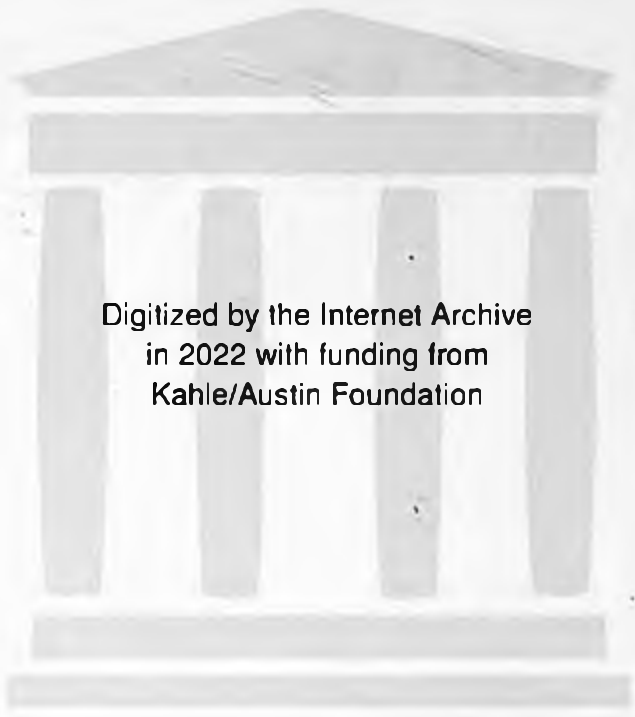


BUILD A BETTER
MUSIC
SYNTHESIZER



THOMAS HENRY



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THOMAS HENRY



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Blue Ridge Summit, PA 17214

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FIRST EDITION

FIRST PRINTING

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Library of Congress Cataloging in Publication Data

Henry, Thomas.

Build a better music synthesizer.

Includes index.

1. Synthesizer (Musical Instrument)—Construction.

I. Title.

ML1092.H25 1986 789.9'9 86-5942

ISBN 0-8306-0255-0

ISBN 0-8306-2755-3 (pbk.)

189.77
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Contents

Acknowledgments	v
Introduction	vi
Part I: Understanding Synthesizers	1
1 The Three Components of Sound	3
Accuracy and Precision—Decibels, Volume and Dynamic Range—Summary	
2 Synthesizer Modules	11
The Voltage-Controlled Oscillator—The Voltage-Controlled Amplifier—The Voltage-Controlled Filter—Envelope Generators—Low-Frequency Oscillators—The Noise Source—The Balanced Modulator—The Keyboard—Exotic Controllers—The Power Supply—Summary	
3 Electrical Standards	25
Supply Voltages—Signal Levels—Control Voltages—Output Structures—Input Structures—Trigger Signals—Selecting Components—Obtaining Parts—Summary	
4 Planning the Synthesizer	36
Rack-Mount Enclosures—Making Rack Panels—Circuit Construction Methods—Attaching Circuit Boards to the Panel—Patch-Over Hardware—Summary	
Part II: Construction Projects	55
5 Deluxe VCO	57
How the Circuit Works—Construction—Calibration—Using the VCO	

6	A Four-Pole Lowpass VCF	65
	How the Circuit Works—Construction—Calibration— Troubleshooting—Using the VCF	
7	A Dual VCA	73
	How the Circuit Works—Construction—Calibration—Using the VCA	
8	A One-Chip ADSR	80
	How the Circuit Works—Construction—Using the ADSR	
9	The Supercontroller	85
	Introducing the SN76477 — How the Circuit Works — Construction—Calibration—Using the Supercontroller	
10	A Digital Keyboard	97
	How the Circuit Works—Construction—Calibration—Using the Dig- ital Keyboard	
11	The Potpourri Module	111
	How the Circuit Works—Calibration—Using the Potpourri Module	
12	A Quadrature Function Generator	116
	How the Circuit Works—Construction—Calibration—Using the Quadrature Function Generator	
13	A Balanced Modulator	124
	How the Circuit Works—Construction—Calibration—Using the Balanced Modulator	
14	The Power Supply	131
	How the Circuit Works—Construction—Using the Power Supply	
	Part III: Putting It All Together	137
15	Assembly	139
	Power Supply Connections—Completing the Synthesizer—Testing and Troubleshooting—Using Your Synthesizer—Summary	
16	Inventing New Modules	148
	Suggestions for More Analog Circuits—Moving Towards Digital Synthesis—Computers for Electronic Music—On to the Future!	
	Appendices	155
A	Suppliers	157
	Books and Manuals—Periodicals	
B	Sources	159
	Books and Manuals—Periodicals	
	Index	163

Acknowledgments

No book is a product of just one person's mind, and this one is certainly an example of that premise. Many people helped in many different ways to make this book a reality. I am grateful for their support and help throughout the years.

I wish to thank Craig Anderton for his suggestions both in the area of electronic design and writing. His many articles and books have set an example of clarity which more authors should try to emulate. He has helped shape my career as an author more than any other person.

Bernie Hutchins, publisher of *Electronotes*, first instilled in me the notion that you can always build a better synthesizer than you can buy. He has also been most kind in allowing me to use portions of some of my articles from *Electronotes* in the preparation of this book.

Thanks also go to Greg Schneck, publisher of *Common Mode Newsletter*, and John S. Simonton Jr., publisher of *Polyphony*, for allowing me to reprint portions of my articles appearing in those publications.

Dr. Kenneth Good, Mankato State University, Mankato, Minnesota, was most kind in reading portions of the manuscript and making a number of suggestions. I am grateful for his patience in this, and the many discussions which we had have shaped the organization of this book considerably. Of course, any errors remaining are my own.

I also want to thank James P. Lynch for taking the fine photographs that grace this book. His care and attention to detail is most appreciated.

Introduction

"You can always build a better synthesizer than you can buy" is the credo of the book you are now reading. Many home experimenters, basement musicians, and electronics hobbyists have proven this over and over again. Though you may find this statement hard to believe right now, by the time you are done reading this book you will come to accept it.

Now no one says it will be easy. To build a good synthesizer requires devotion, attention to detail and a certain amount of sacrifice. These are concepts which each reader will have to learn for himself and in his own way. But to get you started, many tips and hints are scattered throughout the book. These tips and hints are based upon actual work done in the area and not a one of them was included without first being thoroughly tested.

The organization of this book is quite straightforward. In Chapters 1 through 4, the theoretical background needed to build a synthesizer is treated in some detail. However, note that these chapters don't pretend to exhaust the subject of the theory of synthesizers, but instead present only what is actually needed to start building. If you desire additional theoretical information, then you should refer to some of the books listed in Appendix II.

Chapters 5 through 14 detail 10 complete construction projects. Although a thorough treatment of the circuit action is given for each of these projects, the emphasis is on how to actually build and use them. These 10 projects, when built, make up a complete studio-quality synthesizer which will rival any commercial unit.

Chapters 15 and 16 show you how to tie the whole system together into a usable musical instrument. While the previous chapters have been "microcosmic" in nature, these last two chapters will help you to see the

total system organization.

Appendix A gives a list of suppliers of electronic parts suitable for use in a synthesizer and Appendix B shows where you can learn more about the subject.

Even though the structure of this book is quite simple, you are urged to read it several times through. Portions of the first four chapters will make more sense after you have read about the construction projects, and likewise the construction projects will make more sense after reading the last two chapters on system configuration. Building synthesizers is clearly a circular affair. You must jump into the circle somewhere and keep spinning around and around until everything falls into place.

If you don't believe it now, by the time you're done with this book you will know it: you can always build a better synthesizer than you can buy. Good luck in your adventure!

Part 1

Understanding Synthesizers

Chapter 1

The Three Components of Sound

All sound can be broken down into three major components: frequency, amplitude and harmonic content. For example, a flute has a fairly high frequency range, low amplitude and very low harmonic content and it is the combination of these parameters which makes it sound very different from a trumpet. The trumpet, on the other hand, creates sounds that are quite different from a snare drum, and once again the explanation can be found by noting the differences in frequency, amplitude and harmonic content of the two instruments.

This, then, is the fundamental concept of all sound synthesis. If you can figure out ways to control these three parameters, then in theory, you should be able to synthesize any sound at all. Note that this is a two-way street. First, you must analyze the sound (break it down into its three components) to determine what the three parameters are up to. Next you synthesize the sound, using electronic circuits. *Synthesize*, of course, means to build up using simpler components.

Though this fundamental concept is quite simple, it should be noted that in universities and research institutes the theory of sound is studied in greater detail. Using Fourier analysis and other advanced mathematical methods, researchers have discovered countless facts about sound production. *Fourier analysis* is the study of how sine and cosine waveforms can be added together in various proportions and quantities to generate a complex waveform. Besides Fourier analysis, a scientific understanding of how sounds are made is based on a knowledge of acoustics, perception and the physiology of hearing. Fortunately, to build, understand, and use a synthesizer you won't need to get into this at all! A working knowledge of the three parameters of sound is sufficient to allow you to build and use

a practical electronic music synthesizer.

A fundamental principle is that all sound can be explained in terms of these three major components. (Strictly speaking, only frequency and amplitude are actually needed, since harmonic content can be explained in terms of these two parameters. However, the computations and mathematics get exceedingly complex, so let's stick to a three-parameter approach.) Now scientists and musicians tend to speak different languages at times, so let's look at how a musician might explain sound.

When a musician hears a flute playing, he probably won't say a thing about frequency, amplitude or harmonic content. Instead, he will explain the sound in terms of pitch, volume and timbre. He might say that a flute has a high pitch, low volume and a simple timbre, just as he might classify the trumpet as being medium pitch, loud and having a complex timbre. We all have an instinctive feel for the words "pitch" and "volume," but what about "timbre?" *Timbre* (pronounced "tam-ber") is a term which refers to the "richness" of a tone. To get a grip on this fairly elusive concept try the following experiment.

Sing the following vowels, holding on to the note for 15 seconds or so. (You don't have to be a Caruso to perform this experiment!) First sing "AH" (as in "father"), then "EY" as in "fame," then "EE" (as in "niece") and finally "OH" (as in "float"). Even if you sing these vowels at the same pitch and volume, there is still something which makes them different from one another. This is timbre.

Here's a more ambitious experiment that will really drive the point home. This is intended to be a thinking experiment, but if you have the equipment available, you might want to actually give it a try. Assemble a group of musicians around a tape recorder. Now tell a flute player to sound a note, and hold it for fifteen seconds. After the note has started, turn on the tape unit and record it. Now record a trumpet playing the same note as the flute, once again turning on the recorder only after the note has started. By waiting a few seconds before turning on the recorder you will be deleting the "attack" characteristic (to be discussed in Chapter 2). Now repeat the experiment, but this time use a guitar, then a violin and any other instruments you might have handy.

When you play the recording back, you will hear several tones; each one has the same pitch, and if the volume control on the recorder was carefully monitored, each one should also have the same volume. And since the attack transient on each note was truncated by turning the recorder on only after the note had started, each tone should start and stop in roughly the same way. Listen to the tones carefully; do you hear the difference? You should. It's hard to classify the tones, but subjectively the flute should sound pure, the trumpet rather harsh, the guitar smooth, a saxophone rather raspy and so on. You've just heard timbre! You might say, then, that timbre is the tone quality or richness of the sound.

A scientist would say that all sound may be broken down into frequency, amplitude and harmonic content. The musician would use the terms pitch,

volume and timbre. Are the scientist and the musician really saying the same thing with two different sets of words? Is pitch the same as frequency, volume the same as amplitude and timbre the same as harmonic content? Perhaps in everyday life we may use the terms interchangeably, but actually there are differences.

Consider pitch and frequency. You turn on a function generator and create a sinewave with frequency of 10 Hz. Does this tone have frequency? Of course, the frequency is 10 Hz, as mentioned. But does it have pitch? If you hook up your function generator to a speaker you can quickly determine that the answer is no. The tone is too low in frequency to have pitch. Instead of a continuous tone, you will hear a series of discrete clicks.

Now crank out a sine wave at 500 kHz (500,000 Hz). Clearly, by definition, the frequency is 500 kHz, but is there pitch? Once again, the answer is no, since this frequency is way beyond what the human ear can perceive.

It's clear that the human ear can only hear a small proportion of all possible frequencies. In general, tones below 30 Hz are subsonic and tones above 20 kHz are supersonic, meaning that for average ears, the range of audibility is about 30 Hz to 20 kHz. Since pitch only applies to tones that can be heard, it is clear that pitch and frequency are not the same thing at all.

And there's another factor to consider as well. Research in the area of psycho-acoustics has shown that a given tone with a fixed frequency may actually have two different pitches depending on the amplitude of the tone. For example, an 8 kHz tone played quietly may sound different than the same tone played very loudly. Compounding the trouble is the fact that the listening environment may also affect pitch perception somewhat.

A major conclusion may be drawn now: frequency is an "absolute" term and can be used to describe the periodicity of any waveform. But pitch is a "relative" term and is used to describe the ear's (and brain's) perception of frequency. Or to put it another way, pitch is the psychological perception of frequency.

Similar differences exist for amplitude and volume. Amplitude is a scientific term and is generally used to describe the peak voltage level of a waveform. As such, amplitude can be measured quite precisely with instruments. On the other hand, volume is a psychological term; it describes the perception of amplitude.

It should come as no great surprise then, that timbre is the psychological perception of harmonic content. Harmonic content can be measured in the laboratory with oscilloscopes, distortion analyzers, and so forth over a broad range of frequencies, but timbre is how we perceive these harmonics with our ears and brains.

Figure 1-1 shows the six terms thus far described. An inspection of this figure shows that frequency, amplitude and harmonic content are indeed related to pitch, volume and timbre, but it must always be remembered that the latter three terms are used to describe the psychological

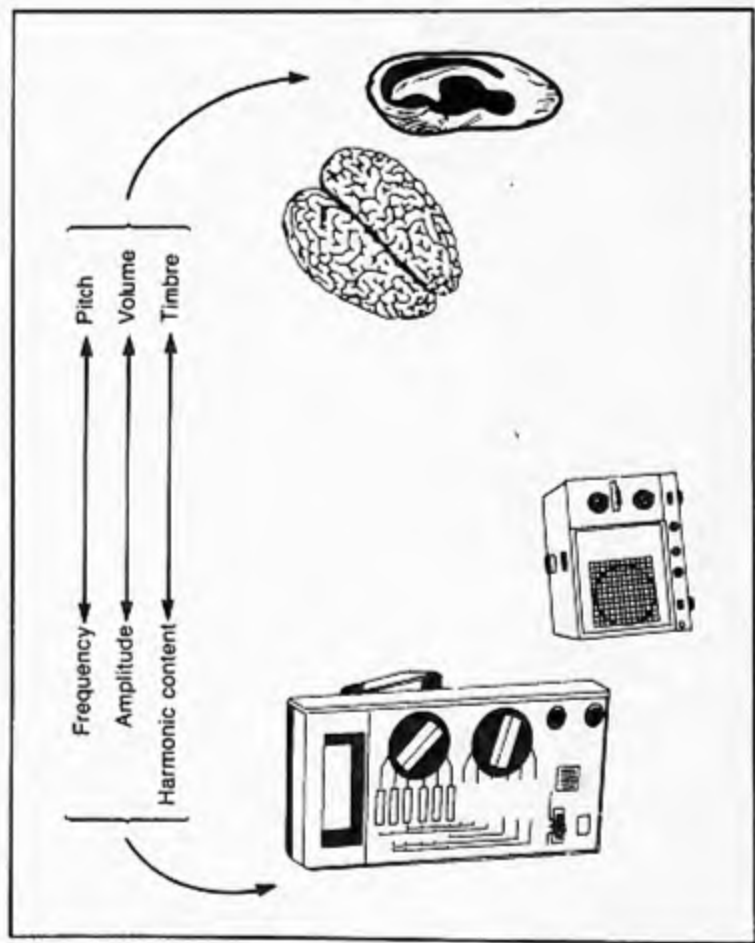


Fig. 1-1. The three parameters of sound.

perception of the former three.

As a builder of electronic music synthesizers, you are in a special position. From the electronic side of things, you must have a feel for frequency, amplitude and harmonic content, since the electronic components used to construct a synthesizer will directly determine the effects of these three parameters. But as a musician playing your completed instrument, you will want to be familiar with pitch, volume and timbre. Being able to see the electronic as well as the musical side of things will enable you to design, build and play electronic music synthesizers with greater versatility.

Too often in the past, an engineer designed a synthesizer but had no idea how the instrument was to be used. The engineer simply created the instrument based upon his ideas of what he thought was important. And since he didn't know any music terminology, it was difficult for him to understand just what it was that the performer wanted in an instrument. The musician, on the other hand, understood in musical terms what sorts of sounds he wanted, but knowing no electronics, was hard-pressed to convey these ideas to the engineer.

Fortunately, in the last several years a new sort of engineer has stepped forward, the musical engineer. The musical engineer more often than not plays some musical instruments, knows how to read music and also listens to new music. And besides being able to work with music, he is skilled in electronics, knows a little about human hearing and has studied acoustics as well. Such an engineer is able to talk with musicians and other engineers with equal ease, and thus is able to design and build musical instruments undreamed of just ten years ago.

Throughout this book we will dart back and forth from the musical side of things to the electronic. Knowing about the three parameters of sound and how the musician and scientist view them is just the start. As we go on, we will be looking at other ideas from both scientific and musical perspectives.

ACCURACY AND PRECISION

One final concept which must be examined before leaving this chapter is precision. We've all heard "sour" notes on instruments before, and they usually give us goosebumps or make us squirm in our seats! What must we know so that we can avoid sour notes from synthesizers? Or to put the question more scientifically, what sort of precision must a home-built synthesizer attain?

Since we know that sound can be broken into three major components, let's analyze the precision needed for each, starting with frequency. The voltage-controlled oscillator (to be discussed later in greater detail) is responsible for creating waveforms of various frequencies. To generate a normal musical scale, it must be able to produce a range of notes controlled by the keyboard. To be tonal or melodic the notes thus generated must obey a certain relationship, and musicians call this the equally tempered scale.

In this particular scale, each note is the twelfth root of two higher than the previous note. In other words, the frequency of middle B can be found by multiplying the frequency of middle C by the twelfth root of two. This relationship is followed all the way up the entire scale. Figure 1-2 is a table showing the frequencies of the various notes in the equally tempered scale.

But here's the hitch: the twelfth root of two is an irrational number meaning that it cannot be stated as a simple fraction. Instead, it may only be expressed as a nonrepeating, nonterminating decimal fraction. This isn't our favorite type of number to work with! It is clear that a voltage-controlled oscillator and keyboard combination must follow an approximation to this scale. How close should the approximation be? As close as possible! Let's look into this more deeply.

As mentioned, each successive note generated by the keyboard/voltage-controlled oscillator combination is the twelfth root of two higher than the previous. Musicians call this interval a *semitone*. So, for example, from C to C-sharp is an interval of one semitone. Expressed as a percentage, the difference between these two notes is about a 6%. Now for more detailed measurements, we can break this down into smaller units. One "cent" is one one hundredth of semitone, and therefore represents a change of about .06%. We need this unit, a cent, to express the accuracy required by a synthesizer, and this may surprise you.

The average "man in the street," with no musical training at all, can hear a five or six cent variation in pitch! So if you play him a true C note, and then immediately play him a note five cents away from true C, he will probably hear the difference. This is all quite dependent on the timbre of the note and the listening environment, of course. In a laboratory, we might not notice any difference, but in the context of a song, a five-cent variation would probably show up as a "sour" note. Also, people tend not to notice sour notes at a live concert, but while listening to a recording in a controlled environment such notes become quite obvious. This is due, in part, to the social quality of a concert. We go to concerts for more than just the music; we also absorb the atmosphere, the light show, being with the crowd and so on. However, sitting at home in an easy chair is another matter and

C	16 352	32 703	65 406	130 81	261 63	523 25	1046 50	2093 00	4186 01
C [#]	17 324	34 648	69 295	138 59	277 18	554 37	1108 73	2217 46	4434 92
D	18 354	36 708	73 416	146 83	293 66	587 33	1174 68	2349 32	4698 64
D [#]	19 445	38 890	77 781	155 56	311 13	622 25	1244 51	2489 02	4978 03
E	20 601	41 203	82 406	164 81	329 63	659 26	1318 51	2637 02	5274 04
F	21 828	43 653	87 307	174 61	349 23	698 46	1396 91	2793 83	5587 66
F [#]	23 124	46 248	92 495	184 99	369 99	739 99	1479 98	2959 96	5919 92
G	24 499	48 999	97 998	195 99	391 99	783 99	1567 98	3135 97	6271 93
G [#]	25 958	51 913	103 82	207 65	415 31	830 61	1661 22	3322 44	6644 88
A	27 500	55 000	110 00	220 00	440 00	880 00	1760 00	3520 00	7040 00
A [#]	29 135	58 270	116 54	233 08	466 16	932 32	1864 64	3729 31	7458 63
B	30 867	61 735	123 47	246 94	493 88	987 77	1975 53	3951 07	7902 13

Fig 1-2. Equally tempered chromatic scale frequencies in Hertz.

we tend to listen more critically.

The point then, is that the average listener (under certain circumstances) can detect a five-to six-cent variation in accuracy. Recall that one cent is equal to .06 % and you can see that the tone generator must achieve at least a .3 % accuracy (5 times .06 %) to be considered musical! This is pretty fine work and so the general rule is that when it comes to frequency or pitch, a good synthesizer must not cut corners in any way, shape or form!

By the way, it is claimed that skilled musicians can detect a three-cent change in accuracy! This is about .18 %, a very small figure. Just to make sure that our synthesizers please everyone, we will shoot for this level of accuracy whenever possible.

DECIBELS, VOLUME AND DYNAMIC RANGE

When it comes to amplitude or volume, things lighten up considerably. As it turns out, the human ear is quite insensitive to changes in volume. Let's put this into a more scientific context by defining a needed term.

The basic unit of measurement for volume is the *decibel*. Unlike some units of measurement, however, this is a relative term and is based upon the ratio of amplitudes of two signals. As an example, you say "this amplifier has a gain of 100 decibels," and by this you mean that the ratio of the output of the amplifier with respect to the input is a certain number. Now, this figure is not a straight ratio, but in fact is related to logarithms, and other mathematical concepts. Fortunately, you don't need to know the details of this relationship; for our purposes, just remember that a decibel is a relative unit and compares the amplitudes of two signals.

Now that you have a general notion of the term decibel, I can go on to the concepts of precision and volume. The human ear, as discussed previously, is quite sensitive to the smallest change in pitch, and under certain circumstances a trained musician can detect changes of less than .18 %. But when it comes to volume the human ear is quite insensitive. Typically, the amplitude of a signal can change by as much as 10 % without the ear perceiving that anything has happened. Thus, while a great deal of accuracy is needed with regard to pitch, you can allow up to a 10 % change in amplitude. This will make the design and construction of amplitude circuits (VCAs, to be discussed later), quite simple.

Offsetting this simplicity somewhat is the fact that the ear is capable of detecting big changes in volume. For example, in a quiet room you can hear a pin drop, but on the other hand, at a construction site you can hear the roar of heavy machinery, the sonic boom of a jet passing over and the clap of thunder. It is clear that we are able to detect a very broad range of volumes with our ears. Recalling the previous discussion concerning decibels, this can be formalized somewhat. Suppose you have two amplitudes to compare: the pin dropping and the roar of machinery. In terms of decibels it can be said that human hearing has roughly a 100 dB (decibel) dynamic range.

100 dB is a very wide range, and the circuits to recreate this range must be fairly fancy. However, it's not an insoluble problem, and later you'll see several ways to design and build such circuits.

Before moving on, let's summarize the accuracy requirements for amplitude circuits. Since the ear is relatively insensitive to changes in volume, you need only approach 10 % accuracy in the design of such circuits. On the other hand, these circuits must have a very broad dynamic range.

Harmonic content or timbre type circuits don't really have the accuracy problems of the circuits mentioned above. When I discuss the VCF (voltage-controlled filter) a little later in this book, I'll mention some of the requirements. For right now, however, it suffices to say that filters for laboratory work need to be quite accurate with regard to cutoff frequency and stable with respect to temperature, but for music they don't need nearly as much accuracy.

SUMMARY

This chapter introduces the basic concepts of sound. You've seen that all sound can be broken down into three major components: frequency, amplitude and harmonic content. Musicians often use the terms pitch, volume and timbre as synonyms for the previous three terms, but we've seen that there really is a difference. Frequency, amplitude and harmonic content are objective concepts; pitch, volume and timbre are the ear's and brain's perception of the latter three. Then the decibel was encountered in a very informal way and you discovered that it is a relative unit of volume. Next the level of accuracy needed for sound synthesizers was discussed and we concluded that frequency circuits need to be very accurate and amplitude circuits not nearly so.

The discussion so far has been quite theoretical. In the next chapter you will meet some of the modules needed to create a practical synthesizer.

Chapter 2

Synthesizer Modules

A complete electronic music synthesizer is composed of a number of modules, usually housed in one cabinet. Each module is an electronic circuit designed to control one of the three parameters of sound mentioned in Chapter 1. Thus, there are modules to control the frequency of a waveform, its amplitude and its harmonic content.

Though there are many different types of synthesizers on the market today, the fact remains that all of them contain circuits to modulate or control the three parameters of sound. Whether it's a small, portable unit, a dedicated device (like a string machine), or a large studio-type synthesizer, you can expect to find various electronic circuits for affecting frequency, amplitude and harmonic content.

Obviously, when inspecting the circuits that make up a complete synthesizer we may catalog or classify them as frequency, amplitude or harmonic content changing circuits. Almost all circuits in a synthesizer will fall under one of these headings. Another way to classify a circuit is by noting whether it deals with audio signals or control voltages. Thus, a circuit may be either an audio circuit or a control voltage circuit. There are other ways of identifying and classifying modules in a synthesizer, but the two schemes just described will be sufficient for most purposes. But don't get too carried away; some circuits may defy description or fall into several categories at once.

