LF sine wave as a control voltage. Adds considerable expressivity in musical passages.

voltage control the most important concept in understanding audio synthesis. The ability to change a parameter output by inputting different control voltages to a module.

voltage-controlled amplifier see **VCA.**

voltage-controlled filter see **VCF.**

voltage-controlled oscillator see **VCO.**

voltage controller a source of control voltage. E.g., the keyboard or an envelope generator, although it can be any source of voltage.

white noise the sound that occurs when the probability of any frequency occurring is the same as that for any other frequency. In practical terms it sounds like all frequencies occurring simultaneously.

Discography

Like the Bibliography, this Discography is meant to be neither scholarly nor exhaustive. More complete discographies may be found in 7, 8, and 17 of the Bibliography. *Contemporary Keyboard* (3) regularly has an extensive record review column of interest to synthesists. Aeon Import Records, Inc., 604 Princeton Street, Fort Collins, CO 80525, has an extensive mail order service featuring electronic and "new" music. In addition, the New Music Distribution Service, 500 Broadway, New York, NY 10012 will send you their extensive catalogue of New Music, electronic and otherwise, for $1.00. With over 1000 *annotated* listings it's well worth it.

Finally, I.E.M.A. and SYNEX (discussed in the first portion of the Bibliography) regularly review new works that are both commercially available and, more important, available directly from the composer via cassette tape. This "underground" synthesis network is very fertile territory for new artists and their music. The address of the composer is given so that if a review intrigues you, you can get her or his tape for yourself. Some of these "home-brew" tapes eventually are released as commercial records. Take heart.

Because there are so many discographies available, I will list only a few examples in different categories of electronic music.

1. Reference

   A. *Electronic Music from the Outside In,* Barton and Priscilla McLean, Folkways FPX 6050

   B. *The Nonesuch Guide to Electronic Music,* Beaver & Krause, Nonesuch HC-73018

C. *Electronic Pioneers,* various pioneers, CRI SD 356 (170 West 74th Street, New York, NY 10023)

D. *Electronic Music,* various "classical" composers, Columbia-Princeton Electronic Music Center, Finnadar QD 9010

2. Western classical music

   A. Faithful to original

      1. Walter Carlos, *Switched on Bach,* Columbia MS-7194
      2. Mychael Danna, *A Synthesized Interpretation of Classic Pieces,* available from *Polyphony* magazine
      3. Patrick Gleeson, *Beyond the Sun,* Mercury SR 180000 (this is Holst's *Planets*)
      4. Brad Slocum, *Sonic Synsations,* B. A. Slocum Records, 810 Ladis Court, Sunnyvale, CA 94086 (realized entirely on an ARP 2600)

   B. Interpretive

      1. Kraft & Alexander, *Tchaikovsky's 1812 Overture and Nutcracker Suite,* London SPC 21168
      2. Tomita, *Holst's Planets,* RCA ARL-1 1919 (compare with Gleeson's above)

3. New wave

   A. Blondie, *Parallel Lines,* Chrysalis 1192

   B. Gary Numan, *Pleasure Principal,* Atco SD 38120

   C. Talking Heads, *Fear of Music,* Sire 6076

4. Computer music

   A. Jon Appleton, *Music for Synclavier and other Digital Systems,* Folkways FTS 33445

   B. *First Philadelphia Computer Music Festival,* available from *Creative Computing,* P.O. Box 389-M, Morristown, NJ 07960

   C. *New Directions in Music,* various composers, Tulsa Studios, 6134 East 13th, Tulsa, OK 74112

   D. *Unplayed by Human Hands,* Prentiss Knowlton, programmer, Computer Humanities CR 9115, 2310 El Moreno St., La Crescenta, CA 91214