THE VOLTAGE-CONTROLLED OSCILLATOR

The heart of any synthesizer is the VCO (voltage-controlled oscillator). It is the duty of this circuit to produce a waveform whose frequency may be controlled by an input voltage (hence the name). So, for example,

if the input voltage is 1V, then the output frequency might be 500 Hz. When the control voltage is 2V, the frequency becomes 1000 Hz, and so on.

The frequency of a VCO depends upon the control voltage present at the input, but how should it be related to the control voltage? Should the frequency depend upon it directly, inversely, linearly, exponentially, or what? In other words, of the countless ways in which the output frequency might vary for a given input, what type is the most useful for musical purposes?

As it turns out, the easiest and most versatile approach is to adopt a one-volt-per-octave response. As seen in the example above, a change of 1V caused the frequency to double, and when the frequency doubles, a note one octave higher is produced. We say, then, that a change of 1V causes a change of one octave.

But what kind of response is this? Extend the example given above and look at the results. For example:

Input Voltage	Output Frequency
1V	500 Hz
2V	1000 Hz
3V	2000 Hz
4V	4000 Hz
5V	8000 Hz

It's pretty obvious that this isn't a linear relation, for if you were to draw a graph of the function you would see a "curved line" rather than a straight line. In fact, the relationship is exponential in nature and since jumping an octave means multiplying the frequency by two, you can see that the "base" of this exponential function is 2.

Thus, a VCO, to be maximally useful for electronic music, should have an exponential response. By going with this type of response the design of keyboards is considerably simplified, since like-valued resistors can be employed in the typical resistor string switching arrangement. Or if you're contemplating a digital keyboard, the design of the digital-to-analog converter is greatly simplified. The point is, by making the VCO exponential in response, whatever is controlling it (analog or digital keyboards for example) can have a simple linear output. By way of contrast, consider the consequences of using a linear VCO. Achieving the equally tempered scale with such a VCO requires the construction of an exponential keyboard! Such a keyboard needs odd values of resistors or lots of trim pots, or both! This is clearly more work than it's worth.

But there's another good reason to go with an exponential VCO and linear keyboard rather than the other way around. This sort of arrangement is capable of *tracking*. Tracking is a term which refers to the ability of one VCO to follow another in some precise manner. For example, you

might want to have one VCO follow another, but with an interval of a fifth between the two. Once set, the two VCOs will maintain this interval up and down the entire scale. While it is possible to cause two linear VCOs to track under some circumstances, the difficulty and instability involved is a rather steep price to pay.

The major drawback of exponential VCOs is that unless good design procedures are followed, they tend to become detuned with a change in temperature. Quite a bit of research has been done in this area, though, and today it is fairly easy to design VCOs with excellent temperature characteristics.

Finally, for maximum versatility, a synthesizer VCO should have a variety of outputs available. Sine waves, triangle waves, ramp waves, pulse waves and so on should all be handy if possible. The more waveforms available, the more sounds your synthesizer will be capable of.

These are the basics needed for a good VCO; as you will see later on, there are all sorts of interesting extras that may be added (See Chapter 5).

THE VOLTAGE-CONTROLLED AMPLIFIER

The next module in a synthesizer is the VCA (voltage-controlled amplifier). As the name suggests, this is an amplifier whose gain may be altered by means of a control-voltage input. A signal is pumped into the VCA and the amplitude of the signal which comes out will be greater or smaller depending on what the control voltage is doing. You can see, then, that a VCA has two inputs; one input is for the signal (usually an audio signal, but not always), and the other is a control voltage.

VCA circuits show up in many areas of electronics. For example, some types of noise reduction equipment for recording studios employ VCAs, as do automated mixdown units. Telephone circuits and some radio circuits use VCAs as well. These types of VCAs can get pretty exotic, but fortunately for most electronic music purposes you can get by with simpler and less expensive units. In Chapter 7 when the circuit for a real VCA is described, you'll learn about response, signal levels, and other factors crucial to the VCA. Right now all you need to know is that a VCA modulates or controls the amplitude of a signal.

THE VOLTAGE-CONTROLLED FILTER

Now stop a minute and ponder this question: what's the difference between a triangle wave and a square wave? Well, obviously if you have an oscilloscope, you can tell the difference just by looking. One waveform is triangle shaped and the other is square shaped. But in terms of hearing, what's the difference? Everyone's ears and listening abilities are a little different, but most would agree that a triangle wave is rather pure or simple sounding, while the square wave is sharp or harsh sounding.

You learned in Chapter 1 that this difference in "tone quality" can be

explained by timbre, or as the scientist would call it, harmonic content. What circuits exist for changing the harmonic content of a waveform? The VCF, (voltage-controlled filter) is the tool you need. However, instead of the singular, you might want to answer in the plural—VCFs—since there are many, many different types of filters available. There are lowpass, highpass, bandpass, allpass, notch, two pole, four pole, and state variable filters; the list goes on and on. But the one thing that all filters have in common is that they modulate or control the harmonic content of a waveform.

A VCF is like a VCA in the sense that it has two inputs and one output. One of the inputs is a signal input. You would typically connect the output of a VCO to this port. The other input is for control voltages. As the control voltage is varied, the harmonic content of the output waveform changes as well. “Wah-wah” circuits for guitars, so common several years ago, are a good example of what one of these filters might sound like.

The response of the filter (cutoff frequency versus control voltage) is also exponential in nature, just like the VCO. All filters have this feature in common, but beyond this, explaining the action of a filter becomes a little more complicated. Until you know just what type of filter you’re dealing with, it’s hard to pin down its effects. In Chapter 6 you will see how to build a lowpass VCF, and it should be clear that this is a very simple filter to understand.

ENVELOPE GENERATORS

The modules described thus far (VCO, VCA and VCF) have all been audio modules in the sense that they usually modify or control audio signals. (There are exceptions to this. For example, a VCA might amplitude modulate a control signal.) But the VCO, VCA and VCF can do little by way of creating interesting sounds unless they are controlled or modulated in some manner. An envelope generator is an example of a control module which might accomplish this aim. Let’s look at envelope generators in some detail.

When an organist strikes a chord, the sound is instantaneous. And when the chord is released, the sound dies away at once (assuming the sustain is not being used). Contrast this with a violin; as the bow is drawn across the strings, vibration builds up slowly but surely. At some point the sound reaches maximum volume. Then when the bow is lifted away from the strings, the sounds decay. Note that the vibrations do not stop all at once, but linger on for a few moments. The point here is that the organ is quite different from a violin in the way in which its amplitude envelope operates. The time taken for the instrument to reach maximum amplitude is called the *attack time*; likewise the time taken for the sound to die away completely is called the *decay* or *release time*. It’s clear that the organ has fast attack and decay, whereas the violin has a relatively long attack and decay.

The VCA can modulate the amplitude of an input signal, but it needs

to be told how to modulate the signal. Should it create a fast attack and decay, or should the times be slow? It's the duty of the envelope generator to tell the VCA how to perform these actions. As the name implies, the envelope generator creates an envelope which is impressed on the amplitude characteristic of the input sound.

Most instruments have fixed attack and decay characteristics. Thus, guitars, flutes, trumpets, drums and so on, have no real way of changing their amplitude envelopes. This is one area where the electronic music synthesizer shines in all its glory, for it may change the envelope of a sound with the twist of a knob. Just about any envelope can be created, from the natural to the really far out.

There are several different types of envelope generators available today. One kind is the AD or attack-decay type. The AD unit allows you to vary the initial attack and decay characteristics. Plucked string instruments, like the guitar, often have an AD envelope. You pluck a string and initiate the attack cycle. After maximum volume is attained the vibrations decay to silence. Figure 2-1 is a graph of the output of an AD generator, and shows how a trigger signal (generated by the synthesizer in some manner) initiates the envelope. The first waveform has a medium length attack and decay while the second one has a very short attack time and very long decay time.

Some instruments, like the violin, don't follow this pattern, though. Think about it; the bow is applied to the strings and the sound builds up to some maximum volume. But unlike the guitar, the sound doesn't decay right away but continues for as long as the bow is in motion. Only after the bow becomes still does the sound decay. This type of envelope pattern is known as the AR (or attack-release) response. The attack initiates the sound, but the sound doesn't go into its final decay or release until after

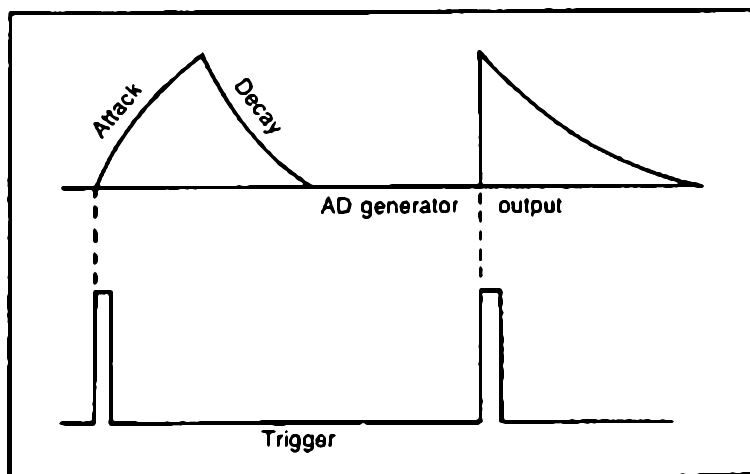


Fig. 2-1. The output of an AD generator.

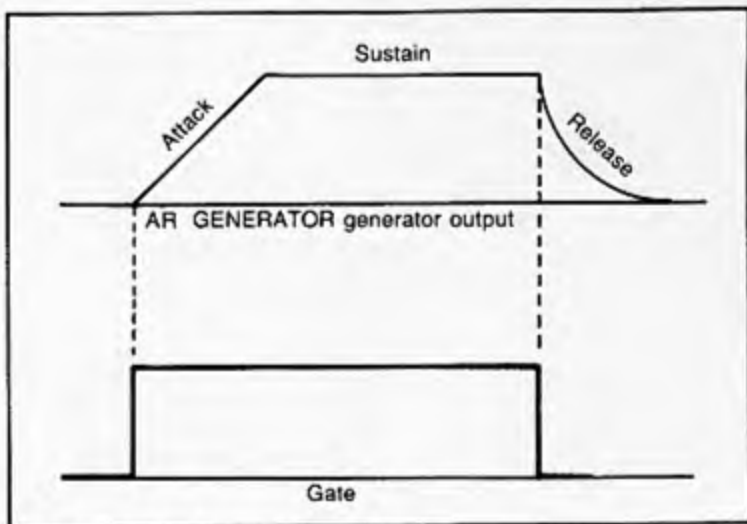


Fig. 2-2. The output of an AR generator.

a certain period of time, known as the sustain. During an instrument's sustain time, the volume remains fixed or steady. I said that a violin sustains as long as the bow is in motion. Another example is the steady sound of an accordion while the bellows are in motion. A synthesizer *doesn't* use a bow or a bellows, and so must depend upon some other means of sustaining the sound. Instead of a trigger signal, as required by the AD type generator, an AR unit requires a gate signal. The gate is the synthesizer's equivalent of a bow or bellows; as long as the gate is high, the sound is sustained.

Figure 2-2 shows the graph of an AR response, with the required gate signal shown as well. When the gate is initiated, the voltage rises to the maximum level and holds there for as long as the gate is present. As soon as the gate is release, the waveform dies away to zero volts again.

So an AD is used for percussive or plucked effects and the AR is utilized for sustained sounds. But do these two different types of envelope generators cover all possible sounds? No, they don't. It has been shown that some instruments, like the piano, have a more complicated envelope pattern. The name given to this pattern is ADSR, which stands for attack-decay/sustain-release.

As the initials of the name suggest, an ADSR allows the user to vary the attack, decay and release times of the envelope as well as the sustain level. You've already encountered the notion of attack time. Quite simply, this is the time it takes for the sound to reach a peak level. The term decay, in this instance, means initial decay. After the sound has reached a peak level, it will decay to some steady state known as the sustain level. The envelope will remain at this level until some condition has been met

(the bow stops moving, bellows stop puffing, gate is removed). After the sustain portion of the curve, the release occurs and the sound will die away to silence. Now it's important to realize that the decay is not the same as release. Both of these terms indicate that the envelope is diminishing in amplitude, but in the case of decay the sound does not die away completely, but goes to a sustained level. Release, on the other hand, allows the sound to die away altogether. Some manufacturers call decay *initial decay* and release *final decay*, but this tends to be a little confusing.

Figure 2-3 shows the graph of a typical ADSR waveform. Note that both a trigger and a gate are required to fire this type of envelope generator. The trigger, of course, tells it when to go into the attack position, while the gate specifies the length of time for the sustain to occur. In Chapter 8, you'll see how to build this type of envelope generator and also add on a new feature known as retriggering.

An envelope generator, no matter what type it may be, puts out a con-

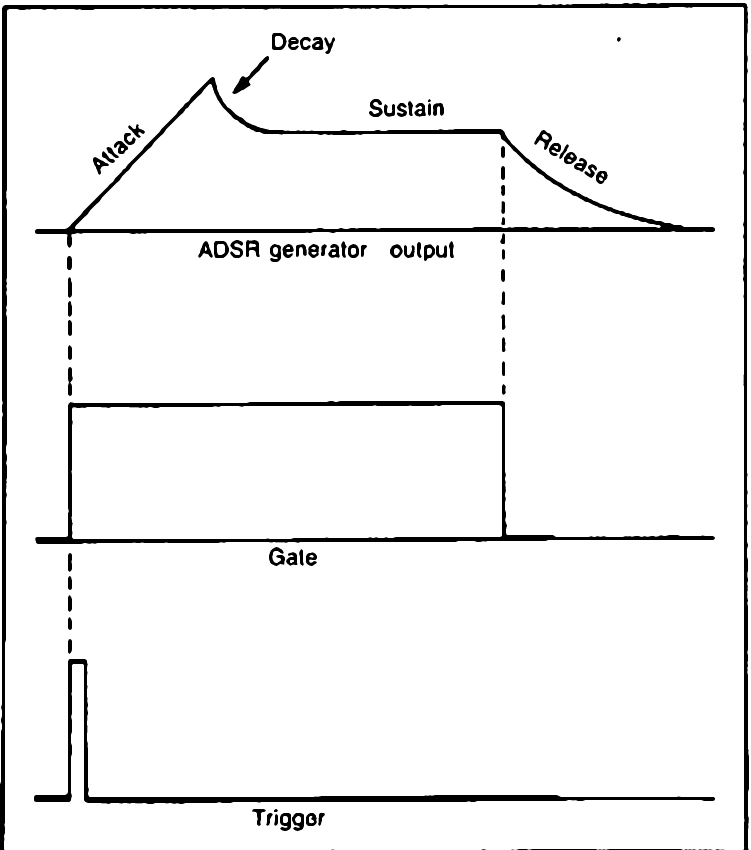


Fig. 2-3. The output of an ADSR generator.

control voltage which will follow a certain pattern. It is this control voltage that tells the VCA, for example, how to open and close, and thus impart an envelope onto the sound.

In most systems, a keyboard is used to tell the envelope generator when to fire. For example, you push a key down and the envelope generator starts to produce a control voltage and this in turn controls the VCA. Whatever sound is going into the VCA is increased and decreased in amplitude under control of the envelope generator. Now you can start to see the flexibility of the system. If you want violin type sounds, you simply turn up the attack time. If you want organ sounds, you turn down the attack time. Thus, an envelope generator and VCA combination is able to produce a broad range of sounds with the twiddle of a knob.

The keyboard tells the envelope generator when to produce a control voltage. It does this by means of gates and triggers. Anytime a key is depressed, no matter what key it may be, a gate signal is produced. Thus the presence of a gate signal means that at least one key has been depressed. A trigger signal, on the other hand, tells the unit that a new note has just started. It's clear, then, that a gate signal indicates an ongoing event, while a trigger indicates the start of an event. In modern synthesizers, a gate will be a dc level of either 0V (no keys depressed) or +5V (at least one key is depressed). A trigger signal is a quick transition from 0V to +5V back to 0V. Typically, the trigger is one millisecond wide.

In Chapter 10 a complete do-it-yourself keyboard is described. At that time a fuller description of gate and trigger signals will be given. For now, however, all you need to know is that the trigger signal tells the ADSR to start attacking, while the gate signal indicates sustain time.

An envelope generator can modulate more than just amplitude, of course. A very common "patch" is shown in Fig. 2-4 and it can be seen that the envelope generator controls both the VCA and VCF. Note that the keyboard puts out the gate and trigger signal, as described above, and

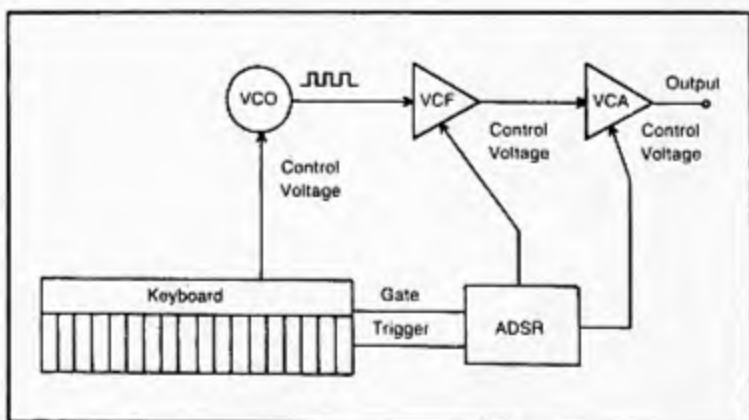


Fig. 2-4. A typical patch using an ADSR generator.

these fire the ADSR. In addition, the keyboard also puts out a control voltage and this modulates the VCO according to the one-volt-per-octave protocol mentioned earlier. Thus, with just a keyboard, VCO, VCA, VCF and envelope generator you can create some pretty respectable synthesizer sounds. But to get some really dynamic sounds, something more in the way of controller-type modules is needed.

LOW-FREQUENCY OSCILLATORS

The LFO (low-frequency oscillator) is one such module. As the name suggests, this unit oscillates at a low frequency. How is it used? Well, one good example is in the creation of *vibrato*. Vibrato is the sound which occurs when the frequency of a note is modulated in a periodic fashion. The pitch "warbles" slightly and this is actually a case of frequency modulation (FM). Many instruments, including the human voice, violin, and saxophone, use vibrato to add color and expression to their sounds.

Studies have shown that vibrato is most pleasing when it occurs at a rate of about 7 Hz. (There's nothing magical about this number; other rates are possible, depending on the style of music, but for most typical songs, 7 Hz seems about right). How can the VCO be made to oscillate in this fashion? First, it must warble sharp, then flat at about a 7 Hz rate. This is where the LFO comes in; an LFO can oscillate at this frequency quite easily and is used to control the VCO. Figure 2-5 shows a patch for vibrato or FM. Note that the keyboard specifies the basic or center frequency, while the LFO causes it to deviate in a periodic manner.

In the past, people have tended to confuse the notion of vibrato with that of tremelo. As you have just seen, *vibrato* is an actual pitch deviation effect or FM. *Tremelo*, on the other hand, leaves the pitch intact and instead varies the amplitude of the signal in some periodic manner. Tremelo is amplitude modulation (AM). It is true that under some listening circumstances it is easy to confuse tremelo with vibrato, but it should always be born in mind that the effects are distinctly different and result from different causes.

At this point, it should be clear that voltage control is the key to how synthesizers work. A synthesizer has many dozens, perhaps even hundreds of parameters, and changing just one parameter will alter its sound. If all these had to be changed by hand, it would take many dozens of people all twiddling knobs at a furious rate just to realize the simplest sounds! But by letting the parameters be capable of voltage control, you can cause many different things to happen at once. For example, in Fig. 2-5, the frequency of the VCO is easily modulated by both the keyboard and the LFO.

THE NOISE SOURCE

You've seen quite a few modules now, and there appears to be no limit to the number of sounds which can be created. Just take a VCO for a sound

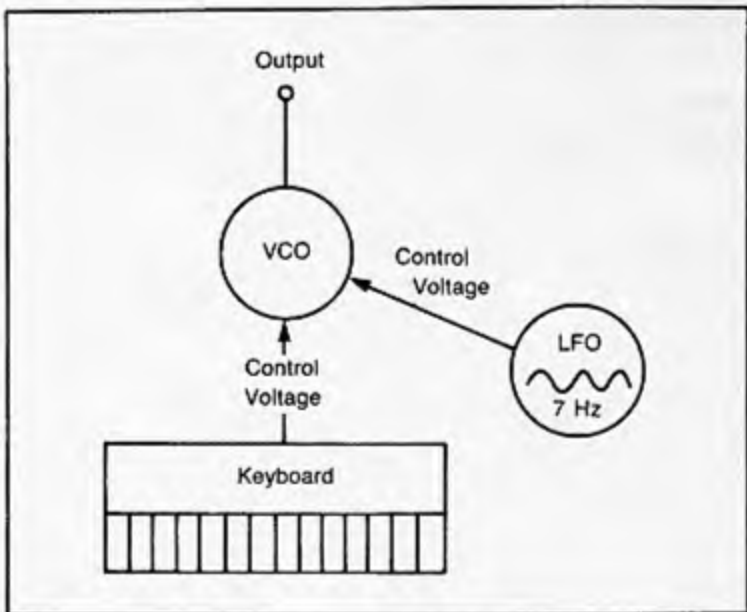


Fig. 2-5. Adding vibrato to a tone.

source, put it through a VCA and VCF to alter its amplitude or harmonic content, control the envelope with an ADSR, and perhaps modulate the frequency a bit with an LFO. It seems as if any sound can be created with this handful of modules. But does this cover all the possibilities?

In answer to this, consider the snare drum. How would you recreate the sound of this instrument with the modules just described? Clearly a VCO is next to worthless, since the sound it creates has pitch and a snare drum really doesn't have just one pitch. In fact, the sound of a snare drum is quite complex and can't be imitated with a VCO. What you need is another sound generator which is "ragged," "noisy," "percussive," and so on. In short, you need a noise source.

A *noise source* generates white noise and this is similar to the hiss or roar that a radio creates when it is between stations. *Noise*, as used in the musical sense, means sound which contains a vast number of frequencies. The term *white*, when used to describe noise, means that the sound contains all possible frequencies, just as white light contains all possible colors.

In everyday life we tend to view noise, including white noise, as objectionable. We all instinctively dislike the sound of hiss between stations on a radio, and phonographs that hiss make most of us want to climb the wall! But actually there are some sources of noise that are pleasant. Water running and rain falling are good examples; who doesn't find these two sounds soothing? And when it comes to music, white noise is quite common. The snare drum, cymbals, and many percussive instruments depend upon it.

So if a synthesizer is to be complete, it should really have a source of white noise. Figure 2-6 shows an interesting patch that makes use of white noise. The noise is sent through a VCF and VCA. As the VCF is modulated the noise will "howl" and change its basic "tone," while the VCA changes its volume. The LFOs sweep the VCF and VCA back and forth thus creating a realistic wind storm, replete with howling, dying winds, bursts, etc.!

THE BALANCED MODULATOR

Since I was just discussing percussion instruments, this might be a good time to mention the balanced modulator. A balanced modulator has two audio inputs and one output. If you apply two sine waves at the inputs, the output will yield a mix of sine waves. However, this signal is not just a hodge-podge of sounds, but in fact consists of sidebands of the original frequencies. As it turns out, the output signal will consist of two frequencies: the sum and the difference of the input frequencies.

To get a hold on the concept, consider this example. You apply a 200 Hz sine wave and a 300 Hz sine wave to the input of a balanced modulator. At the output, you will find a 500 Hz tone and a 100 Hz tone. Note that neither of the original frequencies are present at the output; only the sum and the difference of the inputs occur.

You may wonder what the balanced modulator could possibly be good for. Well, many natural sounds like gongs clanging, chimes chiming, and trash can lids clashing, exhibit a balanced-modulator effect. Instead of just one tone, you will hear a mix of tones, usually a sum and difference of some other frequencies. In passing it should be noted that balanced modulators are most often used for special sound effects of short duration rather than prolonged musical passages, since their sound tends to get on your nerves quite rapidly. Be that as it may, if you plan on doing any percussion synthesis, a balanced modulator is a must.

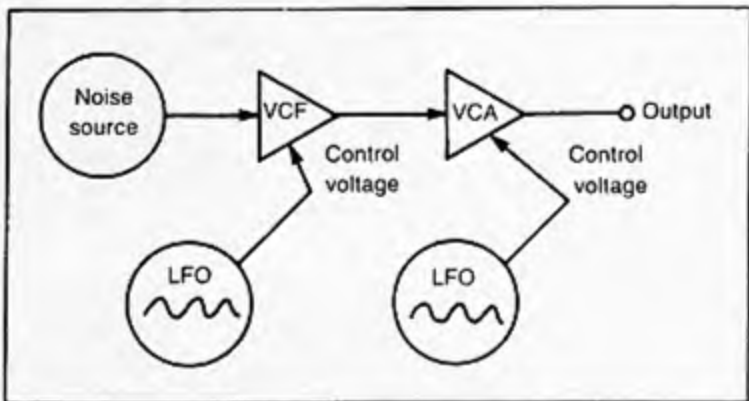


Fig. 2-6. Creating a windstorm effect with a white noise source.

Incidentally, in the very early days of electronic music, these units were known as *ring modulators* since the circuit was configured around a ring of diodes. These ring modulators were rather primitive and since they were passive in nature suffered from signal loss. With the advent of integrated circuits, ring modulators have faded away completely and are now superseded by the balanced modulator.

THE KEYBOARD

You've already met the keyboard briefly. Generally it will look and feel like an ordinary organ keyboard, but as you have seen, it must in fact generate three different signals. These are the gate, trigger and control voltage. The gate and trigger are used to fire the envelope generator, while the control voltage is used to modulate the frequency of the VCO. Unlike an organ, the synthesizer keyboard does not pass or modify audio signals. It merely puts out dc voltages that tell the other modules what to do. (However, in Chapter 10 you will see how to convert an ordinary organ type keyboard for use with a synthesizer.)

The keyboard is in a special class of synthesizer modules, and is known as a controller. A *controller* does exactly what its name implies—it controls the other modules. Historically the organ-type keyboard has come to be associated as the main controller for a synthesizer, but there really isn't anything sacred about it. In the old days many musicians with a traditional music background drifted into electronic music synthesis and found an organ keyboard to be the easiest to adapt to. Nowadays, though, many electronic musicians are entering the field without necessarily having a traditional keyboard training. These sorts of people don't have any preconceptions and are willing to try other modes of control.

For example, several years back some enterprising experimenters started to dabble with a "saxophone" controller. This controller wasn't really a sax, and it didn't depend on air rushing through it to vibrate a reed. But the musician held it like a saxophone and the fingering was identical as well. The upshot was that a sax player could enter into the field of electronic music without having to learn any new technique. He simply played the synthesizer with a controller which acted like a saxophone.

This notion of making a controller behave like a traditional instrument is becoming more and more popular. People feel freer and are more willing to try new and exotic concepts. Organ keyboards and sax controllers are just the start!

EXOTIC CONTROLLERS

The last several years have seen the advent of many astounding new concepts. How about a theremin controller? A *theremin* is an old instrument from the '50s that is played by moving your hands around two antennas. One antenna controls the volume while the other changes the pitch

of an oscillator. The sound is quite eerie, as is often to be heard in old science-fiction movies. The theremin isn't heard much anymore, but some clever people have adapted it to synthesizers. Now when you move your hand closer to the antenna, a greater control voltage is generated, and this in turn can be used to modulate some synthesizer circuit. It's quite a sight to see one of these in action. The performer's hands never touch the instrument!

Another controller that falls into the "weird" category is the rotary controller. This is generally formed by attaching a long stick to a potentiometer and mounting the whole thing in an easy-to-hold case. The performer plays the instrument by waving, jiggling and swirling the stick. This is a good controller to experiment with since it is quite easy to make and very inexpensive. Performing musicians may want to fasten a strap to the affair so that it can be worn like a guitar.

Perhaps the most unique and musically useful controller to come along in quite some time is the pressure sensor. This is a pad, about three inches square, with a pair of wires emanating from it. The performer plays it by alternately pressing and releasing the pad. The output voltage, which can control VCOs, VCAs, or in fact any synthesizer module, increases as the pressure increases. This is a difficult instrument to use all by itself, but is extremely handy when used in conjunction with a keyboard. For example, the right hand might play a melody on the keyboard while the left hand presses the pad, thus creating pitch bends and *glissando* effects.

Before leaving the subject of controllers some mention should be made of pitch and envelope extractors. Envelope extractors are the easiest to understand (and hence make in the home workshop), so let's consider them first. Basically, some traditional instrument is plugged into the envelope extractor and the output will be a control voltage which varies directly with the instrument's amplitude envelope. For example, plug a guitar into the envelope extractor (sometimes called an envelope follower) and monitor the output. The harder you strum the guitar, the greater the output voltage. We say that the control voltage follows the envelope, or alternately, the envelope has been extracted. Note that the pitch is ignored; only the volume parameter is detected by the unit.

A *pitch extractor* (or follower), on the other hand, monitors some input signal and presents a voltage to the output which is directly proportional to the pitch of the instrument. Thinking of the guitar again, with a pitch extractor, the higher the note played, the higher the output voltage. This voltage, by the way, could be fed to a VCO and then the VCO would exactly track the pitch of the guitar. It should be clear that the pitch extractor is a monotonic device. It can only follow one note at a time. So chords are out; it may only be used for solo applications. It probably comes as no great surprise that the fundamentals of pitch extraction are very complex, and indeed many designers consider this to be the single most difficult problem in all of electronic music.

THE POWER SUPPLY

The last synthesizer module to be described (but generally the first you will build) is the power supply. This is perhaps the single most overlooked module in the system and that's a shame. If the power supply is well designed and working well, then the whole system has a good chance of working well. But if designed and implemented poorly it could well cause the entire system to malfunction.

Basically, the power supply is responsible for supplying a steady dc voltage to the entire synthesizer. This voltage powers all of the modules. Of course, wall current is 110V ac and this is entirely unsuitable for modern solid state work. The power supply, then, converts this ac voltage to a steady source of dc. Most modern systems require a bipolar supply with an extra +5V supply being optional. The bipolar supply will be used by the entire system, and the +5V supply will be needed if any TTL integrated circuits are used.

Note that I have emphasized the need for a steady output. Ripple, hum and noise can be detrimental to the system and generally occur when the supply voltages haven't been properly regulated. Fortunately, it is quite easy to accomplish this task nowadays with the ready availability of integrated circuit voltage regulators.

SUMMARY

At this point you've seen quite a few different types of modules: sound sources like VCOs and noise generators, modifiers like VCAs and VCFs, controller circuits like LFOs, keyboards, and pressure sensors. They're all necessary for a complete synthesizer. However, these don't exhaust the list by any means. Future modules for you to think about are animators, phase shifters, delay lines, sequencers, and more. If you keep up to date with the literature then you'll see that there is quite a bit of research going on in electronic music. By no means have all the different types of modules been invented! (See Appendix B for a list of books, journals and magazines). But this is all for the future. Our goal now is to get a "standard" synthesizer up and running. Only after you have mastered the basics should you consider delving into these more arcane topics.

Chapter 3

Electrical Standards

Designing a stand-alone circuit is often quite simple. Since, by definition, such a circuit isn't intended to be interfaced to anything else, the designer can often cut corners and make compromises with respect to voltage levels, power supply, impedances and so on. However, when designing a whole set of modules which are intended to work together, careful consideration must be given to how the individual units will interconnect. In a case like this, some design standards are necessary.

The selection of a good set of standards is not a trivial affair. Many commercial synthesizers which would otherwise have sounded quite good become nightmares when you attempt to interface them with other equipment. And that's the beauty of making your own synthesizer. You can make it any way you see fit and will never have to cut corners to save manufacturing expenses. By the time you're done reading this book you will be convinced that you can build a better synthesizer than you can buy!

So let's look into this business of standards. This chapter will take you through the ins and outs of power supplies, impedances, signal levels and so on. Every construction project in this book (coming up in later chapters) follows these criteria, so it will pay you to read and reread this material. This way you will become familiar with the wants and needs of a modern synthesizer, and when a certain design decision is mentioned later on, you'll know why it was made.

SUPPLY VOLTAGES

The power supply is a fundamental circuit in a synthesizer and obviously provides power for all of the modules. If you look carefully through data books, you will notice that many integrated circuits give optimum per-

formance when driven by a bipolar 15V power supply. All op amps work well at this level, as do CMOS circuits, transistors, and most linear ICs. Additionally, some circuits, such as the Mostek 50240, the SAD-1024, and various rhythm-generating chips won't work well at all on lesser voltages. For this reason a plus and minus 15V is picked as the standard power-supply voltage.

This has one serious limitation. A high supply voltage immediately eliminates any TTL ICs from consideration since this breed of IC only works at +5V and no other voltage. Losing TTL chips is no great loss, though, since most of the functions in electronic music that we would normally accomplish with TTL can be performed better with CMOS. Of course, CMOS works quite well at this level and as an extra bonus draws very little current, thus reducing the load on the power supply substantially. (Generally, TTL is selected over CMOS when clock signals of greater than several MHz are used. Most electronic music circuits have no use for such high frequencies.)

In the past, it was common to specify an additional +5V supply, but recently this isn't quite as common for the reasons given above. This greatly simplifies things; start with a center-tapped 35V transformer, a handful of diodes, capacitors and regulators, and two hours later you can have power for your entire system. However, many experimenters getting into computer-controlled synthesizers, will want to include the +5V supply because most computer circuits will work on no other voltage.

Many beginners often wonder what current rating a power supply should have. The answer to this question depends, of course, on the number and type of modules in the system. Large studio type synthesizers will need big power supplies while portable road units can get by with much smaller supplies. As a general rule one amp per side of the power supply is just fine and will in fact power several dozen modules quite easily.

SIGNAL LEVELS

Now let's talk about signal levels. All audio signals should be 10V peak-to-peak. Another way of saying this is that they should swing between -5V and +5V. Why these values? Well, you want to keep the signal as "hot" as possible to obtain a good signal-to-noise ratio. The hotter the signal the less the relative background noise level. And the same applies to hum and any other type of interference. You might think that you should go to 20V or even 30V signals, acting on the notion that "the hotter, the better." There are good reasons for not going to this extreme, though. The main reason is that most common op amps (the mainstay of all modern synthesizers), have relatively poor output slew rates. The notion of slew rate can get quite technical, but the basic concept is easy to understand. If the slew rate of the op amp is poor, then basically the output stage can't keep up with a large signal swing at a high frequency. The result is a rounding of sharp waveforms and what should have been a crisp tone will sound dull. A swing

of 20V or 30V makes the effects of a poor slew rate quite obvious since large voltage swings are involved. But at 10V peak-to-peak, even common op amps work quite well. So 10V peak-to-peak is a nice compromise giving a relatively good signal-to-noise figure with good frequency response.

CONTROL VOLTAGES

What about control signals? Things are not quite so critical here, since in general no high frequencies are involved. A standard that has become quite popular in recent years is to allow control signals generated by units such as ADSRs, keyboards, pattern generators, etc., to swing from 0V to +5V. There's a good (albeit somewhat disguised) reason for this value of +5V. +5V is exactly one third of the supply voltage of +15V, and often control signal circuits involve comparators that must trigger at some level below full supply. As it turns out, more often than not, one-third of the supply voltage (or what's pretty much the same thing, two-thirds of supply) is often the comparator threshold level. By taking advantage of this relationship, you can often reduce the parts count of a circuit drastically.

Both control signals and audio signals must obey one fundamental rule: they should be dc or direct coupled. By making sure that every signal running through a system is direct coupled, it is possible to use control signals for audio and audio signals for control. With this setup there's no need to differentiate between the two. This may not sound that useful, but many unusual patches are only possible with such a scheme. Direct coupling also implies that the synthesizer will have the widest possible frequency response. Finally, the same type of patch cord may be used for both audio and control voltage connections.

A good example of the freedom afforded by dc coupling occurs in the case of a VCA modulating an envelope signal. By opening and closing the VCA, it is possible to present a greater or lesser envelope signal to some other device. If the VCA were coupled with a capacitor, then this scheme would fall flat. So for maximum versatility, design and build each module (where possible) to allow direct coupling.

OUTPUT STRUCTURES

In addition to voltage level and coupling considerations, you must also take into consideration the output impedance. As a general rule of thumb, if you make all output impedances one thousand ohms (1k), you really can't go wrong. Why? Well, if all synthesizer outputs are standardized to the same value, then it is possible to mix signals simply by ganging the outputs through a *multiple*. (A *multiple* is simply several jacks wired in parallel.) For simple applications there's no sense in tying up a good mixer when a multiple will do the job. Note that this sort of mixing is only possible when all of the outputs have the same impedance. But there's another reason for selecting the value of 1k. If the output is limited to this value, then

an excellent short-circuit protection has been added. You can short the output directly to ground for an indefinite period of time and no harm will be done! Since synthesizers are often used in uncontrolled environments, any extra "goof-proofing" which can be added is always more than welcome!

Before moving on to input considerations one final thing should be said. All outputs should be available in a "simple" form only. By simple, it is meant that an output should terminate in a jack and nothing else. Don't hang a potentiometer on the output; the input is the place to do this. If you control the output of a circuit, you've just lost versatility since every module controlled by that circuit is now stuck with the same voltage. But if an output is simple, and the input of each unit being controlled has its own pot, then it follows that each module can dial in the amount of modulation desired.

INPUT STRUCTURES

Okay, so you've agreed to put potentiometers only on the inputs of devices. What else do you need to know about inputs? For starters, all inputs should have an impedance of 100k or thereabouts. If this value is much higher, you are faced with hum and noise problems, but if it's much lower then the driving circuit will be loaded down. Since you've decided on output impedances of 1k (see above) and the inputs are 100k, it's clear that loading will be on the order of 1%, a very reasonable value.

100k is also a nice selection since 5% and 1% resistors are easily available in this value. Additionally, potentiometers (and every synthesizer has lots of these) including multiturn units, come in this value.

TRIGGER SIGNALS

The standards discussed above cover most common design situations. One final area of concern, though, is the desired pulse width of a trigger signal. As mentioned previously, gate and trigger signals should swing between 0V and +5V. A gate signal is simply ON or OFF for a length of time set by the user. But what about triggers? A good rule of thumb is to set triggers to be 1 millisecond wide. There's no reason to go overboard on this, though. Anything from half a millisecond to two or three milliseconds is just fine. It is usually sufficient to estimate the time constant, and then pick some standard resistors and capacitors that cause the circuit in question to approach this value.

The standards described thus far cover quite a bit of ground and the discussion may have seemed just a trifle theoretical. When the construction projects are described in Chapters 5 through 14, though, their use will become much clearer.

SELECTING COMPONENTS

Related to the notion of standards is the art of component selection.

Take capacitors for example. There are electrolytics, disk, mylar, polystyrene, mica and many other kinds. How does the experimenter know what kind to choose for what task? Well, that's a tricky question and a complete answer would take us through the complex realm of reliability analysis, quality control, temperature design and so on. To build synthesizers you don't really need to know all the theory, so let's develop some rules of thumb on how to pick the right component for the job.

Selecting Capacitors

As noted above, there are quite a few different types of capacitors. Let's talk about the most common types and when and where they should be used. Disk capacitors are perhaps the most common. When should they be used? As it turns out, disk caps don't come in very close tolerances and tend to drift with temperature. This being the case, they should not be used in sensitive, tuned circuits. For example, they would be a poor choice for the main tuning capacitor in a VCO. However, they are a good choice for bypass and decoupling applications. In cases like this, the exact value doesn't matter much and any change with temperature is inconsequential. So imagine a digital circuit with lots of switching and clocking going on. To keep all of that switching "garbage" from being imposed on the power supply lines, you would typically drop a 0.1 mfd. disk capacitor across the supply pins of each digital chip. Note that this is a case where home-built can be made better than store-bought equipment. Manufacturers have a tendency to cut corners and not really put in enough decoupling and bypass caps. But you can!

Mylar capacitors are the next ones to consider. These generally come in 10 % tolerances and are fairly stable with regard to temperature, and as a bonus, their price is fairly low as well. As such, they are often a fine choice for simple (not-too critical) filters, differentiators, and ac-coupling applications. With regard to filters, you should only use them in noncritical circuitry, such as tone controls and basic roll-off filters for noise sources. Do not use them for the filter element in serious VCF designs or the timing element in VCOs. Use polystyrene caps instead.

Polystyrene capacitors are the tightest tolerance and temperature stable type made. Consequently, their price is substantially greater. However, I'm still talking about prices of less than fifty cents each, so they are well worth the money for VCOs and VCFs where precision and stability are important. They generally come in 5 % tolerances, which is quite good for capacitors. You won't have to keep a big stock of these on hand, since most designs can be accomplished with a 1000 pF value.

All of the capacitors discussed thus far have been fairly low in value, typically between 10 pF and .47 mfd. When you need bigger sizes you will have to go to electrolytics. The common electrolytic (it looks like a little tin can) is best for decoupling, ADSR timing elements and power supplies. Decoupling and ADSR applications will need values in the range of 1 mfd.

to 100 mfd., whereas power supplies will need big ones like 1000 mfd. It's important to realize that electrolytics are not very accurate; sometimes their actual value can vary from the marked value by as much as 50 %!

Electrolytic capacitors are generally *polarized*, meaning that they have a definite plus lead as well as a minus lead. They must be installed in a circuit in the right way. (There is such a thing as a nonpolarized electrolytic, but they aren't commonly used in synthesizers.) Usually the plus or the minus side of the cap is marked right on the package, and the plus lead will often be longer than the minus.

Another factor to keep in mind is that electrolytics come in various voltage ratings. You must observe these ratings or you will see a miniature firecracker in operation! Suppose your circuit uses a 15V power supply. Then pick your electrolytics so that they are rated at 16V or more. And here's a curious fact related to this subject: suppose you stick an electrolytic rated at 35V into the 15V circuit just mentioned. The capacitor will tend to downgrade itself and will turn into a capacitor with a 15V rating! In general, electrolytics tend to conform to the circuit at hand, and this is why manufacturers will put synthesizers through a factory burn-in before attempting any tuning up of the circuits. You should follow this procedure too. And the fact that the electrolytics tend to downgrade to the voltage in the circuit is one good reason why it doesn't pay to recycle them for other projects. Suppose you pull the capacitor mentioned above out of the circuit and decide to use it in another project. The markings will say 35V but it's actually a 15V capacitor now! Expect the "firecracker syndrome" if you use this in a 35V circuit!

Tantalum capacitors are also polarized. They are most often used to stabilize a voltage regulator in a power supply but can be used in any application where an electrolytic is called for. But before you go out and buy them in quantity, you should be aware that they are quite expensive. Part of the expense has to do with tolerance; they have very tight tolerances and are fairly temperature stable (compared to an electrolytic). Also, they tend to have a better transient response, which is why they are often used to stabilize power supplies.

Selecting Resistors

Let's talk about resistors now. This component is fairly easy to choose since there isn't nearly the diversity here that there is with capacitors. Basically, resistors come as carbon composition, carbon film and metal film. Occasionally you run into a wire-wound resistor in power supplies, but other than that they aren't too common in synthesizers. Let's develop some rules of thumb to aid in the selection of resistors for music circuits.

At one time the only kind of resistor available was the carbon composition type. But nowadays they are almost a thing of the past, having been replaced by carbon film types. You're really not likely to see them much

anymore, but occasionally they show up as surplus or in bargain bags. They are all right to use in most common circuits, but if you have a choice go with carbon film types.

Carbon-film resistors are the most common and least expensive type on the market today. The price is always less than ten cents, and in fact is usually closer to a nickel. Since the price per unit drops substantially when they are purchased by the hundred, it pays to stock up with some common values. 100 ohms, 10k, 100k and 1M resistors in particular find their way into synthesizer circuits by the dozens, so save money and stock up.

Metal-film resistors are the most expensive and are fairly hard to find, at least at the hobbyist level. Fortunately, you will never have reason to use them in any of the circuits in this book. But one thing that you might want to keep in mind is that metal film resistors are quieter and less apt to pick up stray hum in audio circuits. For this reason they are the best choice for high-quality preamplifiers and other low-noise circuits.

Resistors come in standard values and in a variety of tolerances. Ten per cent tolerances were common in the early days, but today a 5 % tolerance is almost the industry standard. Since they are so readily available and the price so low, there is no excuse not to use 5 % whenever possible. Some circuits, like VCOs and VCFs won't work at their best with 10 % values, so stay away from them!

Occasionally you will need a 1 % value. These are most commonly found in filters and VCOs, usually in the control-voltage summing networks. You won't need a big stock of these; generally 100k values are used, and more rarely other values. You should keep your eyes open for surplus sales and pick up fifty or so, if the price is right. A pack of fifty should last for a lifetime.

A word should be said about the power requirements of resistors. Except for power supplies, which handle a great deal of current, all of the modules in an electronic music synthesizer are low-current devices. This means that you can use 1/4-watt resistors throughout. This not only saves space and money, but makes for very attractive circuit board layouts as well. Sometimes in power supplies you will need bigger resistors, but you can deal with this when the occasion arises.

Potentiometers are easy to choose for projects since there isn't much of a choice available for the experimenter. They have almost without fail 1/4-watt ratings, which is just fine for synthesizers. And electrically, most types are all the same. But there is a mechanical consideration that's of some importance, and that's whether it's a sealed unit or not. Sealed pots last much longer and are practically immune from noise and "scratch" problems but cost much more. Whenever possible try to use sealed potentiometers in the audio portions of your circuits. Unsealed or open types are more suited for control voltage attenuators, where you won't be hearing any "scratch" through the audio system.

Selecting Integrated Circuits for Synthesizers

In general you won't have much choice in the selection of integrated circuits for your projects. Either the IC in question does the required job, or it doesn't; it's as simple as that. But one area where there is a certain amount of latitude is in the selection of op amps.

Op amps (short for operational amplifier) are general-purpose amplifiers which may be configured in a number of ways. You can make integrators, differentiators, filters, preamplifiers, comparators and many more circuits using this versatile integrated circuit. Once there were only a few types of op amps available; today there are many hundreds! All op amps share a number of characteristics in common, such as differential inputs, high gain, feedback capabilities and so on. But when it comes to noise figures, input impedances and slew rates each type of op amp is distinct. Let's look at several of the most common op amps and see how to select the right one for a particular job.

The most common (and easy to use) op amp today is the 741. This chip has been the mainstay of the industry for some time now and is noted for its durability, stability and ease of use. Be that as it may, it does have a number of limitations which make it unsuitable for some applications in electronic music. Its noise figure, for one thing, makes it a poor choice for any audio applications. There are many other op amps which are lower in noise and they should be favored over the 741 for any preamplifier, filter or other audio circuit.

Another drawback is the 741's poor slew rate. As discussed earlier, in everyday language *slew rate* refers to the output's ability (or inability) to swing far enough and fast enough to duplicate the input signal. Poor slew rate means that the output can't keep up with the input and the result is a limited high-frequency response. The 741 is rather poor in this area, and this is another reason why it is a bad choice for any audio application.

Now before you decide that the 741 isn't good for anything, here's the bright side of things! It is a very good choice for control-voltage circuits, since poor slew rate poses no problem in this type of low-frequency application. Likewise, control circuits don't pass any audio signals, so noise presents no real problem. When you add on the fact that it is a very inexpensive amplifier (around twenty-five cents!) you can see that it's just fine for most control-voltage circuits.

There is one instance where it might not be quite right for a control-voltage circuit, however, and that is where heavy-duty buffering is required. When buffering a capacitor in an ADSR for example, you want the op amp to have a very large input impedance. Well, the 741 isn't too bad; typically the noninverting input has an impedance of around 250k. But this still isn't good enough for critical buffering situations. At times like this you should go with an FET input amplifier.

FET type op amps have extremely high input impedances, sometimes

as high as hundreds of megohms! They are the perfect op amp for ADSR capacitor buffers and many types of filter circuits. One good type is the LF351 made by National Semiconductor. This chip is easily available now and the price is under a dollar. Another choice is the TL071. This latter chip is very similar to the former; the main difference is that it is produced by Texas Instruments. Either chip will work well in synthesizer circuits.

When you need dual or quad FET op amps, you again have a choice of brand names. A good dual unit is the LF353, and the amplifiers in this eight-pin package are similar to the LF351 mentioned above. Or if you're using the Texas Instruments line, try the TL072. Sometimes you will need a quad op-amp package. A good choice here is the TL084 available in a fourteen-pin package. Whichever you choose, you will be glad to know that all of the above chips are available from a number of sources. Mail order houses are usually your best bet, although local dealers and Radio Shacks are starting to carry a number of the devices now.

One of the designer's perennial problems is the battle against noise. Audio circuits which are subject to hiss, hum and noise are no fun at all! Neither the performer nor the audience can tolerate much of this, so it behooves us to always keep a lookout for ways to reduce noise. The proper choice of an op amp goes a long way towards reducing noise. While the FET op amps mentioned above generally have good noise figures, for the impedance levels that we're interested in, there are better op amps. For example the 4739 dual op amp is an excellent choice. This chip is very stable, has a relatively good slew rate and a better than average noise figure. In fact, it is the perfect choice for just about any audio application within a synthesizer. And it's just as easy to use as a 741! This chip is made by Raytheon, EXAR and other manufacturers and is available from quite a few mail order houses. You will be using the chip quite often.

When it comes to quad op amps, the 4136 is a nice chip to use. It is similar to the 4739 (except that it's a quad op amp, of course), and has only a slightly higher noise figure. In terms of noise and slew rate, however, it's still infinitely better than the 741. You will be using a lot of these as well.

There are many other op amps kicking around, of course, but the ones mentioned above will cover most every electronic music purpose. So far the rules of thumb for selecting op amps have been based on input impedance, noise figure and slew rate. Figure 3-1 is a simple table which summarizes the various op amps discussed previously in terms of these parameters. If you wish to really drive home the importance of these parameters, obtain spec sheets for the chips and compare them. You will get a better feel for the diversity of op amps on the market today.

Selecting Jacks

There is really only one type of jack that makes any sense for use in a synthesizer and that is the standard 1/4-inch phone jack. These are sturdy,

	Single	Dual	Quad
Standard	741	1458	-
FET	LF351 TL071	LF353 TL072	TL084
Low noise		4739	4136

Fig. 3-1. Op-Amp selection guide.

easy to hook up and not too expensive. Almost all serious commercial equipment uses this size, so if you employ them as well, compatibility is assured. Their chief advantage over all other types of connectors is that they are extremely rugged.

Phone jacks come in a number of configurations. An open-circuit type is the simplest; it consists of a tip connection and a ground. A closed-circuit type is similar, but has an additional "switch" which maintains contact with the tip unless a plug has been inserted. Stereo types exist as well, as do more complicated switching types. In all cases, a number of suppliers stock the jacks and you should have no trouble in obtaining the kind you need.

OBTAINING PARTS

Obtaining parts for synthesizer construction projects is getting easier every day. Several years ago, it was almost impossible to get some of the simplest parts, but we're now entering the "golden" age of electronic music. As such, even some of the more bizarre ICs are becoming quite easy to find.