5. Funk

   A. George Duke, *The Aura Will Prevail,* MPS/BASF MC 25613

   B. Mandre, *Two,* Motown M7-900R1

   C. Prince, *Dirty Mind,* WB BSK 3Y78 (caution: this is an x-rated album)

   D. Stevie Wonder, *Songs in the Key of Life,* Tamla T13-340C2

6. Jazz
   A. Chick Corea, *The Mad Hatter,* Polydor PD-1-6130
   B. Barry Phillips, *Three Day Moon,* ECM 1123, *Mountainscape,* ECM 1076
   C. Terje Rypdal, *Descendre,* ECM 1-1144
   D. John Surman, *Upon Reflection,* ECM-1-1148
   E. Weather Report, *Heavy Weather,* Columbia 34418
7. Pop
   A. Bee Gees, *Spirits Having Flown,* RSO ES-1-3041 (especially the Oberheim eight-voice in unison mode on "Tragedy")
   B. Manfred Mann's Earth Band, *The Roaring Silence,* Bronze BS 2965
   C. Alan Parsons, *Eve,* Arista AL 9504
   D. Toto, *Toto,* Columbia JC 35317
   E. Gino Vanelli, *The Gist of the Gemini,* A & M SP-4596
   F. Gary Wright, *The Right Place,* Warner Brothers BSK-3511
8. Symphonic pop
   A. John Anderson, *Olias of Sun Hillow,* Atlantic SD 18180
   B. Emerson, Lake & Palmer, *Brain Salad Surgery,* Manticore MC 66669
   C. Jan Hammer, *The First Seven Days,* Nemperor NE 432
   D. Patrick Moraz, *Patrick Moraz,* Charisma CA-1-2201
   E. Mythos, *Quasar,* Sky 046
   F. Vangelis, *China,* Polydor PD-1-6199
   G. Rick Wakeman, *The Six Wives of Henry VIII,* A & M SP-4361
9. Synthi-Pop
   A. M. Factor, *Official Secrets Act,* Sire SRK 6099
   B. The Flying Lizards, *The Flying Lizards,* Virgin VA 13137
   C. The Human League, *Traveloge,* Virgin VI 2160
   D. New Musik, *Sanctuary,* Epic NFE 37314
   E. Orchestral Manuevers, *Organization,* Didsong Ltd., DID 6, ℅ Virgin Records
   F. Ultravox, *Vienna,* Chrysalis CHR 1296
10. Meditative
    A. Brian Eno, *Music for Airports,* PVC 7908
    B. Brian Eno and Laarji, *Days of Radiance,* E.g. 203
    C. Steve Hillage, *Rainbow Dome,* Virgin VR-1

D. Iasos, *Inter-Dimensional Music,* Unity Records UR-700, Box 12, Dept. K, Corte Madera, CA 94925

E. Keith Leimer, *Closed System Potentials,* Palace of Lights, 10226 65th Avenue S., Seattle, WA 98178

11. Robot music

A. Devo, *Are We Not Men?* Warner Brothers BSK 3239

B. Kraftwevk, *The Man-Machine,* Capitol SW-11728

12. The "German School"

A. Peter Baumann, *Romance '76,* Virgin P2 34897

B. Edgar Froese, *Ypsilon in Malaysian Pale,* Brain 0074

C. Michael Hoenig, *Departure from the Northern Wasteland,* Warner Brothers BSK 3152

D. Klaus Shultze, *Body Love,* Island ILPS 9510

E. Tangerine Dream, *Sorcerer,* MCA-2277; *Rubicon,* Virgin 2025; *Ricochet,* Virgin 89679 XOT

13. Strictly synthesizers

A. Cluster, *Sowiesos,* SKY 005 Sky Records, Nordhauser Weg 16, D-2000, Hamburg 61, W. Germany

B. Cluster and Eno, *Cluster and Eno,* SKY 010

C. John Duesenberry, *Four Movements for Tape,* Opus 1 Records, Box 604, Greenville, ME 04441

D. Brian Eno, *After the Heart,* SKY 021; *Fourth World Possible Musics, vol. 1,* E.G.S. 107, ℅ Jem Records, Inc., S. Plainfield, NJ 07080; *Music for Films,* EGS 105 with Harold Budd; *Plateaux of Mirrors,* EGS 202; with David Byrne, *My Life in the Bush of Ghosts,* Sire SRK 6093

E. Michael Garrison, *Prisms,* Windspell 11281, P.O. Box 5382 Bend, Oregon 97708

F. Harold Groskopf, *Synthesist,* SKY 043

G. Heldon, *Stand Back,* Egg BA 215

H. Michael Hoenig, *Departure from the Northern Wasteland,* Warner Brothers BSK 3151

I. Jean-Michel Jarre, *Oxygene,* Polydor PD-1-6112

J. Moebius, *Moebius,* Moonwind MW 33801 (433 North Justin Avenue, Orange, CA 92667)

K. Moebius and Plank, *Rasta Kraut Pasta,* SKY 039

L. Roger Powell, *Air Pocket,* Bearsville BRK 6994

M. H.J. Roedelius, *Lust Wandel,* SKY 058 and *Jardin au Fou,* Barclay 90291. Barclay Records, ℅ Egg Records, 143 Avenue Charles de Gaulle, 758.1277, Paris, France