The key to rounding up all of the parts needed is to scrap the notion of "local purchasing." Your local merchants probably won't have the parts you want, and if they do the prices will be two to three times higher than normal. Get into the habit of buying through mail order. Even with today's postage rates, prices will be substantially lower and such businesses generally have a better stock.

Think in terms of quantity and you'll save lots of money. When you see an ad for 741s at ten cents each, send off for a couple of dozen. You will always need 741s and can't go wrong there. Likewise, load up on common value resistors, capacitors and potentiometers. And don't forget hardware; if you see good prices on switches, jacks, or nuts and bolts, load up while the going is good.

To maintain your stock in a sensible fashion, think about picking up a set of those little cabinets with twenty drawers. These are often available at local hardware or department stores. Set aside some of the drawers

for resistors, some for capacitors and others for ICs. In general, start thinking in terms of inventory control. Even on a small, home experimenter level, you will save time and money by planning ahead like this.

For many parts, like the ones mentioned above, you will have quite a few choices of what merchants to do business with. When you find a fast and reliable company, stick with it. Sometimes, however, you will see a super bargain elsewhere or perhaps find that the part is available from only one supplier. This is the case with some of the more exotic music chips.

The best way to become a smart shopper is to load up on catalogs. Send away for several dozen, and don't forget those "bingo" cards in hobbyist electronic magazines. You circle numbers corresponding to various manufacturers, and send in the bingo card. You will then be put on a number of mailing lists. This is one case where the more junk mail, the better! Appendix A lists a number of suppliers carrying parts of interest to the music experimenter.

SUMMARY

In this chapter you learned something about creating a master plan for the construction of a synthesizer—how to see the forest rather than the trees. The notion of design standards was introduced, and various impedances, signal levels and input/output structures were described. It was agreed that all input impedances should be 100k, all output impedances are to be 1k, audio signal levels should be 10V peak-to-peak, and control voltages are to swing between 0V and +5V. Also, trigger signals should be about 1 millisecond wide.

The selection of capacitors, resistors and op amps was described next. When it comes to op amps, the major criteria were found to be slew rate, noise and input impedance.

This chapter was concerned with electrical standards only. In the next chapter, various mechanical considerations will be considered. This includes the notions of packaging, preparation of front panels and special applications hardware.

Chapter 4

Planning the Synthesizer

In the last chapter I discussed various electrical design considerations. Equally important are the mechanical aspects of building a synthesizer. For example, is the unit to be portable, a studio model, prepatched, or what? And like all musical instruments, great care should go into creating an attractive synthesizer. There is no doubt that a pleasant-looking instrument has a subtle influence on the performer. For example, consider guitars. If you were to build a guitar along strictly functional lines, you could probably do it for under \$100, but be left with an awful-looking monstrosity. Most builders of guitars put a great deal of time into creating a work of art. Chrome fittings, pearl inlays, exotic woods—all add to the beauty of the guitar—and the price as well. But the musician who owns and uses it derives a special inspiration, if you will, from it. And so it should be with the synthesizer; a nice-looking instrument pays for itself many times over in inspiration!

The design of a good-looking unit doesn't happen by accident. Much time must be spent mentally going over the various relationships of the modules and possible uses for the instrument. For example, while in the planning stages you might ask yourself these questions. "Will I need to travel with the synthesizer, or will it sit in my home studio?" "Should the VCA module be closer to the VCO or should I put it off to the side?" "If I decide to go with a 'powder blue' motif for the front panels, will I be able to get more of the same colored paint five years from now?" "Should I use dry transfer letters for the legends or perhaps stencils?"

These and other questions should all run through your mind before you actually start building anything. It can't be emphasized enough: a good "game plan" is absolutely essential if you are to create and build a

professional-looking instrument.

This brings up a very important point. The major impetus and justification for this book is that you can build a better synthesizer than you can buy! This has been demonstrated time and time again by innumerable hobbyists, experimenters and musicians around the world. This being the case, you should strive to make the thing look as good as a store bought unit. As it turns out, with a little patience, and the right tools and forethought, this isn't as hard as it seems.

So ponder the mechanical side of things a bit. Use your creative faculties to visualize how the final unit should look. Then start haunting the hardware stores, drafting supply houses and stationary stores in search of the tools and paraphernalia needed to implement your design.

Most modular synthesizers are designed as a mainframe unit and the various modules are bolted into place and electrically connected in some fashion. Each module is built behind a small front panel. The front panel houses the various jacks and controls, and usually the circuit board for the module sits on small angles behind the panel. When a particular module is completed, it is slipped into the mainframe and secured in place. The beauty of this system is that the modules may be easily moved around, or completely exchanged for different ones if desired. If you need to troubleshoot something, you can pull the defective module out for this purpose. This modular approach makes the most sense for home-built equipment and so is the method employed throughout this book.

What size should the mainframe be and how big should the individual modules be? These are not trivial questions. Consider commercial equipment, for example. Moog, ARP, Serge, E-Mu and many other manufacturers produce modular equipment; what sizes do they use? It probably won't surprise you to know that they all use different sizes, and thus there is no mechanical compatibility between the equipment. (Since when have competing manufacturers of anything ever gotten together!) You are in no way compelled to use any of these various sizes, so let's use one that makes the most sense for home studio work: this is the standard rack mount.

RACK-MOUNT ENCLOSURES

A rack mount is a very common arrangement. Many industries and laboratories employ them to house scientific equipment. For example, phone companies, college science labs, recording studios, and many other firms use standard rack equipment. A rack mount can come in many different heights, all the way from small desk units up to ceiling height affairs. The height determines the number of modules which may be installed, and in this dimension they vary wildly. But in terms of width all rack mounts are the same. The width is such that 19-inch wide panels slip right into place. These panels can be bolted down using standard holes as well.

The point is that rack mounts are the most standardized mounting system available today. The width is always 19 inches and the bolt holes also

follow a standard pattern. As an extra incentive, used rack mounts frequently show up on the surplus market, often for ridiculously low prices. In fact, it is a rare surplus store that doesn't have some rack-mount equipment. So haunt the stores a bit and see what you can come up with. Figure 4-1 shows a commercial rack-mount enclosure.

Of course, you don't have to buy a rack mount. You can always build your own using 1-inch x 10-inch lumber. To make a nice job of it, you might consider covering the finished product with vinyl, using white glue and a staple gun. Attach some handles, metal corners and rubber feet and in one afternoon and you can have a homemade rack mount that will rival ready-built units! Figure 4-2 shows a home-made double-sided rack enclosure.

Several of the electronic music kit manufacturers are starting to offer modules in rack-mount size now. This, once again, adds to the versatility since the modules can slip into any standard rack-mount enclosure.

Before going on to consider the individual panels it should be mentioned that there is nothing sacred about rack mounts or indeed any type of mount. If you come up with some other arrangement that suits you, then you have accomplished all that needs to be accomplished. But for maximum versa-

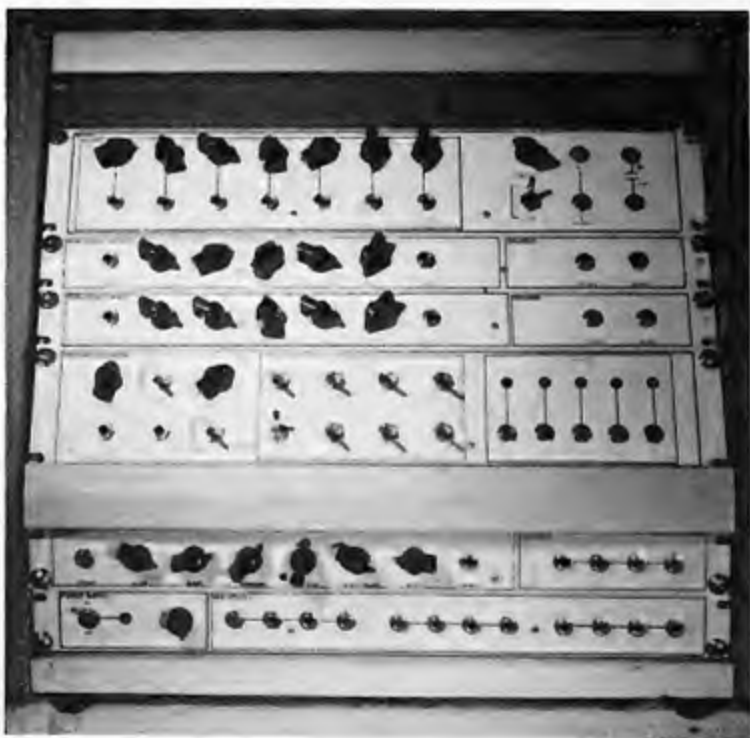


Fig. 4-1. A commercial rack enclosure.

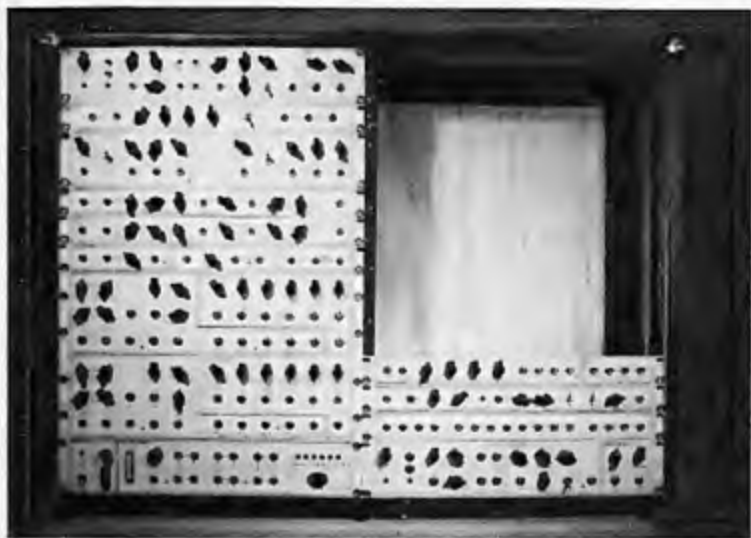


Fig. 4-2. A home-made rack enclosure.

tility with the option for future expansion, consider the rack enclosure carefully.

MAKING RACK PANELS

Fabricating attractive front panels for a synthesizer can be a real challenge at times, so it may be worthwhile to describe a system to simplify the procedure. You will note the word "system," for it is only by some systematic approach that consistent results can be expected.

It's important to set high standards of attractiveness when it comes to front panel design. Remember, you're not just building electronic circuits but are actively engaged in designing a musical instrument. An attractive instrument helps put you, as a musician, in a more creative mood. If you have ever seen a Stradivarius or an Amati, then you will appreciate that the esthetics of an instrument is not a trivial affair. And the sense of pride you will feel when someone comments on not only the fine sound but also the good appearance of your synthesizer is no small thing either!

What is going to be revealed here, then, is a method for taking ordinary blank rack panels and turning them into front panels for synthesizer modules. The end result should be quite attractive, and could easily be mistaken for a "professional" job.

Perhaps the first question to consider is where to obtain materials. Well, you will need a blank rack panel, and they are fairly easy to locate. Many electronics dealers (see Appendix A) stock these in a variety of sizes. The prices might alarm you though! For example, a blank, double-width panel goes for around \$5 to \$8! (All rack panels are 19 inches wide; the height

comes in multiples of 1 3/4 inch. A single width is 1 3/4 inch, a double-width is 3 1/2 inch, a triple-width is 5 1/4 inch, and so on.) Well, these are expensive times we live in, but that's a little steep even for the wealthy experimenter!

You should check the alternatives, since you won't want to play your home-brew synthesizer in the poor-house! And along these lines, here's an interesting fact. Local sheet metal houses will cut you the right-sized panels for anywhere from one-fifth to one-tenth the prices mentioned above. Typical prices will run about \$1 for single-width panels, \$2 for double width and so on. This is quite a saving!

Of course you will have to cut your own mounting holes, but for that kind of price break it's worth it. (It takes about ten minutes to cut the four notched mounting holes.) When you go to the sheet metal firm, be sure to tell them that you want 1/8-inch aluminum stock, 19 inches wide by whatever panel height you have in mind. And you will probably make the man happier if you order more than one panel at a time, since he can cut a big strip 19 inches wide, and then sever it down from there to the individual panels.

You may get a choice of quality stock, in which case go for the least expensive (i.e., the worst looking), since in a later step we will be removing all the surface blemishes anyway.

Now what about making those mounting slots? Figures 4-3 and 4-4 give the basic idea. In Fig. 4-3, four holes have been drilled or punched in the corners. Their diameters should be 1/4 inch. An easy way to locate their positions is to use a standard store-bought panel as a template. You can also take the measurements off the rack case or use some of the standard positions indicated in Fig. 4-5. (Figure 4-6 shows the standard spacing of holes for a rack case. The spacing is such that any two panels of any size may be positioned together.) Now, with a hack saw, slots are cut into the holes as shown in Fig. 4-4. Finally, a flat file and a rat-tailed file are used to smooth up the edges of the cuts.

After following the above steps, you will have a standard rack panel for a fraction of the normal cost. The only trouble now is that it probably looks pretty ugly, with lots of nicks and scrapes! Don't worry! Simply load an electric sander with some fine sandpaper and go to it. Don't be concerned about the sandpaper "design" being etched into the panel, since this will be covered later by some pain primer. But go at those nicks and scrapes, and after several minute's sanding you will have a smooth panel.

The next step is to buff the panel with some 00 gauge steel wool. This will do wonders for the surface and will give it a smooth and silky feel. After buffing, wash the whole panel with soap and water to remove any grit and steel wool particles. Wipe the panel dry with a lint free cloth.

Now is the time to put a temporary "cover" on the panel as an aid to the layout process. Using strips of masking tape, cover the entire front panel surface (decide which side you want to use for the front first). This step is easier if you use two-inch wide masking tape. The general idea is

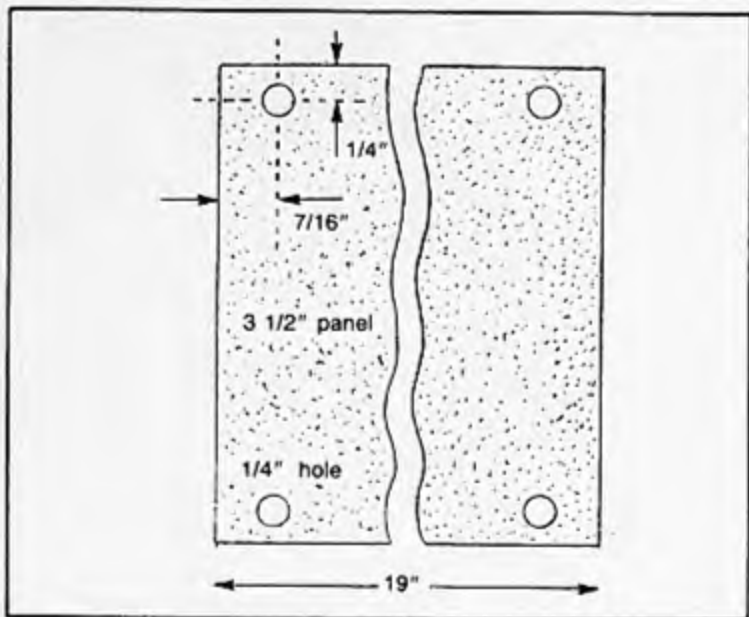


Fig. 4-3. The panel is drilled with 1/4" holes. Spacing of holes is always 1/4" from edge or 1 1/2" from edge.

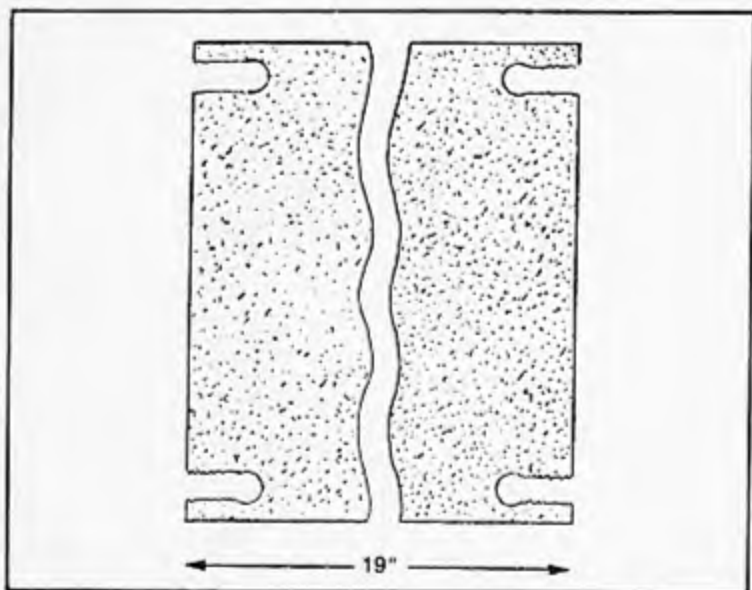


Fig. 4-4. The slots are rough-sawed out. Cut slightly inside and smooth edges with a flat file.

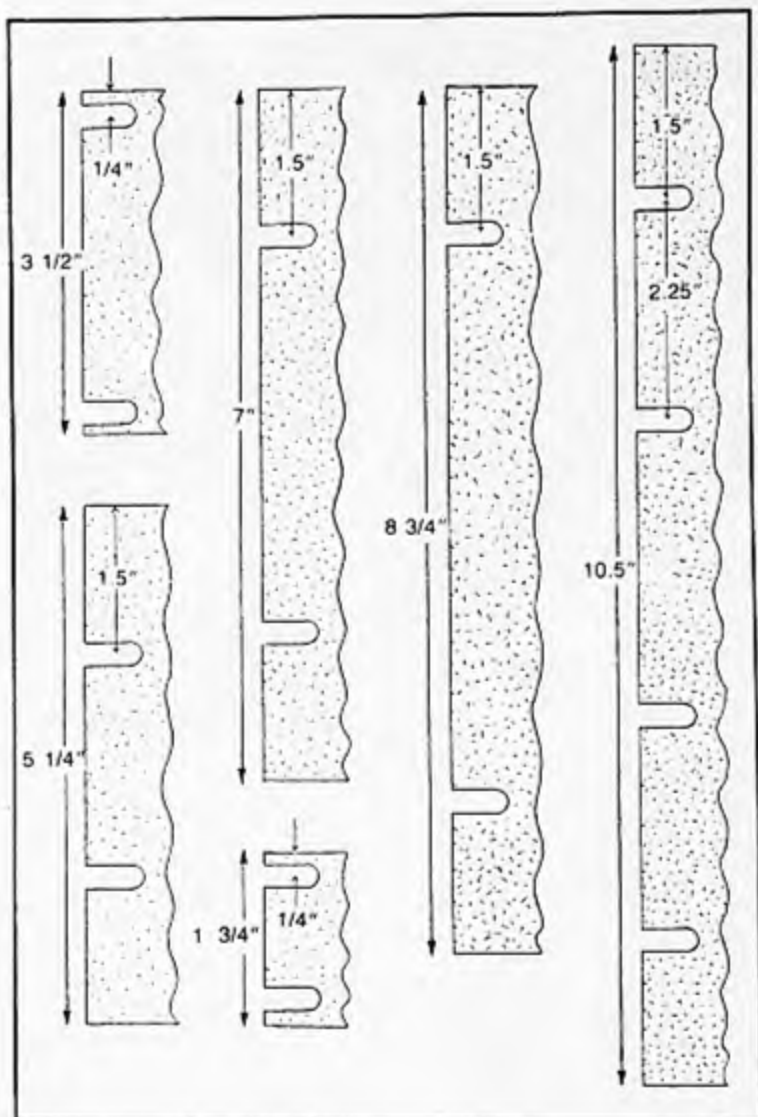


Fig. 4-5. Standard rack panel sizes and mounting holes.

to cover the front surface with a protective layer which will accept pencil markings.

Working from a life-sized mock-up prepared earlier (more about this later), mark the various spots on the panel (on the tape) where you will need to drill holes. Then with a metal punch and hammer, indent these various centers which you have just marked.

A portion of the left side of the panel mounting case showing a 3 1/2" panel mounted with a larger panel below. Note how the sequence of holes spaced at alternating 1 1/4" and 1/2" spacings allows for any sequence of panel sizes. The mounting case edge might be wood for a homemade case. Commercial racks are metal, and the holes shown are tapped with a 10-32 standard thread. If the case is wood, large flat-head wood or sheet-metal screws can be used to mount the panels.

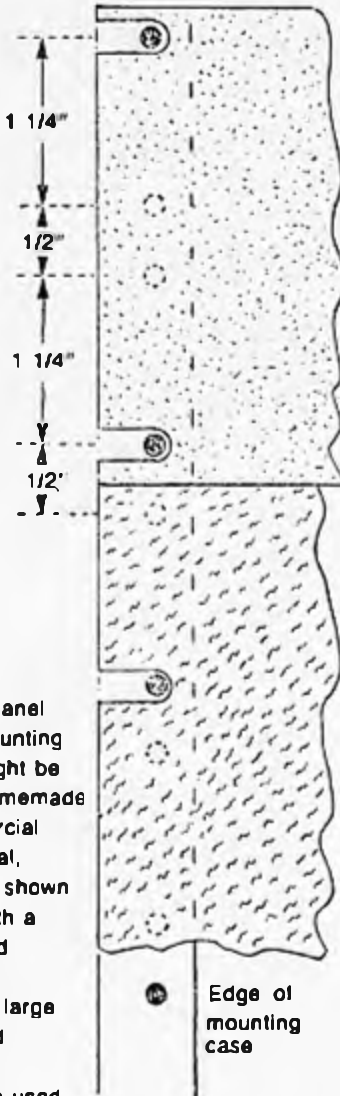


Fig. 4-6. Left side of the panel mounting case.

At this point you can drill or punch the holes in the front panel. If you use a drill be sure to make some pilot holes first, and in general be very careful since that metal can really cut you if the drill bit should snag and start the whole panel spinning around. The best approach is to clamp the panel to the workbench on top of a piece of scrap lumber to hold it firmly in place.

After the holes have been made you can remove the tape. You are almost ready to paint the panel, but first you should clean the surface with a cotton ball and some alcohol. The alcohol serves to clean off any soap film from an earlier step, and will also clean off any adhesive left from the masking tape.

Now paint the panel with some bare metal primer. Don't neglect the primer since it has the property of smoothing out or filing in any small abrasions. The end result will be very smooth and silky to the touch. When the primer is dry, paint the panel with any color spray paint that you like. And here's a tip concerning the selection of spray paint for the job. Do not use cut-rate, bargain paints. All too often they have inferior hardening compounds and simply don't dry correctly. Get a good appliance-type spray paint. Some hobbyists have even used auto spray paint to good advantage. If you have a choice, epoxy spray paints give an extra hard surface and are well worth the extra cost. Pick a light color so that dry transfer letters and other markings will show up on the surface. If you have access to white dry transfer letters, then black is a good color, since it tends to mask any dirt or smudges.

After letting the colored paint dry overnight, put the panel in an oven and "bake" it for about a half-hour at 250 degrees or so. This helps dry up any of the oils or solvents left in the paint and removes all of the tackiness. Be very careful not to contaminate any food in the kitchen and be sure to keep the windows open to vent any fumes emitted by the oven. Remember, be careful! You will find that this step really improves the durability of the panel finish, so give it a try if at all possible.

At this point you can label the panel. Dry-transfer letters give a very nice touch and are really quite easy to use. If you make a mistake, simply lay a piece of masking tape on the error, lift it up, and the letters will lift up with it. In the case of VCOs, you may want to actually draw the output waveforms below their respective jacks. Use a fine tipped marking pen for this. Such pens have indelible ink and are usually used for resist ink work (for printed circuit boards). In addition to the pen, use a standard schematic template to generate the various shapes. The use of the template and the special pen will really look good on your modules, and it's almost impossible to tell that the various symbols were created by hand. There are all sorts of tricks to create graphics and you will discover more as you go along. By the way, don't neglect straight lines either. Lines to separate various sections of the panel can really make the module easier to use. Think in terms of human engineering and you won't go wrong.

When you are done laying out the panel to your satisfaction, spray-

paint the surface with clear plastic or lacquer. Allow this to dry and spray again, and then one more time. After letting the panel dry overnight, bake it again for about a half-hour at 250 degrees or so. This final baking will darken the panel a shade or two, so plan ahead and use a lighter background color than you think you need. The purpose of all of this, of course, is to create a very hard protective surface on top of those relatively fragile dry transfers. Despite this protection, you should take extra care when mounting jacks, pots, and the other controls. A slip of a pliers or a wrench while tightening a nut can easily spoil all your hard work.

You are now down with the panel and it will be both durable and attractive. You will no doubt be surprised at how the panel no longer betrays its heritage; no one will ever know that it is homemade and started out as a piece of ordinary aluminum.

As mentioned above, you should lay out the panel from a mock-up. It is essential to prepare for this by doing a full-scale drawing of how you want the panel to appear. By doing it full scale, you can play around with the actual pots, jacks, and switches themselves to see how they fit together. You will soon find out that the mock-up is the single most important aspect of creating nice panels.

This, then, is a working method for making rack panels. It may seem like a lot of work, but you will find that if you have a clear idea of what's involved, you can actually go through many of the steps without much thinking! And here's a tip to make things go easier. It may sound a little crazy, but the night before building anything, as you lay down to sleep, visualize in your mind just what you will be doing the next morning. You'll find that this mental preparation makes the actual work go very quickly, since you are essentially going through the motions. After all, you've already built the thing (in your mind) so what's so hard about building it again?

CIRCUIT CONSTRUCTION METHODS

Besides rack panels, rack mounts and other mechanical considerations, the hobbyist must also consider electronic circuit construction techniques. There are a number of possibilities, such as printed circuit boards, perfboards, general purpose breadboards and so on. Each experimenter will have his or her own favorite technique. Some people can build large circuits very quickly with perfboard and flea clips, but not feel at home at all with printed circuit board methods. Another person might prefer breadboard methods. Let's look into these various techniques of circuit construction.

There is no doubt that printed circuit boards give the most reliable and consistent results of any method. They tend to look nicer and produce less hum, noise and stray capacitance. But that's only if they've been laid out correctly! A poorly designed printed circuit board can be a nightmare! So, if you're going to get into printed circuit boards, be sure to take your time and learn the ropes well. Many of the electronic hobbyist magazines

periodically explain circuit board techniques.

If you have access to the facilities, photographic circuit board construction is by far the best. In this method, you create some artwork for the board, using various patterns and tapes on clear plastic. You can arrange and rearrange the patterns as needed. To make it easy to see where you're going, the artwork should be designed on top of a "light box." A light box, as the name suggests, is a box containing a piece of frosted glass with several fluorescent lights inside. The lights shine up through the artwork and make it easy to detect any errors.

After the artwork has been prepared (this is termed *positize*), a negative is shot. This takes a special camera, capable of reproducing very large pages. Generally, you will have to go to a commercial printer to have this done and the price can vary from under \$1 to \$7 or more. If you are near a college or university, you will often find that reprographics departments will shoot the negative for a very reasonable fee.

After obtaining a negative of the artwork, you will then use it to duplicate the pattern on a copper-clad board covered with a layer of photosensitive plastic. The negative leaves a pattern on the plastic and after development, the board is immersed in a special acid solution. The acid etches away the unwanted copper, leaving the desired traces and patterns on the board.

With a drill press and a very fine bit, you will then drill the various holes for the ICs, capacitors, resistors and so on. After this, you must scrub the copper traces with a fine steel wool to remove any residual plastic and give the copper a bright shine. This aids in the soldering process later on.

Printed circuit board construction, besides giving attractive results, makes it very easy to create duplicates. For example, if you need four VCOs for your synthesizer, you can run off the required circuit boards quite easily. After all, the artwork is done; simply run off as many boards based on this artwork as needed.

Not everyone will have access to the facilities required by this method. For these experimenters a good alternative is to use breadboards. These are general-purpose circuit boards with a number of parallel traces, runs and buses, with a systematic grid of holes predrilled. You insert the various components, solder in place and add jumpers as needed. The buses on these boards make it quite easy to supply the bipolar 15V runs. Flea clips can also be of great value with this method.

Another possibility is plain old perfboard with flea clips. This method isn't seen as much now as it once was, but is still a viable way to get circuits up and running. The flea clips are used to secure the components to the board, while very fine wire (like the wire-wrap variety) serves to make the interconnections. Once you get the hang of it, this method can be very quick, and is perhaps the fastest technique of all. The main disadvantage is that the result is often quite ugly!

Whatever method you adopt, practice and get proficient at it. Don't be afraid to experiment and try out new ideas. But most of all, learn pa-



Fig. 4-7. The front of a completed module.

tience. A synthesizer is a large collection of circuits, so take your time and make each module a work of art. If you do so, you will be left with a musical instrument that you'll be very proud of!

ATTACHING CIRCUIT BOARDS TO THE PANEL

After making a front panel and a circuit board to go with it, you will want to figure out some way of connecting the two together. The best method is to attach the circuit board to the front panel with two small angles. These can be cut out of scrap aluminum stock and bent in a vise with a hammer and board. After forming the angles, drill the mounting holes. The choice of nuts and bolts is up to you, but #4 hardware seems to be the best.

After attaching the circuit board to the front panel, you may then complete the final wiring. Use stranded hookup wire, being sure to tin the ends of each piece before soldering in place. Also, consider using a variety of colors here; troubleshooting is made much easier later on, if you can chase the various wires down by following the colors.

Pay attention to wire routing. Don't make a jumbled mess of it; instead, lay out the wires in an orderly, sensible fashion. Use cable ties and clamps to secure bundles of wires, and try to follow the natural angles of the circuit board and panel. Be artistic! It's a well known fact that a neat attractive layout stands a better chance of working than a rat's nest. Figure 4-7 shows the front of a completed module, while Fig. 4-8 shows how the circuit board is attached and wired to the panel.

Here's a tip that many newcomers to electronics don't know, but old radio amateurs are very familiar with. To make twisted pairs or triples



Fig. 4-8. The back of a completed module

(or any amount of wires), cut the wires you need to the required size. Now grab one end of this bundle of wires, and fix it into your bench vise. Grab the other end and clamp it into the chuck of a drill. Do this just as if you were putting a bit in the drill, using the key. Now start the drill! Your wires will be twisted into neat "ropes." Such twisted pairs are very useful for making power supply runs to the circuit board.

One final tip concerning circuit board/rack panel work. Keep a good selection of #4 hardware on hand and use it for all of your panels. Having a standard appearance greatly improves the esthetics of the final synthesizer. You'll want to keep a selection of bolts, nuts and lock washers handy at all times.

PATCH-OVER HARDWARE

On the subject of mechanical design, you should not neglect the idea of "patch-over" hardware and how to implement it. Before you build your synthesizer, of course, you will want to have a very clear concept of just how you will configure your system. If you are planning to build a portable road synthesizer, you might want to consider a patch-over scheme.

First, what is patch-over hardware and what is it good for? A patch-over scheme is a means whereby you can have your synthesizer prepatched or normalized to some "standard" arrangement. For example, many synthesizer voices start with a VCO which is fed to a VCF and finally goes to a VCA. In a patch-over system this arrangement would be available automatically to the user without the use of any patch cords. What sets this arrangement apart from a strictly normalized scheme, however, is the ability to override the internal signal routing with outboard patch cords. A patch-over arrangement allows you to have the best of two worlds. Most of the time you won't have to use any cords to arrive at your final sound. But for those times when you come up with an "unusual" arrangement, you can override the internal patching and achieve any result available to a studio-type synthesizer.

Patch-over schemes are most suited for smaller (nonstudio type) units. They are especially valuable if you play in a band and expect to take the unit on the road with you.

To get our bearings, consider the design of a small system with three sound sources consisting of two VCOs and one noise source. Then you will want a lowpass VCF, one VCA, one ADSR, perhaps one AD (attack-decay),

one keyboard and some sort of keyboard interface, one Sample and Hold, one LFO and the power supply. A synthesizer of this size (minus the keyboard) could easily fit in a rack case 19 inches wide by 28 inches high.

Figure 4-9 shows a patch-over scheme suitable for this complement of modules. Consider the audio trail first. The basic arrangement is VCO to VCF to VCA. But notice that this trail can be broken, if desired, by inserting plugs into J10 or J8, allowing you to add additional audio processing if needed.

Since several of the modules are controlled by the keyboard, there are a number of 1-volt-per-octave lines as well. Both VCOs are controlled by this voltage as is the VCF. It is important to have this sort of input on the VCF to allow "tracking." This guarantees a constant waveform over the VCOs entire range. Note however that J4, J5 and J11 allow you to override the 1V/octave inputs. This is especially handy if you have two keyboards and want them each to control a VCO.

The keyboard outputs a gate and trigger signal as well. These signals are both needed to generate an ADSR waveform. If desired, J1 and J2 can be used to disable this normal arrangement and provide for external gating and triggering of the ADSR. This arrangement is commonly used to allow the LFO to trigger the ADSR for repeating envelope effects.

The AD unit also requires a trigger signal. Just like the ADSR, this trigger input can be overridden, only this time with J3.

The envelopes are commonly used to modulate the VCF and VCA. Note that a switch is associated with both of these modules (S2 and S1, respectively). These switches allow you to choose which envelope generator will control which module. If you don't want any envelope control, J7 and J6 allow you to substitute some other signal.

This, then, is a practical patch-over scheme. There are a few more tricks that can be employed, and I'll discuss them in a moment. First, however, it should be mentioned that Fig. 4-9 only shows the controls and inputs on the synthesizer which are affected by the patch-over scheme. Of course, the synthesizer has many more inputs, outputs and controls but they aren't shown on the diagram to keep things simple.

The other inputs and outputs just mentioned are less "standard" and no single patch-over scheme seems appropriate for them. As it turns out, with the system described in Fig. 4-9, you should be able to accomplish just about any patch with under ten cords!

An extremely useful option is a patch-over mixer, to be added onto one of the VCOs. Figure 4-10 shows the arrangement. If your VCO has a variety of outputs, such as sines, pulse waves, triangles, ramps, etc., then this is just the thing for you. Having lots of outputs is great, but the drawback is the increase of cords needed to accomplish a patch. But add the mixer to the VCO module, as shown in Fig. 4-10 and you can get around this!

Without any patch cords inserted, the mixer allows you to combine the VCO outputs to a single mixed output, which can then go on to the VCF. However, suppose you want to use the mixer for something else (like

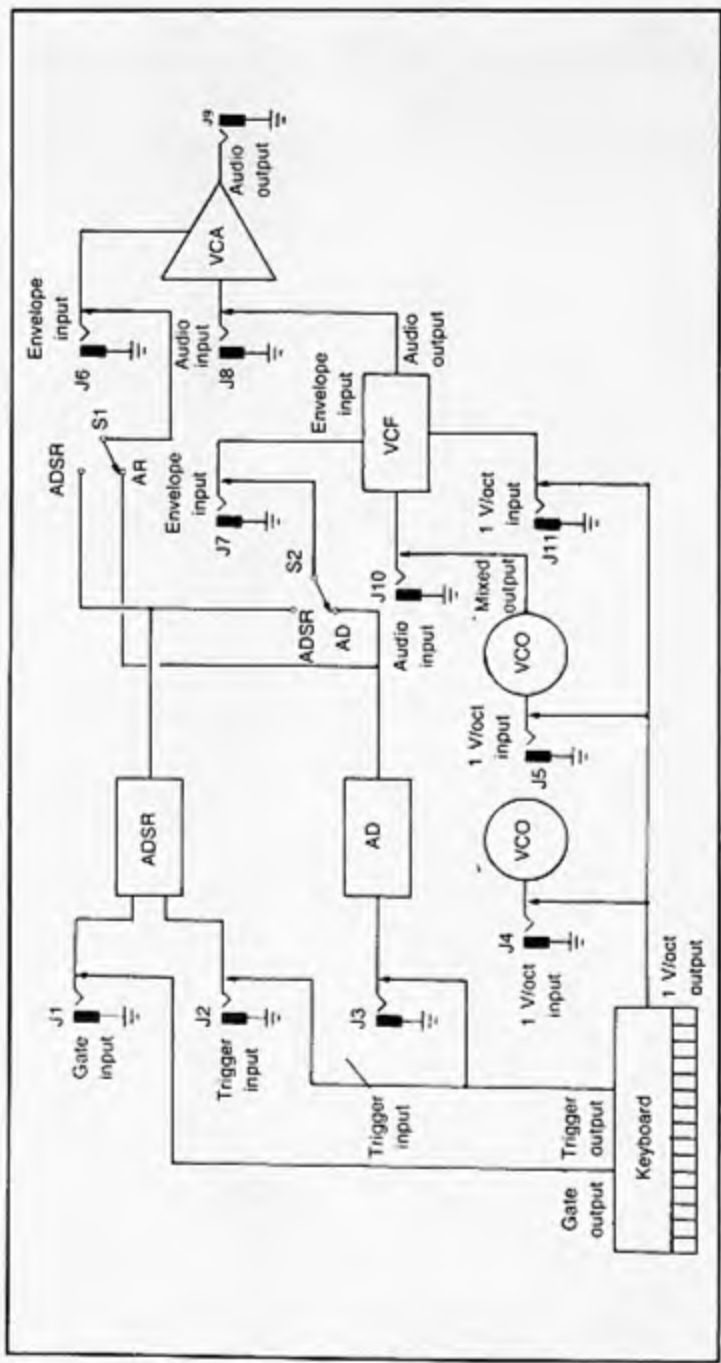


Fig. 4-9. Master patch-over scheme for a small synthesizer.

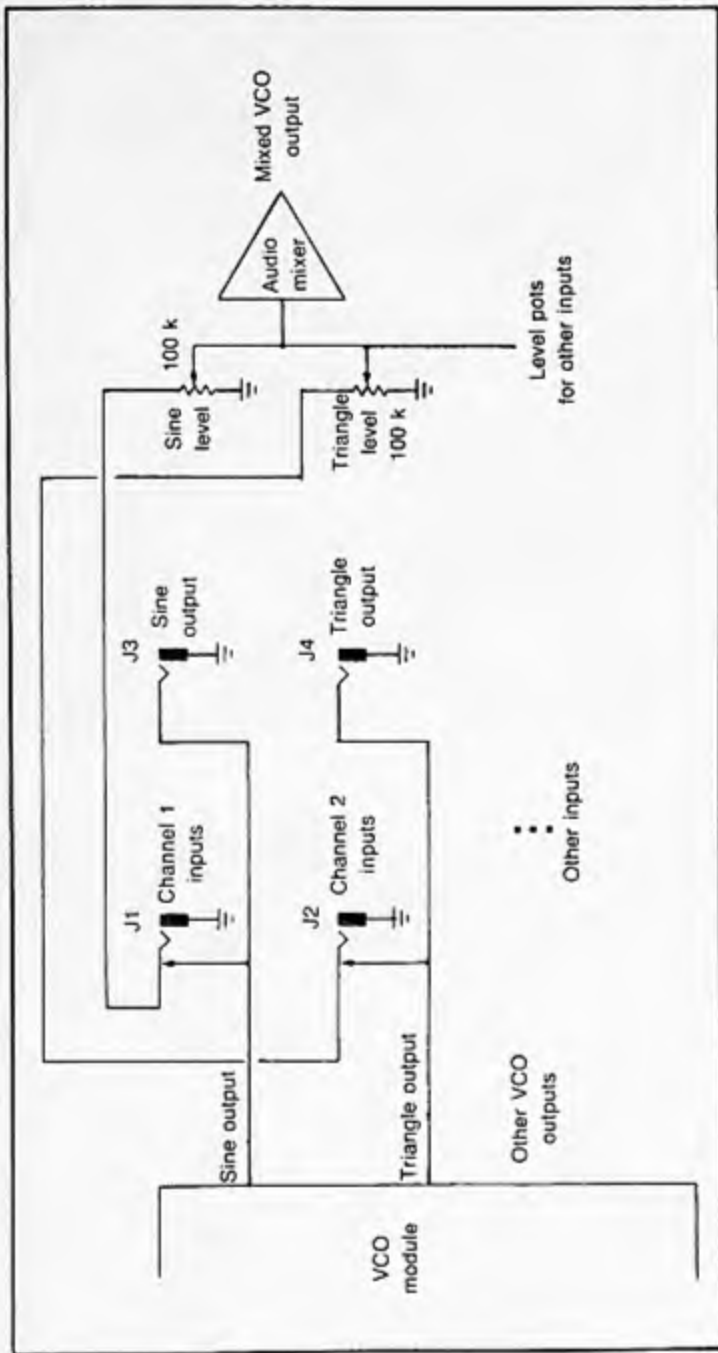


Fig. 4-10. Patch-over mixer for a VCO.

mixing some control signals). You can do that too! Simply insert plugs into the inputs (J1 and J2 in the figure), and away you go. Note that the VCO outputs are still available from the straight output jacks (J3 and J4 in the figure). If your VCO has five outputs, for example, you might want to go ahead and make the mixer a six-in, one-out arrangement and allow one of the inputs to remain "uncommitted." This is handy when you want to mix in some white noise or some other sound.

Figure 4-10 shows the mixer as essentially audio in nature, but actually you should implement it with dc coupling so that you can mix control signals as well if needed.

Figure 4-11 shows the finishing touches for a master patch-over scheme. This is a very simple arrangement, so not much need be said. Essentially, when using the Sample and Hold, you will find that you tend to sample the noise source under LFO control quite frequently. The scheme reflects this fact. Sometimes, however, it is fun to staple the VCO ramp wave output, in which case you can simply patch the VCO into J1. It's as simple as that!

This then is a complete patch-over scheme ready for use in a small portable synthesizer. Of course, you have the freedom to modify this in any way you see fit and can thus personalize it to your needs. If after looking at Figs. 4-9 through 4-11 you see something you want or don't want, get into the do-it-yourself spirit and make the changes. One tip, though: take your time and spend several days making rough sketches to arrive at a workable scheme. Mull it over, contemplate, meditate or whatever. Some arrangements just don't suggest themselves in one sitting!

SUMMARY

In this chapter I have discussed the practical construction and design of a home-built synthesizer. To standardize the arrangement somewhat, it was decided that a rack mount with rack panels is the best way to configure the system. A method for fabricating rack panels was described.

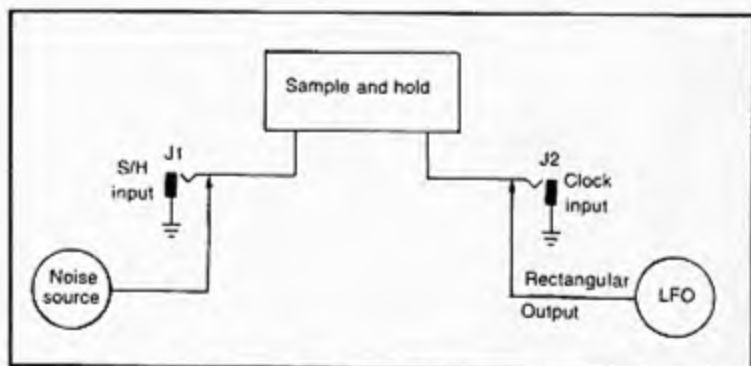


Fig. 4-11. Sample and hold patch-over scheme.

with suggestions for attaining attractive results. Then various construction methods for building the electronic circuitry were described. Printed circuit board construction gives the most reliable and consistent results, but breadboard or perfboard methods are adequate as well. Finally, a complete patch-over scheme was detailed, suitable for use in a portable synthesizer.

Chapters 1 through 4, while not a complete exposition of synthesizer principles, have covered all of the theory needed to start building. In the following chapters actual construction projects are described.





Part 2

Construction Projects

Chapter 5

Deluxe VCO

This chapter explains how a very high-quality, voltage-controlled oscillator is developed. This VCO has many features such as a variety of waveforms (including a sub-octave output), soft and hard sync, pulse-width modulation and a choice of linear or exponential FM inputs. And of course, it has the standard one-volt-per-octave exponential response with full temperature compensation, making the circuit suitable for the most demanding applications.

If you're familiar with how VCOs were built in the "old days," then this design might surprise you. As you'll see, it is accomplished with just three chips! One integrated circuit in particular makes this economy of parts possible, the CEM3340 (manufactured by Curtis Electromusic Specialties). This chip contains a complete VCO with various waveshapers, voltage regulator and exponential converter onboard and is specially designed for electronic music applications. The other parts in the circuit merely provide support for this chip. If you are unfamiliar with the CEM3340, be sure to obtain the application note and spec sheet and spend some time studying it. (See Appendix A for a list of suppliers.) Your time will be more than repaid by the ease with which high-quality circuits can be whipped up using the chip.

There are a number of design considerations in this circuit that pop up again and again, so to save space, Fig. 5-1 summarizes these common features in the form of a "spec sheet." These specs will make many facets of the design easier to understand.

HOW THE CIRCUIT WORKS

The following circuit operation and description is much simpler be-

Outputs: Sine, triangle, ramp, pulse and sub-octave pulse. All output impedances 1k, all signal levels 10V p-p, all outputs dc coupled. Pulse and sub-octave continuously mixable to a single output. Frequency range from 0.5 Hz to 35 kHz.

Inputs: All input impedances, except hard and soft sync, are greater than 50k.

Control voltages: 1 volt per octave, exponential FM, linear FM (ac coupled), coarse and fine tuning.

Sync inputs: Soft sync set for 10%. Hard sync switch selectable for reset on rising edges, falling edges or both edges.

Pulse width: manual or voltage controlled.

Fig. 5-1. Specifications for the deluxe VCO.

cause of the CEM3340, which does most of the hard work for us. (Refer to the schematic in Fig. 5-2.) J1 and J2 are the two exponential control-voltage inputs. The voltage applied to J1 can be attenuated by R46, which frequency modulates the VCO in an exponential manner. The voltage at J2 is left alone and supplies the 1-volt-per-octave signal needed for chromatic control of the VCO. As in Chapter 2, a modified organ-type keyboard typically provides the control voltage to this input, and the VCO thus controlled will follow the equally tempered scale.

J3 and its associated attenuator R47 form the input for linear frequency modulation. This would commonly be used for vibrato (FM by an LFO), or for creating gong sounds (FM by another VCO). The input is ac coupled, which makes it very easy to use, because you don't have to worry about any dc offsets shifting the fundamental pitch.

R44 and R45 are the coarse- and fine-tuning controls. R44 covers a very wide range; in the farthest counterclockwise position, oscillation is well below 0.5 Hz. At the other extreme of rotation, the oscillation is around 35 kHz. The fine-tuning control has a much more restricted range and covers a musical interval of about a fifth.

I have now covered the entire frequency control input structure comprising two tuning controls, the exponential FM input, the 1V/octave input, and the linear FM input. Now let's take a look at the sync inputs.

The soft-sync signal is injected at J8 and should be some sort of rectangular pulse. This sync input responds only to negative-going triggers. Resistor R34 is installed to limit the amount of syncing to about 10%. This is about right for most melodic applications since more syncing causes greater distortion and starts to sound like hard sync, while less sync gives inferior phase locking. However, feel free to experiment with R34 if desired.

The hard-sync signal is injected at J10. Now to understand switch S1 and its associated circuitry, you must be aware that the CEM3340 hard-sync input responds to either rising edges or falling edges. This being the case, it is easy to see that if the switch is in position 1, only positive sync pulses are transferred to the sync input at pin 6. Negative pulses are blocked by D1. Likewise, when the switch is in position 3, only negative pulses

are transmitted to pin 6. When the switch is in position 2, both positive and negative pulses are coupled to the sync input of the chip. The upshot is that we have three types of sync sounds.

The sounds available with this feature really are quite different with the falling edge position giving a more "throaty" growl and the both edges position giving a lighter, shriller sound. The rising edge position is really redundant since it sounds the same as the falling edge position (the outputs created by the two differ only in phase, and the human ear is relatively insensitive to phase differences). However, it's well to include the option anyway, because it would create different pictures if you were doing Lissajous experiments and it comes along for the price of a switch.

The pulse-width modulation input is at J9. In the absence of any plug being inserted, pot R48 manually controls the pulse width via a fixed voltage supplied by voltage divider R32 and R29. However, inserting a plug into J9 removes the fixed bias voltage and allows for voltage-controlled pulse-width modulation.

At this point we have covered all of the input considerations for the VCO; now it's time to talk about the outputs. Remember, the "standard" is to have 10V peak-to-peak outputs, centered about ground, with a 1k output impedance. With that in mind, let's look at the triangle wave output first.

The triangle is available at pin 10 of the CEM3340, however it is a nonstandard voltage. Op-amp A2 (which is one-quarter of a 4136) level shifts and amplifies this signal by a factor of two, making it standard. Note that the output impedance is made to be 1k by R6.

The triangle wave also feeds the sine converter formed by Q1 and Q2. If you're used to the old 3080 operational-transconductance-amplifier sine converters, this one will give you quite a surprise. Distortion is incredibly low, even with garden-variety transistors. What makes this converter superior to the 3080 type is the inclusion of feedback via R3, which practically nulls out the "pip" on the extreme end points of the sine wave. In the old 3080 type converters, this pip was quite audible as harmonic distortion. Trimmer R49 ("sine distortion trim") minimizes the odd harmonic distortion, while trimmer R35 ("sine symmetry trim") adjusts the even-order harmonic distortion.

The ramp wave makes its exit via J6. It should be obvious that A1 level shifts the ramp output at pin 8 of the CEM3340 just like A2 did for the triangle wave.