N. Klaus Schultze, *Cyborg,* Brain 2/1078, ℅ Metronome Music GMBH, Ubersearing 21, 2000 Hamburg 60. W. Germany; *Dune,* Brain 0080.225

O. Synergy, *Games,* Passport PB6003 (read about how it was made in *Polyphony,* vol. 5, no. 5 (Jan./Feb. 1980)

P. Toy Planet, *Toy Planet,* Spoon 011LC7395, Spoon Records, ℅ Hildegard Schmidt, PO Box 350209, D-5000 Koln 30, W. Germany.

14. Some albums and cassettes from the underground network

A. Devarahi, "Suchness," 1643 Riverview St., Eugene, OR 97403

B. Michael Gilbert, "The Call," 104 Riverglade, Amherst, MA 01002.

C. Leon Lowman, "Syntheseas," Box 262, Narragansett, RI 02882.

D. Michael C. McDonald, "Future Dance" b/w "Runner" (45 r.p.m. record), P.O. Box 70103, Eugene, OR 97401-0106

E. Mark Petersen, "Geodesium," P.O. Box 3023, Boulder, CO 80307.

F. Keith Slane, "Star Captain," Route 1, Box 189-D, Pryor, OK 74361.

15. "New" music—not necessarily all synthesis

A. Philip Glass, *North Star,* Virgin P234669

B. Steve Reich, *Music for 18 Musicians,* ECM 1129

C. Terry Riley, *A Rainbow in Curved Air,* Columbia M57315; *Shri Camel,* Columbia M-35164

D. Rosenboom & Buchla, *Collaboration in Performance,* 1750 Arch Records S-1774

E. Morton Subotnik, *Silver Apples of the Moon,* Nonesuch H-71174; *Touch,* Columbia MS-7316

Annotated Bibliography: Books and Periodicals

Instead of listing these books and periodicals alphabetically by author or title, I am listing them by what I consider to be their relative cost-effectiveness to the new synthesist. I am listing only those titles that I feel would definitely be worth being familiar with. If a book or periodical is not on this list, that does not mean I don't find it valuable; I may never have heard of it. However, if it is on this list I have personally found it worthwhile. Extensive and scholarly bibliographies may be found in *Electronic Music* (12) and *The Development and Practice of Electronic Music* (17).

There are at least two "underground" synthesis networks in the United States: I.E.M.A. (International Electronic Music Association), P.O. Box 456, Salamanca, NY 14779; and SYNEX (The Synthesizer Exchange Newsletter), P.O. Box 294, Corte Madera, CA 94925. Both of these home-based newsletters create a forum where synthesists can easily meet one another, exchange information, and keep abreast of current events. Of particular interest are the reviews in both publications of homemade tapes: synthesists who don't necessarily have access to very expensive equipment but who have done their best and are putting it out for their peers to hear. These are just about the only sources for learning of these tapes. Cost is moderate (SYNEX is free if you correspond with the publisher). Highly recommended.

1. *Polyphony* magazine, P.O. Box 20305, Oklahoma City, OK 73156. Formerly concerned only with PAIA equipment, for the past three years this magazine has broadened its sphere to the point where it is now an indispensable aid to all synthesists. More technically oriented than *Synapse,* it has many construction and other how-to articles, sophisticated modifications to presently existing equipment, discussions of equipment, record reviews, and frequent computer/synthesis interface information. Judging

from correspondence, the readers are still largely PAIA owners. Issues from 1978 on recommended.

2. *Synapse* magazine, 1052 West 6th Street, #424, Los Angeles, CA 90017. This magazine was concerned only with synthesizers and electronic music; almost every article was appropriate for the beginning synthesist. Record and equipment reviews, interviews with prominent synthesists and technical innovators, miscellaneous columns, even the advertisements made this a wealth of information. Has not been published since 1979. Find as many back issues as you can.