The pulse wave is available at pin 4 of the CEM3340. R26 is a pull-down resistor for the emitter of the internal transistor. The pulse developed across this resistor, then splits off in two directions. First it feeds pot R55, which is the Pulse/Sub-octave mix control. Then the pulse goes to the clock input of a CMOS 4013, configured as a binary divider. The output of the binary divider goes to a voltage divider comprising R8 and R19. This resistor string chops the sub-octave wave down to about +13V, which is the level of the original pulse. This means that both sides of R55, the mix control, see signals of the same amplitude, allowing smooth tran-

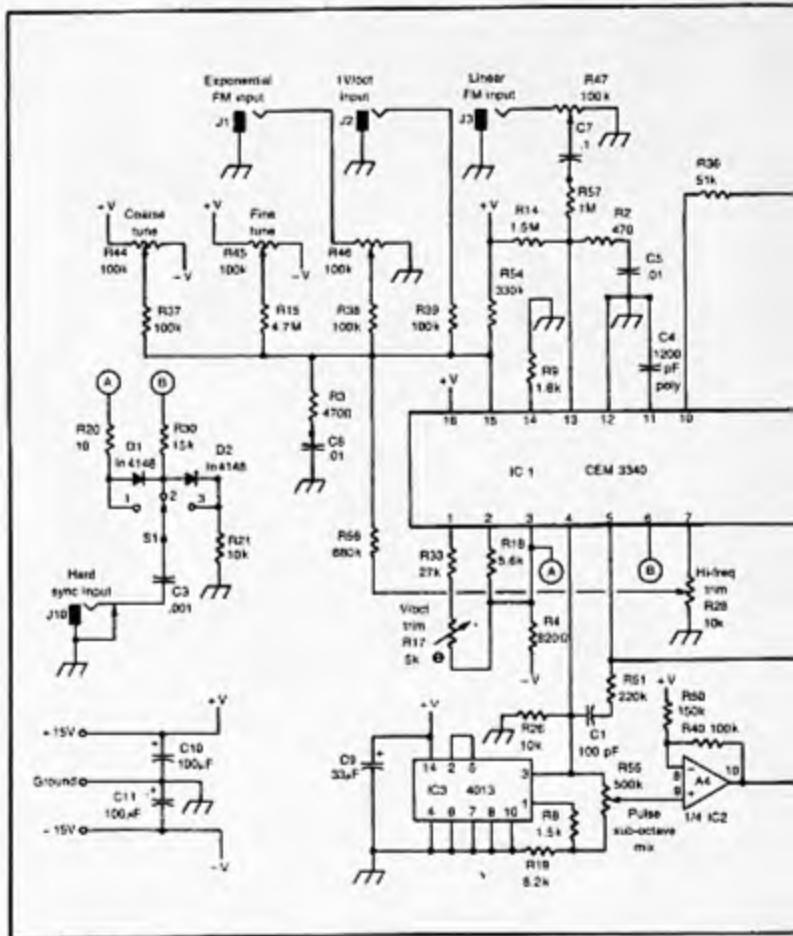
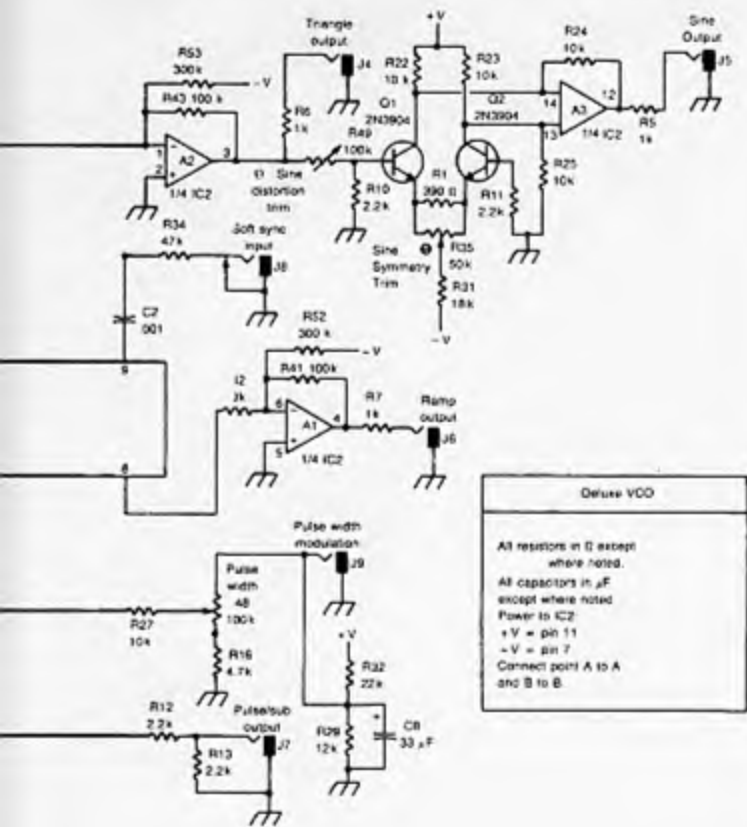


Fig. 5-2. Schematic diagram for deluxe VCO.

sitions. The wiper of R55 is buffered, amplified, and level shifted by A4, which is configured in a rather unusual way. If you imagine that R50 goes to ground, it is easy to see that A4 becomes just an ordinary noninverting amplifier. However, pulling R50 to the positive supply instead of ground level shifts the input signal and at the same time retains the noninverting amplifier characteristic. The result is an output that is 20V peak-to-peak centered about ground, which is then chopped down to 10V peak-to-peak by divider R12 and R13. This combination of 2.2k resistors gives approximately a 1k output impedance at the same time.

So, by utilizing this somewhat strange configuration, rotting R55 in one direction gives full pulse wave with no sub-octave. Rotating it in the other direction adds an increasing amount of sub-octave wave, until the



Default VCO

All resistors in Ω except where noted.
 All capacitors in μF except where noted
 Power to IC2
 +V = pin 11
 -V = pin 7
 Connect point A to A and B to B.

pot has been rotated all the way, at which point the output is full sub-octave. And the 10V peak-to-peak signal is maintained the whole way around the rotation!

By the way, R51 and C1 form a positive feedback loop for the internal pulse generator of the CEM3340. This adds hysteresis to the waveform and helps square up the pulse-wave output. The result is a very crisp and clean pulse wave, suitable for driving IC3, the 4013. CMOS chips, of course, require a very clean clock signal.

CONSTRUCTION

Construction of the VCO is best accomplished using printed circuit

board techniques, although other methods may be used as long as "rat's nest" wiring is avoided. Exponential converters are rather sensitive devices and a mass of tangled wires coming and going from the exponential converter pins of this chip can often create stray capacitance problems. Other than this precaution, there is nothing critical about the construction of the VCO.

Figure 5-3 shows the complete parts list for this project. You will have to make a special order for the CEM3340 from PAIA Electronics, but all of the other parts are available from a number of sources. (See Appendix A.)

C4, which is the tuning capacitor for the VCO, should be polystyrene for maximum stability. Recall that the VCO is the single most important module in the synthesizer and that the human hear is very sensitive to fluctu-

Resistors		Capacitors	
R1	390	C1	100 pF disc or polystyrene
R2, R3	470	C2, C3	.001 mfd. mylar
R4	820	C4	1200 pF polystyrene
R5-R7	1k	C5, C6	.01 mfd. mylar
R8	1.5k	C7	.1 mfd. mylar
R9	1.8k	C8, C9	33 mfd. electrolytic
R10-R13	2.2k	C10, C11	100 mfd. electrolytic
R16	4.7k		
R17	5k trimmer		
R18	5.6k	Semiconductors	
R19	8.2k	IC1	CEM3340 VCO chip
R20-R27	10k	IC2	4136 quad op amp
R28	10k trimmer	IC3	4013 dual flip-flop
R29	12k	D1, D2	1N4148 diode
R30	15k	Q1, Q2	2N3904 npn transistor
R31	18k		
R32	22k	Miscellaneous	
R33	27k	J1-J7	open circuit phone jack
R34	47k	J8-J10	closed circuit phone jack
R35	50k trimmer	S1	3-position rotary switch
R36	51k		
R37-R43	100k		
R44-R48	100k potentiometer		
R49	100k trimmer		
R50	150k		
R51	220k		
R52, R53	300k		
R54	330k		
R55	500k potentiometer		
R56	680k		
R57	1M		
R14	1.5M		
R15	4.7M		

Fig. 5-3. Parts list for deluxe VCO.

tuations in pitch. It follows that the tuning element (C4) should be as stable as possible and polystyrene is ideal for this purpose. And in general, use the highest quality parts throughout the construction of this project.

While not strictly necessary, R17 (the volts-per-octave trim) could be a multiturn trim pot. This eases the task of tuning somewhat and allows for very precise adjustment. However, for most musical applications this isn't necessary. Just use a good quality trim pot and you won't go wrong.

IC3 is a CMOS device so all of the usual static precautions should be observed. Use a socket for the chip. Actually, you should use sockets for all of the integrated circuits since this eases the task of troubleshooting if needed.

This is a very fancy VCO and it has many features that even the best commercial units don't have, so take your time and do a good job. Make an attractive panel, and if possible apply graphic techniques. For example, below the output jacks, draw pictures of the various waveforms (sine, triangle, pulse, etc.). Pictures are always easier to interpret than words. Besides making the panel attractive, take pride in your wiring and use tie-bands, clamps or whatever it takes to create a pleasant looking layout. Chances are the circuit will work like a champ right off the bat if you take your time and do things in a neat and orderly fashion.

CALIBRATION

Tweaking the module isn't all that hard, but it does take patience. Let's start by tuning the VCO for a one-volt-per-octave response. There are a number of ways to accomplish this, and each way requires varying amounts of test equipment. For example, you could tune it with a function generator and oscilloscope using Lissajous figures, or you could use a frequency counter and digital voltmeter. However, the simplest (and in many ways the best) way is to simply tune it with your own ears. After all, if you can't hear that anything's wrong, then nothing is wrong! Here goes.

Plug the keyboard control voltage into the one-volt-per-octave input at J2. Then hook up an amplifier and speaker to the triangle wave output and adjust R4 and R45 until you hear a tone. Now repeatedly play a middle C and one C above this, back and forth. Adjust R17 until you hear an exact one-octave interval. That's all there is to it!

Now try out all of the keys; do you hear a good equally tempered scale up and down the whole keyboard? If you do, then you can skip doing anything with R28 (and this will probably be the case). If the high end sounds a little flat, adjust R28 somewhat. The purpose of this trim is to boost the frequency of the very high range of the VCO, since it is in this range that the so-called reset time for the timing capacitor, C4, becomes of greater concern. After adjusting the trim pots, dab a little fingernail polish on to hold them in place.

The sine trimmers, R49 and R35, are best adjusted with an oscilloscope, with your ears serving as the final arbiter. While watching the wave-

form, tweak R49 until a nicely rounded sinusoid is formed. Then adjust R35 for a symmetric waveform. While performing both of these trims, listen to the sine wave on an amplifier and speaker (at around 500 Hz), and note the relationship between waveshape and timbre. Believe it or not, your ears will allow you to tweak up the sine wave for very low distortion. However, if you have a distortion analyzer . . .

USING THE VCO

This is a complicated circuit with lots of options, so really take your time in getting familiar with it. Learn to make the most of its features and you'll soon have many new and unusual sounds underway.

There are quite a few controls, so if you're having trouble when you first use the unit, make certain that you've accounted for the effects of all of the knobs. For example, if you're not hearing any sound at all, check to see that R44, the coarse tune control, is in the audible range. Recall that this VCO oscillates all the way from the sub-audible to the supersonic so be sure that you've got it in a range you can hear. Another case where you might not hear anything is when the pulse width control is set such that the duty cycle is 0 %. A zero percent duty cycle means no frequency is present! Adjust the control slightly and you should hear some sound begin.

Try out all of the outputs and confirm that they work. Then listen critically to the various timbres that these waveforms create. Notice how the sine wave is very pure while the pulse wave is harsh. Then compare the triangle to the ramp and note how the ramp contains even harmonics while the triangle wave doesn't.

If you have built two of these VCOs try out the sync inputs. Soft sync is used to lock two VCOs together in some sort of offset and is most often used for harmony effects. No distortion should be apparent. Hard sync is a little more difficult to use in that you must adjust the coarse and fine tune controls very carefully (of both VCOs) to achieve the best effect. Experiment!

Finally, give the vibrato a try. You will probably want to use the linear FM input for this, but feel free to try the exponential input as well. Note how the linear input limits you to a maximum of about one octave deviation, while the exponential input allows for around a dozen octaves.

These are just suggestions to get you started. You will find this to be a very versatile VCO, but to really get the best sounds from it, be sure to set aside enough time to practice. Just like any musical instrument, you must be prepared to commit a fair amount of time to rehearsal.

Chapter 6

A Four-Pole Lowpass VCF

Voltage-controlled filters come in all sorts of varieties. There are lowpass, highpass, bandpass, band reject, notch, allpass, two pole, four pole, polygonal and phase; the list seems to go on forever! But for your first filter, it is best to stick with an old standby, the four-pole lowpass filter. This type of filter, aside from being very easy to use, approximates the sounds of many natural instruments, and for this reason makes an excellent choice for a first filter.

In the past, constructing such a filter was a little messy. For a four-pole unit, you needed to build four separate gain control stages and this tended to add to the cost and complexity of the circuit. Through the magic of LSI (large-scale integration), it is now possible to build a lowpass filter with just a handful of parts, and the equality will actually be much better than that attainable with a discrete design.

The heart of this whole circuit is the SSM2040 integrated circuit. This chip is manufactured by Solid State Micro Technology and can be obtained from several sources including PGS Electronics (see Appendix A). Essentially this IC contains four voltage-controlled amplifiers, all sharing a common control stage. The response is already exponential; all you have to do is add a little support circuitry and the unit will be ready to go!

Figure 6-1 shows the specs and some of the features of this unit. Note especially that this VCF has a voltage-controlled resonance input, a feature that many commercial units don't even have. And as another special bonus, the unit can also be used as a voltage-controlled sine-wave oscillator with a very pure output waveform.

Features of the four-pole lowpass VCF

- Wide sweep range, typical ten octaves.
- Voltage-controlled resonance.
- All input impedances are greater than 50k.
- Output impedance is 1k.
- 10V p-p signal levels.
- Fully temperature compensated.
- Four-pole response.
- Able to oscillate in a pure sine wave.
- Standard power supply voltages of plus and minus 15V.

Fig. 6-1. Features of the four-pole lowpass VCF.

HOW THE CIRCUIT WORKS

Refer to the schematic in Fig. 6-2. C3 through C6 are the tuning capacitors. These capacitors should be polystyrene for maximum temperature stability; luckily, this value (.001 mfd or 1000 pF) is very common in polystyrene. Resistors R10 through R18 round out the RC network and should be 5 % tolerance units for best matching. These capacitors and resistors then, are the components which set the basic tuning of the circuit. With the values shown, the filter will be in a good range for audio synthesizer work. Resistors R1 through R4 have nothing to do with the tuning; it is their duty to attenuate the incoming signal to meet the operating requirements of the SSM2040. With the value as shown (200 ohms), distortion is minimized.

The audio input is applied to J3. This jack is fed to an attenuator, R39, and this allows you to reduce the amplitude of any incoming signals. A feature such as this is often needed when applying a mixer output to the VCF. A mixed signal, composed of several waveforms could easily exceed the filter's nominal 10V peak-to-peak input. Amplifier A1 and its associated circuitry attenuate the input by a factor of ten. This must be done since the SSM2040 likes to see signal swings of 1V peak-to-peak.

Since the signal is attenuated before entering the filter chip it's clear that the output must be boosted to compensate. Amplifier A3 and its associated components see to this function. Note also capacitors C1 and C2. These are selected to roll off the high end response (up around 100 kHz) to reduce the chance of any undesired radio frequency interference and, more importantly, spurious oscillation.

Part of the output is fed back to the input via IC4, a 3080 operational transconductance amplifier (OTA). This provides the voltage-controlled resonance function. The 3080 is set up as a standard voltage-controlled gain stage. R34 and R6 chop the input signal down to allow the amp to work in its linear region. Trimmer R42 is adjusted to minimize any "thumps"

caused by a dc offset being fed through the OTA.

A4 is configured as a current-to-voltage converter. This is needed since the OTA is a current output type device. Trimmer R20 is adjusted so that a maximum voltage generated by R41 (the initial resonance control) just sets the filter oscillating. By setting this trimmer so that the filter just begins to oscillate, clipping can be avoided and the net result is a very pure sine wave output.

Amplifier A5 forms the control voltage input stage. A5 and Q1 actually perform a voltage-to-current conversion. Since pin 5 of IC4 expects to see a control current, this conversion is necessary. D1 is in the circuit simply to protect Q1 from reverse voltage conditions. This insures that the circuit is "goof-proof." The actual voltage control input is at J4, and the voltage thus applied may be attenuated by R40. R41 allows a manual adjustment of the resonance.

Amplifier A2 is set up as a summing amplifier and provides the frequency control for the circuit. J1 is the 1-volt-per-octave input. This input would normally be used for a tracking keyboard control. The 1-volt-per-octave response can be tweaked for extreme accuracy by R22.

R36 and R37 are the coarse- and fine-tuning controls. The coarse-tuning control covers a range of about twelve octaves, while the fine-tuning control covers a range of less than one octave. These controls are especially handy when using the filter in the oscillation mode.

J2 is the envelope input jack. Since envelope signals are usually +5V or less, this input has been given a gain of three. Thus, a +5V input can cause the filter to sweep across its entire range.

C7 is put into the feedback loop of A2, and in this configuration helps minimize "pops" caused by modulating the filter's cutoff frequency rapidly.

To provide total temperature compensation, R8, which is actually a thermistor, is included in the circuit. This component will automatically compensate for any temperature drift. The result is a filter which will stay in tune, no matter what the playing environment is like. Since this filter may be used like a VCO, (by setting it oscillating), temperature compensation is definitely a must.

Rounding out the circuit, power is provided by a bipolar 15V supply. The supply is decoupled by R46-R47 and C8-C9.

CONSTRUCTION

Figure 6-3 shows the complete parts list for this project. You will have to make a special order for the SSM2040 and thermistor R8 since these are fairly unusual parts. (PGS Electronics is one good source; see Appendix A.) All of the other parts are common and may be obtained from just about any mail-order electronics supplier. Use fresh, good quality parts for this project since it is a "tuned" circuit. Avoid surplus parts if at all possible.

Building the four-pole lowpass filter is not difficult, but all of the usual

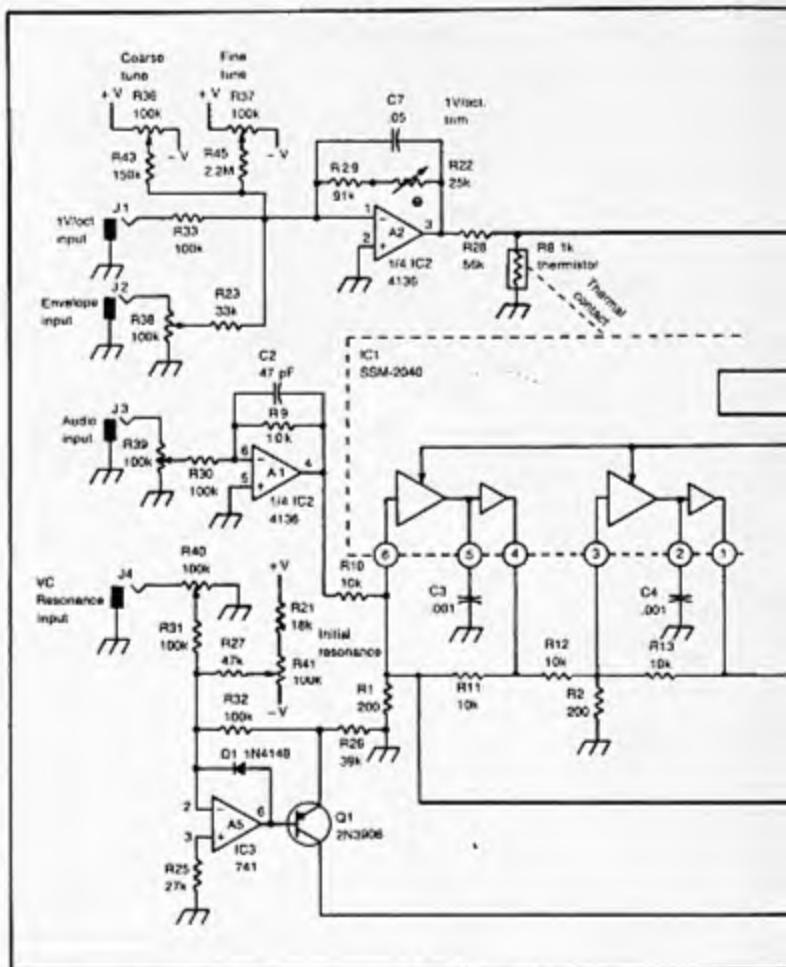


Fig. 6-2. Schematic diagram for four-pole lowpass VCF.

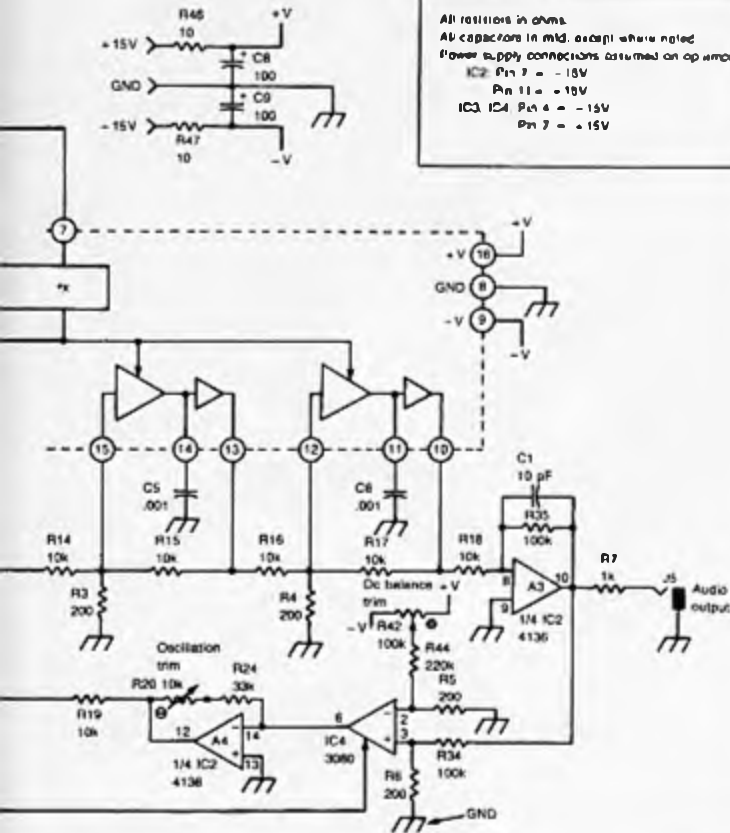
cautions concerning neatness in wiring apply. A printed circuit board is the best method of construction, with perfboard and wire-wrap being okay as well. Whatever you do, though, keep the wires neat and straight! And if the circuit board is more than several inches away from the front panel, use shielded cable to connect the audio jacks and controls to the board. Ground only one side of the shield in the cable and this will keep possible ground loops under control.

Even though the construction of this project is quite straight-forward, here are some tips to keep you on the right track. Old pro's will be familiar with most of these, but beginners ought to pay particular attention.

When installing the four IC sockets, a piece of masking tape can be

Four-pole 120-pole VCF

All resistors in ohms.
 All capacitors in mfd, except where noted.
 Power supply connections assumed on op amps.
 IC2: Pin 7 = -15V
 Pin 11 = 10V
 IC3, IC4: Pin 4 = -15V
 Pin 7 = +15V



used to hold them in place while soldering. Tape the socket down to the parts side, then flip the board over and proceed to solder. Watch for excessive heat which could lift pads, and beware of solder bridges.

Observe the orientation of all polarized components. These include the electrolytic capacitors, the diode and the transistor. Electrolytics are usually marked with a plus or a minus right on the package and the plus lead is generally longer than the minus lead. For diodes, a black band generally marks the cathode. For the transistor, with the flat side facing you, pins sticking downward, the orientation is usually from left to right: emitter, base, collector.

Here is the correct way to mount R8, the thermistor. First slip IC1

Resistors		Capacitors	
R46, R47	10	C1	10 pF disk
R1-R6	200	C2	47 pF disk
R7	1k	C3-C6	.001 mfd. poly
R8	1k thermistor	C7	.05 mfd. mylar
R9-R19	10k	C8-C9	100 mfd. electrolytic
R20	10k trimmer		
R21	18k		
R22	25k trimmer	Semiconductors	
R23-R24	33k	IC1	SSM2040 VCF chip
R25	27k	IC2	4136 quad op amp
R26	39k	IC3	741 op amp
R27	47k	IC4	3080 OTA
R28	56k	D1	1N4148 or 1N914
R29	91k	Q1	2N3906 pnp transistor
R30-R35	100k		
R36-R41	100k potentiometer	Miscellaneous	
R42	100k trimmer	J1-J5	.open circuit phone jacks
R43	150k		
R44	220k		
R45	2.2M		

Fig. 6-3. Parts list for the four-pole lowpass VCF.

into its socket. Now mount R8 right on top of IC1 and solder in place. If desired you may apply some white silicone heatsink grease between the IC and the thermistor. This is optional, but its use greatly increases the thermal bond between the two parts.

Now double-check your work. Example the circuit board carefully. Remember, the SSM2040 is *not* short-circuit proof and the chip could be destroyed very easily by a random solder bridge. So look everything over. Are there any inferior solder joints, solder bridges or shorted connection wires? Check the wiring to the panel; is everything right?

If your answer is "yes" to the above, then install the rest of the ICs in their respective sockets, and complete the front panel wiring. Incidentally, the controls and jacks easily fit behind a standard 1 3/4-inch by 19-inch rack panel.

CALIBRATION

First calibrate R42, the dc balance trimmer. With a dc voltmeter, measure the voltage on the wiper of R41 and adjust it for 0V. Now turn down the audio input, R39. Apply a 10V peak-to-peak, 2 or 3 Hz signal to the voltage-controlled resonance input at J4 and turn up R40 all of the way.

You may use a function generator, VCO or an LFO for this function. The frequency is not critical. You merely want to create some sort of alternating signal at the input to the voltage-controlled resonance stage.

While monitoring the dc voltage at pin 4 of IC2. Adjust R42 for the least deflection of the meter. This minimizes any "thumps" or "pops" in the audio portion of the resonance circuitry.

Next you tweak up the oscillation trimmer, R20. Start out by turning down the audio input control, R39. Next, patch the output of the VCF to an amplifier/speaker combination set to a comfortable listening level. Now turn down R40, the voltage-controlled resonance input, and turn R41, the manual resonance control, up full (positive).

Adjust R20 until oscillation just begins to occur. It may be necessary to turn the coarse tune pot, R36, up or down a bit to get the filter oscillating in the range of human hearing. If R20 is advanced too far, the unit will still oscillate; however, the waveform will not be a "pure" sine wave and may start to clip. So be sure to adjust R20 to the point of oscillation and no further.

Now you are going to "tune" the filter for a one-volt-per-octave response. Essentially you first start the filter oscillating and then tune it just like any VCO. Turn down the audio input attenuator, R39 and turn up R41 to start the filter oscillating. Connect a keyboard to the 1-volt-per-octave input at J1. Adjust the coarse and fine tuning controls to put the oscillation in mid-range (500 Hz to 1000 Hz).