3. *Contemporary Keyboard* magazine, P.O. Box 28836, San Diego, CA 92127. This "biggest" magazine covers all keyboards, not just synthesizers, but there is always something of value to synthesists. Interviews with Jan Hammer, Wendy Carlos, and Larry Fast are just as frequent as with Keith Jarrett, Elton John, or Oscar Peterson. Particularly recommended are the columns by Bob Moog, Pat Gleeson, Roger Powell, Tom Coster, and the CK staff.

4. Herbert A. Deutsch, *Synthesis: An Introduction to the History, Theory and Practice of Electronic Music,* Alfred Publ. Co., Inc., Port Washington, NY, 1976. Although the synthesizers discussed are generally outdated, the basic discussion of synthesis theory and recording technique will be extremely helpful to the beginner. The presentation is cursory but effective. A record that is included with the book demonstrates synthesis and basic recording techniques.

5. *Home Recording for Musicians,* Craig Anderton, Guitar Player Books, Music Sales Corp., 33 West 60 Street, New York, NY 10023. There's more to taping your synthesizer than just plugging it into the tape deck (although it *is* a lot easier than recording acoustic instruments). Craig's columns appear everywhere, and they are always literate and considerate of the reader. His book explains clearly and comprehensively everything you will need to know to get started in home recording. Also recommended: *Electronic Projects for Musicians,* same author.

6. *Electronotes* periodical, 1 Pheasant Lane, Ithaca, NY 14850. Specifically for music engineers rather than synthesists, this is the best technical periodical of electronic music. The editor, Bernie Hutchins, cares so much about his readers and his work that he publishes his phone number in each issue and invites readers to call and ask about anything relevant to electronic music. The primary emphasis is sophisticated state-of-the-art circuit design for useful electronic music modules, but there are also occasional record reviews (by the ubiquitous Craig Anderton) and field reports on electronic music happenings. *Electronotes* will generally be difficult for anyone who has needed THE COMPLETE GUIDE . . . to explain voltage-control, but if a person's reach should exceed her grasp, this is the periodical to reach for. Highly recommended.

7. Thomas Wells, *The Technique of Electronic Music,* Macmillan, Inc., 866 Third Avenue, New York, N.Y. 10022 (1981). The best book on *theory* of voltage-controlled synthesis. No experiments, lots of math, more a college text than a "how-to" book. If you understand THE COMPLETE GUIDE . . . you're ready for Wells. Advanced block diagrams, questions and answers, much technical knowledge. However, despite the 1981 copyright, the extensive bibliographies after each chapter, as well as the synthesizers referred to, are generally outdated, being no later than 1976.

8. *Electronic Music: Resources for Performance Groups and General Music Classes,* Cross Creek Press, P.O. Box 12533, St. Louis, MO 63141 (1979). A plethora of information, in the form of a computer printout, of interest to the synthesist: extensive bibliography and discography, electronic music scores, and list of periodicals likely to have articles of interest to the synthesist. This is an invaluable resource.

9. *Modern Recording's Buyers Guide,* 14 Vanderventer Avenue, Port Washington, NY 11050. This annual publication is the first place to shop for all those outboard goodies you're going to want sooner or later. It lists and compares thousands of products without recommending any of them. You'll want to know about all the "black boxes" available before you invest in one. You should also be aware of *Modern Recording* magazine, created for the semiprofessional recordist.

10. *How to Make Electronic Music,* Drake, Herder, and Modugno, Educational Audio Visual, Pleasantville, NY 10570 (1975). Deals very basically with synthesis, more deeply with introductory recording techniques. A good primer on recording electronic music.

11. *The Source Book of Patching and Programming from Polyphony* (a compilation of the best from the readers of *Polyphony* magazine), Polyphony Publishing Co., Oklahoma City, OK (1978; no editor or author listed). Although this patch book is strongly oriented toward PAIA modules, a VCO is a VCO is a VCO, so you can use these patches as starting points on any synthesizer. Some computer programs for interfacing synthesizers and computers are found at the back of the book.

12. *Electronic Music,* Allen Strange, Wm. C. Brown (1972). Comprehensive introduction to electronic music assumes some minimal knowledge (which you have more than, having finished THE COMPLETE GUIDE . . .). Discusses many approaches to synthesis not discussed in this book, including location modulation, quad recording, live "classical" performance. Should be viewed as complementary to THE COMPLETE GUIDE.

13. *The ARP 2600 Patch Book,* from ARP Instruments, 45 Hartwell Avenue, Lexington, MA 02173. One hundred patches from ARP which should be viewed as starting points. Useful if you own a 2600.

14. *The Waveform Music Book,* Mary Snow, Lariken Press, 3110 26th Street, Lubbock, TX 79410 (1977). Privately published instruction book and patches for the 2600. Block diagrams are difficult to understand but recommended for ingenuity of thought and as starting points for further exploration on the 2600.

15. *The Recording Studio Handbook,* John M. Woram, Sagamore Publishing Company, Inc., Plainview, NY 11803 (1976). This book deals in great detail with recording techniques of all types. More than you'll ever need to know about making clean tapes unless you go professional.

16. *The Acoustical Foundations of Music,* John Backus, W. W. Norton & Co., Inc., New York (1969). To be a fine synthesist you need to understand more than voltage control; you need to understand what music is, what physical properties musical instruments have in common and how they differ, and acoustics. This standard textbook explains all these and more: environmental acoustics, differently tuned scales, etc.

17. *The Development and Practice of Electronic Music,* Appleton and Perera, editors, Prentice-Hall, Inc., Englewood Cliffs, NJ 07632. A standard among serious electronic music students, this challenging and academically oriented series of essays on various aspects of electronic music is essentially a "classical" approach. In addition to an extensive (pre-1975) bibliography, it has six essays entitled "Origins," "Sound, Electronics and Hearing," "The Tape Studio," "The Voltage-Controlled Synthesizer," "The Uses of Digital Computers" (100 pages on this alone), and "Live Electronic Music."

18. *Electronmusic,* Robert Devoe, EML Laboratories, Inc., Cernon, CN (1977). If THE COMPLETE GUIDE . . . is the linear and Western approach to synthesis, this is an attempt at the intuitive and Eastern approach (complete with dedication to Shiva, the

Hindu god of destruction). Interesting ideas if you know about synthesis already, particularly his ideas about electronic music notation (the author believes notation should be art). Half the book concerns taping. His "ultimate studio" is the stuff of which daydreams are made. The book is biased toward EML equipment (note the publisher).

19. *Learning Music with Synthesizers,* Friend, Pearlman, and Piggott; and *Lessons in Electronic Music,* Friend, Pearlman, and Maltzman, Hal Leonard Publishing Corp. (1974). Co-author Pearlman is Alan R. Pearlman (ARP). The first book is oriented toward the ARP Odyssey, the second toward the ARP AXXE. Recommended if you have those instruments.

Books worthy of special notice:

These three books do not properly belong in this bibliography, since they are not about synthesizers. However, they are about subjects that might be of interest to synthesists and are of such unusually high caliber, in both content and readability for the non-expert, that I recommend them to you without reservation.

The first of these books is called *The Secret Guide to Computers.* This book comes in eight wonderfully readable volumes that cost a *total* of $29.60. The author, Russ Walter, publishes his own books from 92 Saint Botolph Street, Boston, MA 02116. You may buy only volume 1 for $3.70 (Russ pays postage) and get a feel for how successful the communication is. Like Bernie Hutchins of *Electronotes,* Russ publishes his phone number in his book and invites people to call at any time with any questions for all the free (and extremely competent) advice they want. Remember: you may one day want the incredible flexibility that a microprocessor can give you in controlling a synthesizer. Here is the place to learn what you need.

The second of these books is *The Dancing Wu-Li Masters,* by Gary Zukav. New York, NY: Wm. Morrow, 1979. Synthesis is a limited zone of the study of waves, otherwise known as energy. This book expands one's perceptions of reality by exploring, in language any interested person can understand and be blown away by, the psychedelic reality of the subatomic world (quantum physics, but don't let that scare you), the "warped" reality of the macroscopic universe (relativity theory), and the limited reality, the illusion, of our "zone of middle dimensions." To understand the implications of this book is to have a powerful humility.

Recommended as similar (but with more emphasis on comparisons to Eastern mysticism), is *The Tao of Physics,* by Fritjov Capra. Berkeley, CA: Shambola Press, 1975.

The third book is *Gödel, Escher, Bach: An Eternal Golden Braid,* by Douglas Hofstadter. New York, NY: Vintage Books, 1979.

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As founder and present director of Synthesis Seminars of America, Devarahi teaches and gives demonstrations of synthesizers at seminars throughout the United States and Europe. When not "on the road," he teaches electronic music at Lane College in Eugene, Oregon.

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