Now while playing a C and an octave above that C, adjust R22 until a one octave span is heard. Keep alternating between the two C's and listen for the octave interval. There are other ways to adjust R22 using test equipment such as oscilloscopes, function generators, etc., but the method above is more than accurate enough for any reasonable musical application. However, for special purposes, purists may wish to refer to a method detailed by Bernie Hutchins in *Electronotes* #75, page 5. (See Appendix B.)

TROUBLESHOOTING

Experience has shown that malfunctioning circuits are almost never due to IC problems. Human error is more often than not the culprit. If you are having trouble, here are some tips to help you along.

Get a handle on the problem. Write down all of the symptoms. You can't fix something unless you really know what's wrong. Then apply logic. Starting from the symptoms, narrow down the problem to the logical function. Do not start looking for details right away; look for the general area. Example: no oscillation at any setting—problem must be in the resonance stage.

Now start looking for the dubious component. First check all solder joints. If necessary, remelt the solder and try again. Then look for any suspicious physical damage to components. Finally, check wires going to and from the panel.

In a circuit of this nature (lots of pots) be sure something's really wrong before tearing into it. Remember, some combinations of controls may yield odd effects. Despite the number of controls and jacks, the circuit is fairly simple, so the only test instruments you really need are your eyes, an insulated probe (such as a pencil) and possibly a multimeter.

Above all, remember that there is no such thing as magic in electronics. Any problem must follow logically from a cause. This is an immutable rule!

USING THE VCF

You will find that this filter is exceptionally easy to use. Since there are quite a few controls and jacks available to the user, many different patches are possible. To get you started, here is how to do a standard "wah-wah" synthesizer patch.

First consider the audio path. Patch the output of a VCO to the input of the VCF (J3) and then take the output of the VCF (J5) and connect this to the input of your VCA. If your VCO signal is 10V peak-to-peak or less, you may open attenuator R39 all the way.

Connect the keyboard gate and trigger to an ADSR. Now patch the output of the ADSR to the VCA and J2 of the VCF. Depending on the strength of your ADSR, you may wish to fiddle with attenuator R38 a bit. The keyboard control voltage output should go both to the VCO and also to J1, the 1-volt-per-octave input of the VCF.

Now set the resonance to a minimum via R41. Since this pot is strung between the positive and negative suppliers, the minimum position is actually at about mid-rotation (if you have used a linear taper pot).

Now play some music! If you have accomplished the patch correctly, you should be hearing a traditional synthesizer sound. Now increase the resonance control, R41, and you will hear the "wah-wah" sound become more intense. Play around with the coarse- and fine-tuning controls and also attenuator R38 to alter the effect.

As you have seen, the use of the SSM2040 makes the construction of a high-quality lowpass VCF quite simple. You will find this filter has a very low noise figure and is useable in the most demanding of situations. As with any multiparameter module, though, you must experiment, practice and rehearse to get the maximum benefit from the circuit. And don't forget to try out the sine wave oscillation function; you will probably be quite surprised at the quality of the sine wave! Distortion figures of well under 1% are easily possible.

Chapter 7

A Dual VCA

The VCA (voltage-controlled amplifier) is responsible for imposing an amplitude envelope on some audio signal. At first glance it may seem like a synthesizer really only needs one VCA, but upon further reflection it starts to look like the more VCAs the better! Besides imparting envelope characteristics on audio signals, they are also useful for dynamic-depth FM, delayed vibrato, quadraphonic panning and several other important electronic music applications.

But VCAs can get expensive, and often the beginner had to make do with fewer than he or she would really like. However, things have changed recently, and LSI (large-scale integration) has now produced a chip containing not one, but two very high quality VCAs! This chip is the CFM3330 and is produced by Curtis Electromusic Specialties. Usually, exotic integrated circuits like these are fairly hard to obtain on the hobbyist level, but thanks to a special arrangement, you can obtain this unusual part from PAIA Electronics. (See Appendix A.) The price is under \$10, so there's no excuse not to have all of the VCAs you need!

This circuit uses only two chips total and is so small that you may want to build it with a companion dual ADSR. If you do so, note that the dual VCA and dual ADSR both fit conveniently behind a 3 1/2-inch by 19-inch rack panel. You'll find that a dual VCA/dual ADSR module makes a pretty sharp combination!

As mentioned, this is a simple VCA circuit, and that's the name of the game. Actually, in general VCAs can get pretty wild. For example, the ones used in a noise-reduction unit or in an automated mixer can get very complicated. This is because such applications demand very wide range, low noise, and accurate tracking. However, for electronic music it isn't

necessary to go that far. As I mentioned in Chapter 2, the human ear isn't nearly as sensitive to amplitude variations as frequency changes. This being the case, you can essentially ignore accurate tracking as being a rather minor consideration in electronic music. However, a fairly wide dynamic range is important, and this also implies the need for a low-noise circuit. Finally, price is important. You'll want as many VCAs as you can get, and a low price makes this feasible.

As mentioned, the CEM3330 meets these requirements quite well. It's inexpensive, has low noise and a very wide dynamic range, and was designed especially for electronic music applications. The chip has lot's of features, but that doesn't mean you have to use them all! So in this design, you'll strip it down, streamline it, and sometimes ignore various features of the CEM3330. What's left is a terrific, low noise, simple-to-build, dual VCA!

I recommend that you obtain a copy of the spec sheet and application note for the CEM3330 since it contains much useful information concerning the workings of the chip. You can get this at the same time as you buy the chip itself. Look the sheet over; it will make much of what's in this chapter a little easier to understand.

HOW THE CIRCUIT WORKS

Figure 7-1 shows the schematic. Since this is a dual unit, only one-half of the circuit will be described (the other half is identical, except that the power supply connections are shared by both halves). The audio input enters through J2; S1 chooses either ac coupling (for audio processing) or dc coupling (for processing control voltages). If the audio signal were not centered about ground (i.e., if it has a dc offset), you would hear a terrific "thump" every time you opened the VCA quickly. You can certainly do without that, and that's the purpose of putting C7 into the circuit. To further reduce thumping, trimpot R19 allows you to trim out any residual offset in the chip itself. To set this trimmer, repeatedly hit some fast envelopes and adjust R19 for minimum thumping in the audio output.

R26 is an attenuator that pares the signal down to size as needed. This design follows the usual standard: the expected audio signal should be 10V peak-to-peak and the control voltage should be a 0V to +5V control signal. With R26 wide open, the VCA accepts the full 10V peak-to-peak audio signal.

A few words should be said about D1, R5, and C5. Ordinarily, pin 4 should remain at a voltage somewhat below 0.7V. If that voltage were to rise above this level, latch-up and possible damage to the chip would ensue. D1 makes sure this never happens by clamping pin 4 to a maximum of 0.7V (a diode drop). R5 and C5 form a compensation network. Their job is to help C1 (the actual compensation capacitor) keep the amplifier from breaking into supersonic oscillation.

The CEM3330 is a current-in, current-out, current-controlled ampli-

fier. R15 converts the input signal to a current, while the output current is converted back to a voltage by R16 and op amp A1. The output is then presented to J1. Once again, this circuit is set up to give a 10V peak-to-peak output signal under normal conditions and R7 puts the impedance at 1k.

The exponential control at pin 6 is not used in this circuit, so we simply ground it. The reason for using the linear input instead of the exponential input is simple. Almost all ADSRs (including the one to be described in Chapter 8) already put out an exponential control voltage. This being the case, it makes sense to follow the ADSR with a linear VCA since the end result will be an exponential envelope anyway. This is perhaps the most "natural" envelope. However, suppose that you were to follow the exponential ADSR with an exponentially responding VCA. The result would be "exponentiating the exponential," an unusual envelope to say the least. Let's play it simple and ground pin 6!

Pin 7 is the linear control input, which, unlike pin 6, is at virtual ground (which in plain language means that you can sum as many inputs into this node as you want). The full-strength control voltage input feeds J3 and then R13. The value of this resistor is such that a 0V to +5V input yields full off to unity gain. This input would normally be used with an ADSR. J4 is another control voltage input with an associated attenuator, R24. This would most often be used for tremolo or amplitude modulation applications. R25 (which should have a linear taper) is an initial set pot, with the wiper centered giving full off. This pot can be used to manually open the VCA, or to offset any possible bias present in a control signal applied to J4.

C3 is a rather new development. If you have the spec sheet for the CEM3330, you won't find C3 mentioned anywhere. But if you are a subscriber to *Synthesource* (see Appendix B), you may remember a little note that was presented in the Winter 1981 issue, page 11, on how to stabilize the linear control input. This capacitor helps prevent spurious oscillations at low control currents.

That's all there is to it! Because the above explanation also goes for the other half of this chip, you now have a full understanding of the complete dual, voltage-controlled amplifier module. It should be obvious that this circuit represents a real breakthrough in price, simplicity and quality!

CONSTRUCTION

Unlike other circuits in a synthesizer, this one is fairly critical with regard to layout. Since this is an amplifier and since small currents are involved at some settings of the controls, it is most important to minimize stray capacitance, hum and noise. The best way to accomplish this is to use a printed circuit board. If you must use another means of construction, pay strict attention to the wiring layout. Route all of the wires neatly and efficiently, and be sure to use shielded cable if the various inputs are more than several inches away from the front panel.

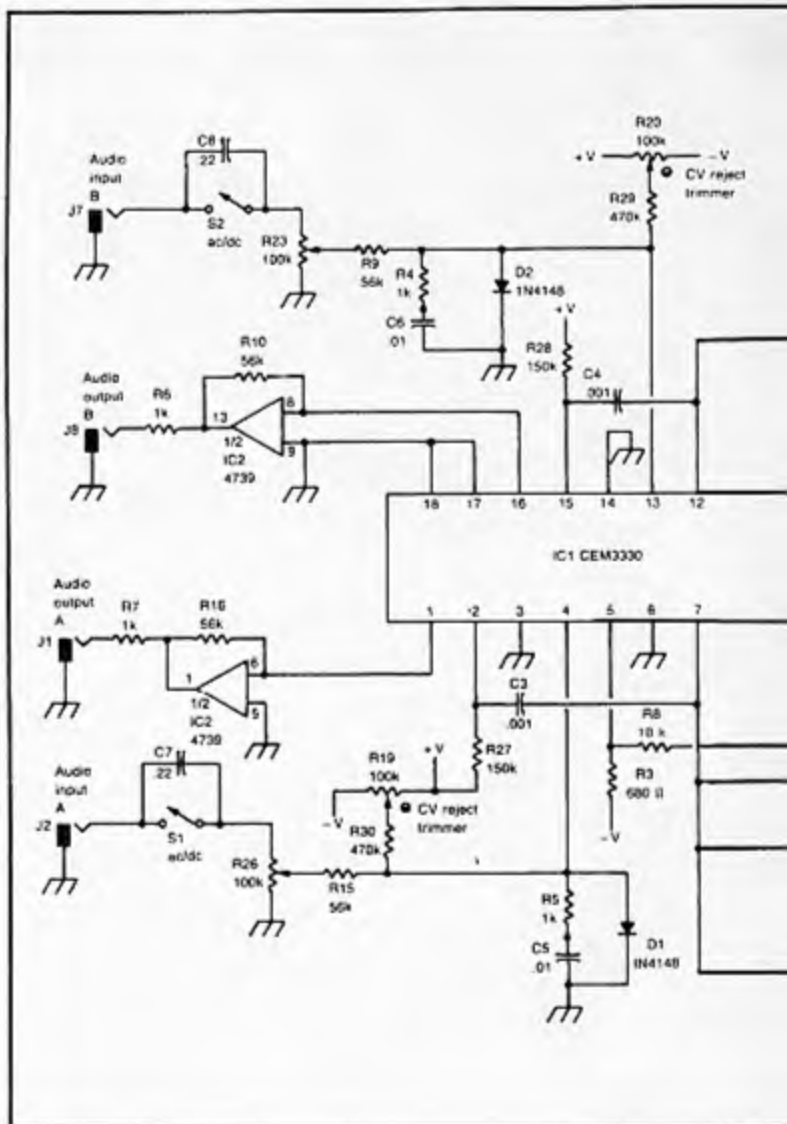
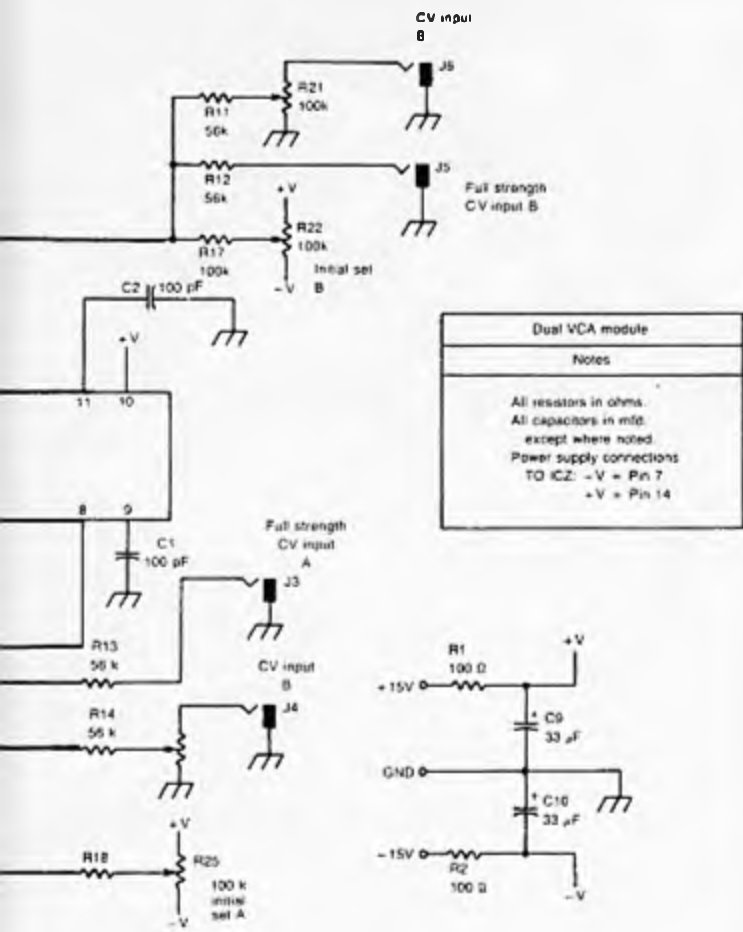


Fig. 7-1. Schematic diagram for the Dual VCA module.

And speaking of panels, as mentioned earlier, this circuit takes up little room. You may want to combine it with a dual ADSR and put both circuits behind one 3 1/2-inch by 19-inch rack panel. If you do, be sure to run separate power supply lines to each sub-circuit. This will keep the ADSR from interacting with the VCA and vice versa.



The initial set pots, R22 and R25, should have linear tapers, since these act as center-null controls. The other pots should be audio taper for best results.

Other than the conditions mentioned above, construction of the circuit is straightforward. As usual, neatness and patience pay off in the long run.

Resistors		Capacitors	
R1, R2	100	C1, C2	100 pF disc
R3	680	C3, C4	.001 mfd. mylar
R4-R7	1k	C5, C6	.01 mfd. mylar
R8	10k	C7, C8	.22 mfd. mylar
R9-R16	56k	C9, C10	33 mfd. electrolytic
R17, R18	100k		
R19, R20	100k trimmer	Semiconductors	
R21-R26	100k potentiometer	IC1	CEM3330 dual-VCA chip
R27, R28	150k	IC2	4738 dual op amp
R29, R30	470k	D1, D2	1N4148 diode
Miscellaneous			
J1-J8	open circuit phone jack		
S1, S2	SPST toggle switch		

Fig. 7-2 Parts list for the Dual VCA module.

Figure 7-2 shows the complete parts list.

CALIBRATION

The only trim pots to adjust are R19 and R20, the control-voltage reject trimmers. These only take a minute to trim up. Let's see how.

Connect an amplifier/speaker combination to the output of the VCA at J1. (Watch the volume of the amplifier, this can get loud!) Now turn down input attenuator R26 and put the initial set pot, R25, at mid rotation. Next apply a series of +5V triggers to J3. These can be provided by an LFO or a keyboard. If you're using a keyboard, repeatedly tap a key to keep generating a trigger.

Now while monitoring the sound, trim up R19 until the popping and clicking is minimized. Repeat the above steps for the other half of the VCA and you're all set to go.

USING THE VCA

Since this is a general-purpose module, you will probably find many different ways to use it. Here are a few tips to get you started. When modulating dc control signals, close switch S1. This will allow the control voltage to be passed by the VCA without any interference from C7. But when controlling audio signals, be sure to open S1 and you'll avoid clicks, thumps or pops.

Remember that J3 expects to see a 0 to +5V control voltage. If the control voltage is greater than this, switch over and use J4 instead. Then you can attenuate the signal down to size with R24. Remember too that the initial set pot, R25, sums in a fixed bias. If you're hearing some "bleed-

through." turn this pot counterclockwise until the sound disappears.

And here's an experiment for you to ponder. How would you do an automatic "ping-pong" effect using the two VCAs, an LFO and an inverter? Think about it, and if you have the modules, give it a try. The sound is just great through a stereo system!

Chapter 8

A One-Chip ADSR

The technology of electronic music is growing so quickly that it is often difficult to keep abreast of all the developments. For example, even though you try to keep up to date on new LSI (large-scale integration) chips available for music applications, one will slip by you every now and then. For example, are you familiar with the SSM2056 ADSR chip? If you aren't, then you'll be amazed when you check it out and find that this IC incorporates all of the design features discussed in this book! The supply voltages and input and output levels are 100 % compatible with the circuits discussed so far. And best of all, the complete ADSR circuit can be implemented with just one chip!

Even if you have already built another kind of ADSR, you will want to check this one out. Why? Well, as you continue to gain experience in electronic music, you will find that no two modules are exactly alike. Even two "identical" circuits will have subtle differences that make one more suitable for a particular application than the other. This is most apparent in the case of filters (everybody seems to have a favorite filter), and to a lesser extent with ADSRs. As you'll find out after building it, this ADSR has quite a different "feel" from most other units. In particular, it seems to offer a bit more control over short attack and decay times, thus making it eminently useful for percussive sounds. Besides the good "feel," this circuit is easy to build and inexpensive to boot.

HOW THE CIRCUIT WORKS

If all of this sounds good to you, here's what you need to know to get it up and running in your system. (Refer to the schematic in Fig. 8-1.) The

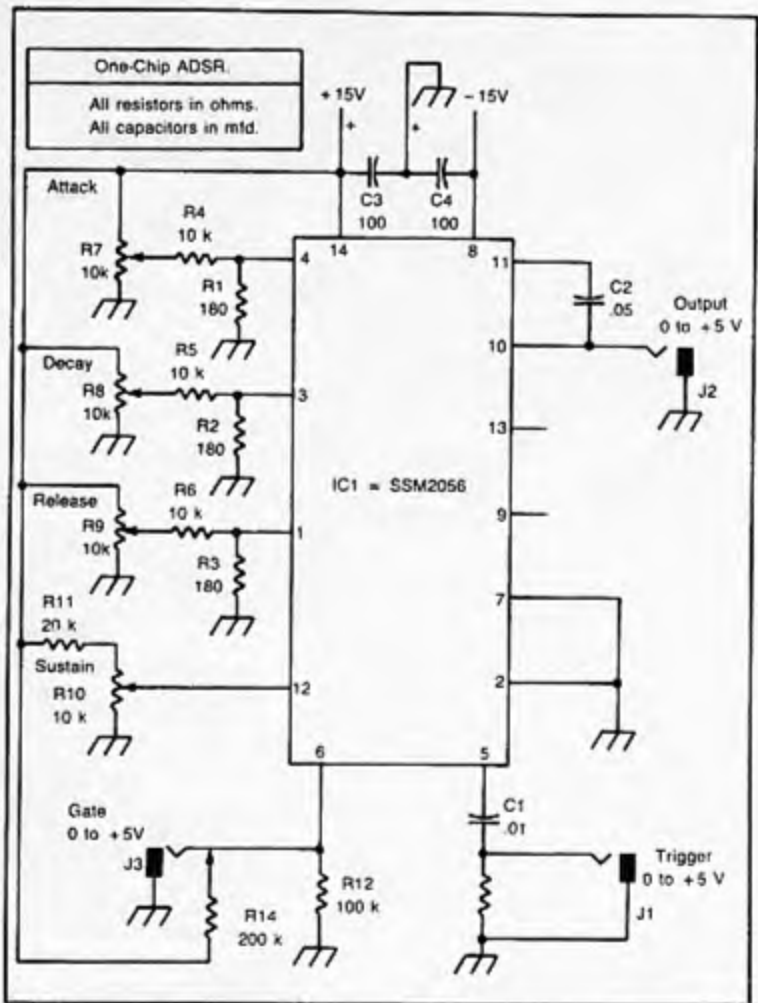


Fig. 8-1. Schematic diagram for the one-chip ADSR.

heart of the whole circuit, of course, is the SSM2056. Later, some sources for this part will be mentioned. This chip was designed to operate off a standard bipolar 15V supply, with pin 14 being at +15V and pin 8 at -15V. Since an ADSR has quite a bit of switching going on in it, fairly hefty capacitors are strung across the power pins (C3 and C4). A value of 100 mfd. has been employed here and though this may be more than is actually needed, it never hurts to be extra careful when it comes to decoupling and bypassing.

A trigger signal is sent to the chip via J1, R13 and C1. This input signal should be a standard +5V, 1-millisecond-wide pulse. The gate signal

is coupled to the IC through closed circuit jack J3. This jack is set up to provide a constant +5V gate signal to the chip if no plug is inserted at the moment. (Voltage divider R14 and R12 drop +5V from the power supply.) With the addition of this switching jack, it is possible to fire the ADSR with a trigger only, thus giving AD type envelopes. For most purposes we will want a full ADSR response, but sometimes an AD envelope is more suited to the application at hand. This is especially true for percussion effects.

If a plug is inserted into jack J3, the constant gate feature is disabled. The plug now presents the normal ON/OFF type keyboard gate to the chip. A little later we'll see how to make the most of the various types of envelope patterns available with this circuit.

The time parameters are set via potentiometers R7 through R9. Thus the ATTACK, DECAY and RELEASE times may be easily set with the turn of a knob. Since the hot sides of these pots are hooked up to the +15V power supply line, the wiper voltages must be attenuated. R4 and R1 form a typical voltage divider. The maximum voltage of +15V is dropped to about 260 mV, a level the SSM2056 likes to see.

The SUSTAIN control is handled a little differently. A fixed 20k resistor (R11) is added to potentiometer R10 to drop the +15V supply line to about +5V. Thus the hottest setting of this pot is at +5V. This in turn means that the SUSTAIN voltage is variable from 0V to +5V, as we would expect it to be. By the way, 20k is a standard 5% value and can be easily obtained from a number of dealers (see Appendix A). Do not substitute a 22k resistor, for example, for this critical value.

C2 is the timing capacitor for the whole ADSR. Use a good quality capacitor here; mylar is perhaps the best choice, being both fairly stable and not too expensive.

The output is at J2. Only one jack is shown here, but in most systems, you will want to tie four jacks in parallel for the output structure. You'll probably find many cases where you will use one ADSR to drive several circuits. So by making a number of output jacks available, you will get around using up some multiples elsewhere in your system. Four jacks should be the limit, however, since this is about the maximum that the internal buffer of the SSM2056 is able to handle.

Incidentally, this is one type of circuit where you won't want a 1k output impedance. In fact, you will want the lowest impedance possible since you will often be driving many other circuits. This was discussed previously.

CONSTRUCTION

Having described the circuit's operation, I want to talk about obtaining parts and building the thing. Figure 8-2 shows the parts list. First of all, where do you get the SSM2056? In the past, obtaining single-unit quantities of the more exotic integrated circuits was rather difficult. However, things are easing up now, and you should have no trouble at all. PGS Elec-

Resistors		Capacitors	
R1-R3	180	C1	.01 mfd. mylar
R4-R6	10k	C2	.05 mfd. mylar
R7-R10	10k potentiometer	C3, C4	100 mfd. electrolytic
R11	20k		
R12, R13	100k		
R14	200k		
		Semiconductors	
		IC1	SSM2058 ADSR chip
		Miscellaneous	
		J1, J2	open circuit phone jack
		J3	closed circuit phone jack

Fig. 8-2. Parts list for the one-chip ADSR.

tronics is one source (see Appendix A). The price is under \$6, though there may be a shipping and handling charge involved.

The two rather important resistors, R11 and R14 (20k and 200k, respectively) are standard 5 % values and can be obtained from a number of places. One good source for resistors is Jameco Electronics (see Appendix A). The price is around six cents each for resistors, but you'll have to make sure you meet the minimum order requirements. Write to both PGS and Jameco for catalogs and ordering information.

Since this is such a simple circuit, it can easily be built on the Experimenter Printed Circuit Board available from Radio Shack. This is one of those generalized breadboard rigs that has a number of rows of pads and traces suitable for building IC-type circuits on. The stock number is 276-170 and the price is about \$3.

Circuit construction is not critical, because there are only control-voltage-type frequencies involved. Along with the circuit board mentioned above, use hookup wire and flea clips to finish the construction.

The one-chip ADSR can easily be mounted behind a standard 1 3/4-inch by 19-inch rack panel or combined with the VCA of Chapter 7 and mounted behind a larger panel. In either case use some small angles and #4 hardware to fasten the circuit board to the front panel. You'll probably find that you'll have some front panel space left over. You might want to dedicate this extra space to a one-by-four multiple. Multiples are always handy things to have, so when a chance presents itself, like this, seize the opportunity to throw one in!

USING THE ADSR

There are no trim pots used in this circuit, so you can hook it up and

try it out right away. Because ADSRs may be new to some readers, here are a few tips to get you started. For a full ADSR response, apply both a gate and trigger from the keyboard. Now depress a key and hold it. The instant you depress the key, the envelope will launch into its ATTACK portion. When the signal reaches +5V, the DECAY portion will start up. The signal will decay to the level set by the SUSTAIN control and will hold there for as long as the key is depressed. Now release the key, and the RELEASE portion starts up.

To get an AR-type response, once again apply both a gate and trigger. Turn the DECAY control down all the way, and the SUSTAIN control up all the way. Now when a key is depressed, you will get an ATTACK/SUSTAIN/RELEASE pattern, typical of an AR unit.

For AD effects, you need only apply a trigger signal. Turn the SUSTAIN and RELEASE controls down all the way. Now whenever you touch a key, the unit will go into an ATTACK/DECAY cycle automatically. As was mentioned earlier, this is usually the appropriate waveform for percussive effects.

Of course, these three arrangements just described are just the start. There are countless other settings of the controls and hence countless other sounds. Let your ear be the judge of which are most appealing to you! And remember, this circuit allows for full retriggering, so the versatility is even greater.

It probably didn't seem possible, but you now have a very complete one-chip ADSR! Alert readers will note a similarity to Craig Anderton's "Voltage-Controlled Envelope Generator," (*Contemporary Keyboard*, May 1982, pp. 20-23). Anderton's circuit employed the SSM2055, an earlier generation of the SSM2056 used herein, and also offered voltage control of the various parameters. If you're looking for even greater versatility (at the expense of greater complexity, though), check this circuit out. But for most common applications, you'll probably find the simple manually controlled version is more than adequate.

If you're just learning the how-to's of building synthesizer gear, you'll find this ADSR is a great project to start! Pay attention to the supply voltages, capacitor polarities, watch your soldering techniques, and before you know it you'll have a very professional quality ADSR up and running!

Chapter 9

The Supercontroller

If the VCO, VCF, and VCA form the heart of a synthesizer, then this module is surely the brain! It gives you most standard controller options (LFO, noise source, and sample and hold), but also includes some extras that have no real precedent on commercially available synthesizers.

The module is called the Supercontroller and is actually the equivalent of three modules in one. As such, the circuit is quite large, but if you analyze each function by itself the circuit action is quite easy to understand.

INTRODUCING THE SN76477

The heart of the Supercontroller Module (SCM) is the SN76477. This chip is known as a complex noise generator and was originally designed to provide sound effects for pinball machines and other computer type games. It contains a VCO, LFO, envelope generator, noise source, VCA, mixer, and regulator all in one 28-pin package! When you first meet the chip, you'll probably think that it does a lot. And it does. The chip certainly represents a triumph of large-scale integration, incorporating many different sound functions in one package.

As time goes on and you become more familiar with the chip, though you'll find that the VCO doesn't have a very large sweep range and is temperature sensitive. The noise source is fairly static and there isn't a very wide selection of different sounds. Most importantly, you will probably notice that despite the large number of functions, the chip is basically organized in a fixed manner. That is, it contains lots of sub-circuits, but they are all interconnected internally and rearrangement of them seems impossible. This being the case, you might draw the conclusion that the SN76477

is suitable for pinball machines and kiddie games, but isn't really well adapted for serious synthesizer work.

Well, look again! As it turns out, with some external buffers and other parts, it is possible to separate the functions, get a broader range of sounds from the noise source and do many other neat things. And best of all, by doing so, you unlock the basic potential of the chip and make it really great for serious synthesis.

As mentioned, the heart of the Supercontroller Module is the SN76477. Before attempting to build this circuit, be sure to read over the application note that comes with the chip. In this way, you will be better prepared to understand some of the nonstandard ways in which the IC is put to use. Also, because this is such a big circuit, you should take your time during the planning stage: read the application note and study this chapter carefully. The time you spend preparing to construct the circuit will be more than repaid when you build it, turn it on, and get it working right off the bat.

By the way, the Supercontroller Module has been well tested and revised to make it as versatile and "bug free" as possible. In fact, this is a third generation design and incorporates a number of features suggested by professional musicians. As such, you will find yourself using it often and attaining sounds unheard of in other synthesizers. A little later in the chapter, some sample patches are suggested.

HOW THE CIRCUIT WORKS

The SCM requires a regulated and well-filtered source of +15V, -15V, and +5V dc. If you only have a bipolar 15V supply, simply add an onboard +5V regulator to the circuit. The current drain on the +5V supply is quite low, so no heatsink is required on the regulator. Carefully note which points on the schematic connect to which power supplies; incorrect connections will keep the SCM from working correctly and could cause damage to the SN76477.

As mentioned, the SCM consists of a sample and hold, LFO and noise source. Let's consider the noise source first and see how it works. The white noise source in the SN76477 is not the back-biased transistor you might be familiar with, but a binary pseudo-random noise generator. Without getting too heavily into the theory of this (see Don Lancaster's *TTL Cookbook*, Howard W. Sams, 1974, pp. 277-283 for a good treatment), all you really need to know is that the noise source is basically a shift register, with various bits moving down the register at a rate determined by a master clock. This clock can be either internal or external to the chip.

To use the internal clock, a resistor is connected from pin 4 of the SN76477 to ground, and this sets the clock rate. For best results however, you should avoid the internal clock completely and use an external clock instead. Actually, the "external" clock in the SCM is internal to the chip, being the VCO. This is a good way to save parts, space and wiring hassles.

Before getting too heavily entrenched in details, something should be

said about why you might want to change the clock rate. The best answer is "try it, you'll like it!" The sound is an incredible swooshing noise not unlike phasing or flanging. The noise takes on a new tonality and sweeping the clock changes the spectra in a dramatic and eerie manner.

Let's analyze how the external clock is set up. Refer to the schematic in Fig. 9-1. R15 and a capacitor selected by S1 (C3, C7 or C8) set the basic VCO frequency. A control voltage applied to J2 is summed through A1 and applied to the control voltage input of the VCO at pin 16 and this sweeps the VCO frequency. Since A1 is an inverting stage, an increasing voltage at J1 yields an increasing frequency. R48 (the initial noise clock pot) offsets the VCO if desired, or can be used as a manual sweep control. Trimmer R46 sets the zero point of the VCO so that 0V applied to J2 gives the minimum VCO frequency. Diode D4 prevents any inadvertent negative voltages from creeping into the SN76477. The VCO's square-wave output is taken off of pin 13 and sent to the external noise clock input at pin 3. Pin 4 is tied to the +5V line which programs the chip to accept an external clock.

The output of the noise source is tapped off of the noise filter capacitor at pin 6. The noise must be taken at this point since the chip's normal output, pin 13, is already committed to the VCO. The signal is buffered by source follower Q7, and amplified and level-shifted by A10. The output at J3 is a standard 10V peak-to-peak with a 1k impedance.

This covers the operation of the noise source. As a review, consider the basic elements again: the VCO is used as an external clock; it then drives the noise source via pin 3, and the output of the noise source is tapped at pin 6. By sweeping the VCO, the noise source is also swept and the result is a very unusual sound, not available on most other synthesizers.

Check out the LFO circuitry next. The LFO has lots of options, so let's examine them one by one. To simplify matters, we'll ignore Q1 and its related circuitry for the moment.

Ordinarily, the only LFO waveform accessible at the output of the SN76477 is a square wave; however you can get a triangle wave if you tap the waveform off of the timing capacitor, making sure that you buffer it sufficiently to prevent loading. The timing capacitor is selected by S2 (C9, C11 or C13), and this sets the basic range for the LFO. The triangle wave is taken off of one of these capacitors and is buffered by A2. This op amp is set up as a voltage follower, and if an FET type is used, it will have an input impedance of many megohms!

The triangle wave at this point goes from about +0.4V to +2.8V for a total swing of 2.4V. This is hardly standard! But when amplified by A4 and level shifted by R55, the output becomes a 10V peak-to-peak (centered about ground) triangle wave. R55 sets the approximate offset of the output; trimmer R47 centers the output exactly about ground. LED D6 at the output of A4 monitors the LFO rate.

As mentioned, S2 sets the basic range of the LFO. Potentiometer R57 gives continuous control over the rate. Between S2 and R57 you will be

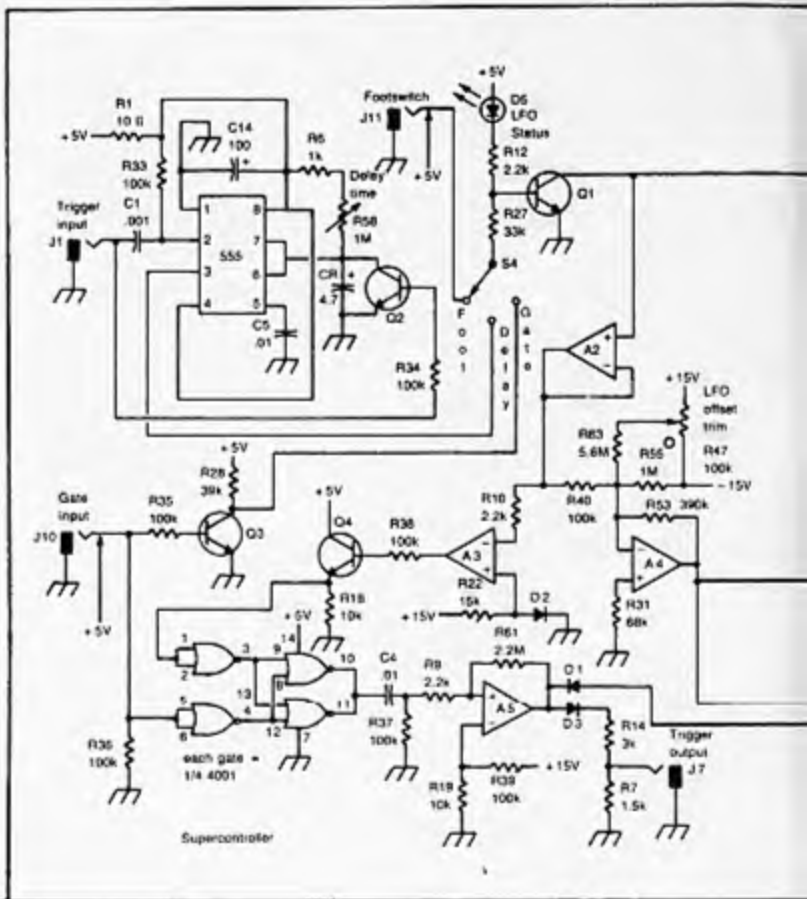


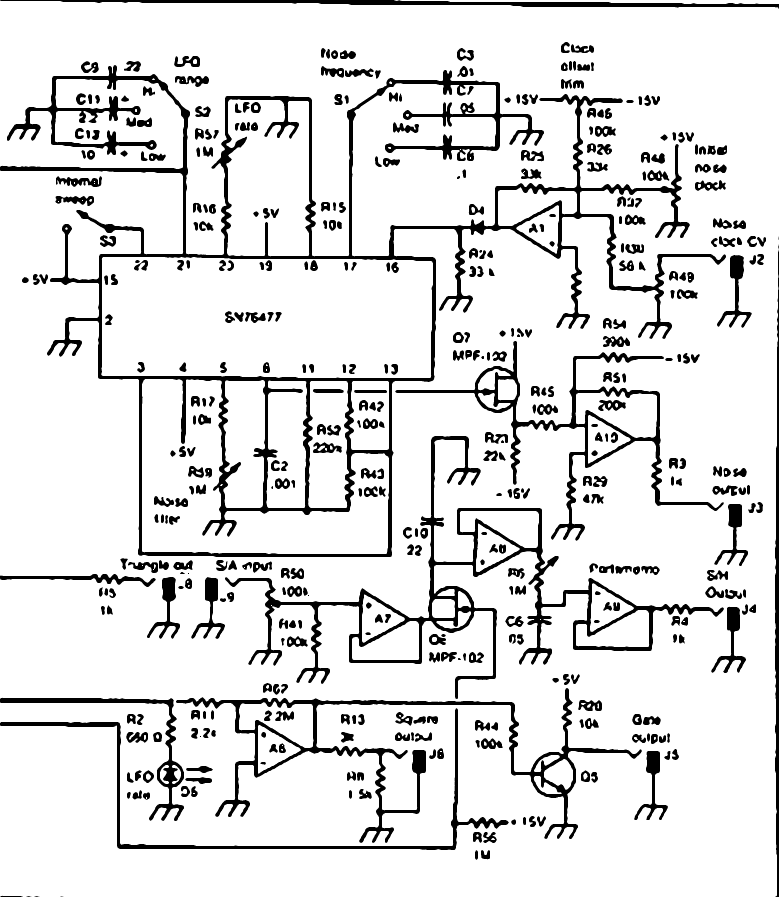
Fig. 9-1. Schematic diagram for the supercontroller.

able to attain a wide range of LFO rates.

The triangle wave also feeds comparator A6 and its related components which generates a square wave. R62 adds 0.1 % of hysteresis, which gives the resulting square wave a nice clean edge. R8 and R13 attenuate this signal, giving a standard 10V peak-to-peak square-wave output with a 1k output impedance, centered about ground.

Since there are many times in synthesis when a 10V peak-to-peak signal is inappropriate, a gate signal going from 0V to +5V is also derived from this square wave and is available at Q5's collector (the transistor is powered off the +5V supply, hence the 5V swing). This gate would typically be used for driving ADSRs in sync with the LFO.

Now let's shift our attention to the circuitry associated with Q1. This is set up as a switch that can discharge whichever LFO timing capacitor



is selected by S2. Not only can Q1 discharge the timing cap, but it can also hold the LFO waveform output at 0V.

With S4 in the footswitch position, and an ordinary SPST push on/push off footswitch plugged into jack J11, closing the footswitch ties the junction of R12 and R27 to ground. Thus Q1 is off and the LFO runs normally. Status LED D5 is lit to indicate that the LFO is running. However, opening the footswitch releases the junction of R12 and R27 from ground, which initiates two events. First, the status LED is extinguished; second, Q1 is turned on, and this shorts the LFO's timing capacitor to ground. In other words, the LFO is in an "hold" state and the extinguished LED indicates this fact. Footswitch control of the LFO, while simple to implement, can really add a lot of versatility to your sound.

Now consider the delay function. To make it clear how this works, let's

consider a simple patch. First, set S4 to the delay position; then, imagine that you have taken a +5V trigger out from your keyboard to the delay trigger input at J1. Patch the triangle wave output of the LFO to an FM input of a VCO. Now let's follow the chain of events. Push a key down on the keyboard and a trigger enters J1, thus turning the 555 timer on for a time determined by C12, R6 and R58. This is a one-shot circuit, and so pin 3 goes high. Note that pin 3 is connected via S4 to the R12/R27 junction mentioned above. Since this junction is high, Q1 turns on and therefore turns the LFO off for the delay time set by the one-shot. After the one-shot turns off, the R12/R27 junction is brought to ground and the LFO turns on again. The result is a pleasing delayed vibrato effect, which is extremely well suited to creating string sounds.

Note that C12, the timing capacitor for the one-shot is bridged by Q2 in the same way that the LFO capacitor is bridged by Q1. This guarantees that each trigger coming in to the delay circuitry shorts out C12 momentarily, so that its charging always starts from 0V. In more technical terms, the one-shot is retriggerable. Thus, as long as your keyboard is putting out triggers, the LFO is off (remember, each new trigger discharges C8 and starts the one-shot all over again). But as soon as the triggers stop, and as soon as the one-shot turns off, the LFO turns on again. And of course, the status LED monitors the whole thing. R58 adjusts the delay time.

Next let's consider the gating function of the LFO. To give you an idea of where we're going with this, let's set up the LFO so that when you push a key down on the keyboard it puts out a series of triggers, suitable for driving an envelope generator. The result is repeating envelopes under keyboard control, similar to a banjo plectrum effect.

Put S4 in the gate position and connect the gate output of your synth to the gate input jack, J10. Suppose that no key is depressed. With no gate present, Q3 is off, and its collector is at +5V. The collector is coupled to the junction of R12/R27 via S4, hence Q1 is on, and the LFO is in a hold state. Now, depress a key. This sets up a chain reaction; current flows through R35 into the base of Q3, thus turning it on. This pulls the collector (and the R12/R27 junction) down to ground, which shuts off Q1, and allows the LFO to run. Hence the various LFO outputs are off and running.

Take a look at the trigger output. Suppose the gate input is off (no keys down); this holds the LFO triangle wave (at pin 21) at 0V. This is buffered by A2, and the 0V output is applied to the inverting input of A3. A3's noninverting input is held at 0.7V by virtue of the voltage drop across diode D2 in series with current limiting resistor R22. Since the noninverting input voltage exceeds that of the inverting input, A3's output is high. This "output high" condition is chopped down to a +5V level by Q1 (which is configured in a noninverting manner), and the output of Q4 is coupled to an AND gate composed of the four sections of the 4001. The AND gate output is differentiated by C4 and squared up to a nice 1 msec pulse by A5, which is then clipped and attenuated by D3/R14/R7, presenting a +5V output trigger.

However, remember that I said the gate input is off. As a result, the other input of the AND gate is low (pins 5 and 6 of the 4001), so no triggers appear at the output yet. But push a key down (gate on), and the AND output goes high, which allows triggers to pass to the output.

This may seem like an elaborate way to generate triggers under keyboard control, but it is necessary. Without this logic scheme, there would be a perceptible delay time between pushing a key down and the appearance of the first trigger. Other ways of attempting this fail miserably and make it extremely difficult to use in a musical context.

The triggers at A5's output also clock the S/H (sample and hold). A7 is an input buffer for the S/H, and its output is applied to FET Q6, which acts like a switch. With no trigger applied, R56 keeps the FET pinched off, hence no current flows through the FET channel. But when a negative-going trigger hits the FET's gate, it turns on, allowing whatever voltage is at the output of A7 to pass to the hold capacitor C10. Then the FET turns off again and the charge is safely "sealed in" by the FET on one side and A8 on the other. A8 must be a FET-type op amp. The output of A8 reflects the charge on C10, and that charge passes across R60 into C6. R60 is a *portamento* control that can "glide" or "slur" the sampled voltages together. The output is buffered by an FET-type op amp, A9, and is then presented to the S/H output.

This, then, covers the complete operation of the Supercontroller Module. Even though it is a big circuit, by analyzing each sub-circuit one at a time, it is possible to understand the entire unit. You shouldn't expect to be able to fathom the circuit at one sitting, however; you'll probably have to reread this section and trace the circuit through a number of times before you understand it completely.

CONSTRUCTION

Because this is such a large project, perhaps the best method of construction is with a printed circuit board. However, there is absolutely nothing critical about the circuit, so there is really no reason why you couldn't build your version with perfboard and flea clips. Figure 9-2 is the complete parts list.

The prototypes of this circuit were built behind standard 3 1/2-inch by 19-inch rack panels. Figure 9-3 shows a suggested layout. The circuit board can be supported on angles and this will decrease the length of wires needed to connect it to the panel controls.

There wasn't room to show them in the schematic, but each power supply line (+15V, -15V and +5V) should have a 100 mfd. electrolytic capacitor draped across it. These serve as decoupling capacitors and help keep the noise source from running down the supply lines to other modules in your system, so be sure to include these three capacitors!

CALIBRATION

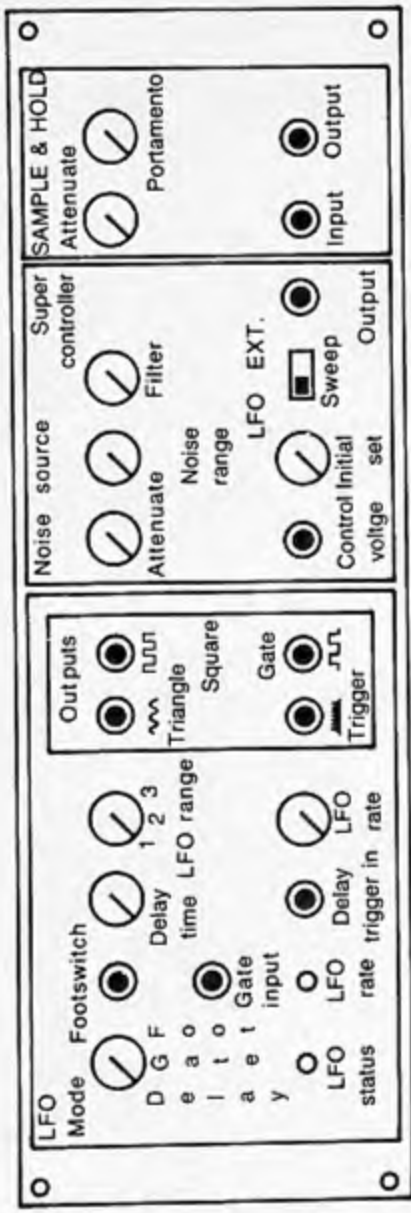
Tweaking the Supercontroller is quite easy. First you need to set R47,

Resistors		Capacitors	
R1	10	C1, C2	.001 mfd mylar
R2	680	C3-C5	.01 mfd mylar
R3 R6	1k	C6, C7	.05 mfd mylar
R7, R8	1.5k	C8	.1 mfd mylar
R9-R12	2 2k	C9, C10	.22 mfd mylar
R13, R14	3k	C11	2.2 mfd electrolytic
R15-R20	10	C12	4.7 mfd electrolytic
R21	12k	C13	10 mfd electrolytic
R22	15k	C14	100 mfd electrolytic
R23	22k		
R24-R27	33k	Semiconductors	
R28	39k	IC1	SN78477 noise chip
R29	47k	IC2	1458 (A1, A10)
R30	56k	IC3	TL084 (A2, A7, A8, A9)
R31	68k	IC4	4136 (A3, A4, A5, A6)
R32-R45	100k	IC5	555 timer
R46-R47	100k trimmer	IC6	4001 quad NOR gate
R48-R50	100k potentiometer	Q1-Q5	2N3904 NPN transistor
R51	200k	Q6, Q7	MPF-102 FET
R52	220k	D1-D4	1N4148 diode
R53, R54	390k	D5, D6	LED
R55, R56	1M		
R57-R60	1M potentiometer	Jacks	
R63	4.7M	J1-J9	open circuit phone jack
R61, R62	2.2M	J10, J11	closed circuit phone jack

Fig. 9-2. Parts list for the supercontroller.

which is the dc offset trim for the triangle wave output. The fastest way to set this is to simply monitor the triangle wave on an oscilloscope and set the control so that the triangle wave is symmetrically oriented about ground. If you don't have a scope, monitor the triangle wave with a center-zero voltmeter (most inexpensive VOMs have such a function). Then set the trimmer so that the needle on the meter wings an equal amount on either side of zero volts, or if you're not fussy about zeroed-out triangle waves, simply set the trimmer in mid-position and leave it. This gives good enough results for most applications.

To adjust the clock offset trimmer, R46, monitor the noise source with an amplifier. Turn R48 and R49 completely down, then spin trimmer R46 around a few times to get familiar with its effect. At one extreme you won't hear anything through the monitor amp; at the other extreme you will hear a very shrill white noise sound. Starting from a no-noise position, ease the trimmer up until the noise just starts. This is the optimum position. By setting the trimmer in such a way, the initial set pot, R48, will have a full range effect.



Rack panel: 3.5" X 19"

Fig. 9-3. Suggested front panel layout for the supercontroller.

Well, that's it . . . all built and all tweaked up; now you're ready to make some music!

USING THE SUPERCONTROLLER

There are innumerable uses for the Supercontroller, and hardly a week will go by that you won't find some new, off-the-beaten-path way of using this machine. As with any musical instrument, experience is what really counts. To help you get started here are four very simple patches. But to reiterate, these simple patches are for example only. Once you're familiar with these, you'll find countless others.

Figure 9-4 shows the first patch for a delayed vibrato effect. Follow this patch chart carefully using your own system and the SCM. Note that triggers come from the keyboard to the delay trigger input. To make this more fun, you might want to set up the VCO, VCF and VCA to approximate the sound of a violin.

Now push a key down and hold it. At first the note will appear without any vibrato, and after a certain amount of time the vibrato will enter. Experiment with various settings of the delay time pot, R58. You will note that since the delay time is retriggerable, a series of triggers will keep firing and refiring the timer for as long as you keep playing. The upshot is that as long as you keep your fingers moving on the keyboard no vibrato will occur, but as soon as you stop and hold a key down, the vibrato will

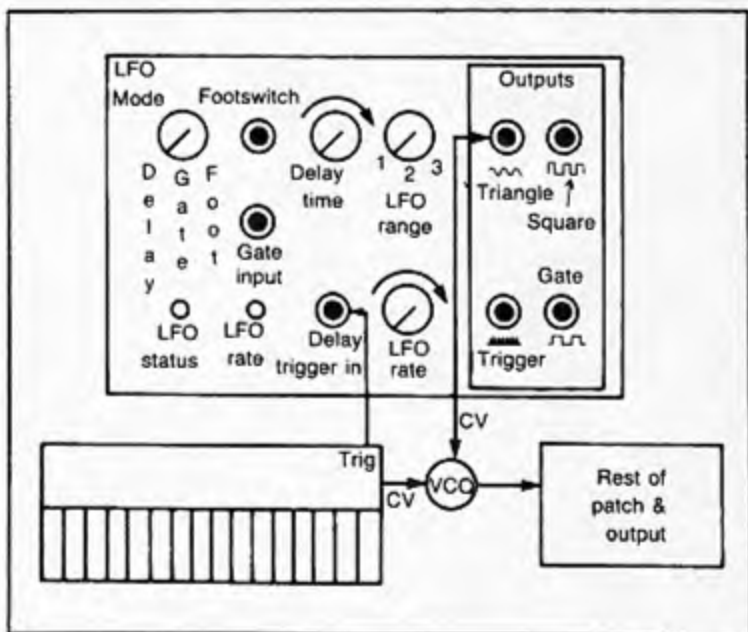


Fig. 9-4. Patch for the delayed vibrato effect.

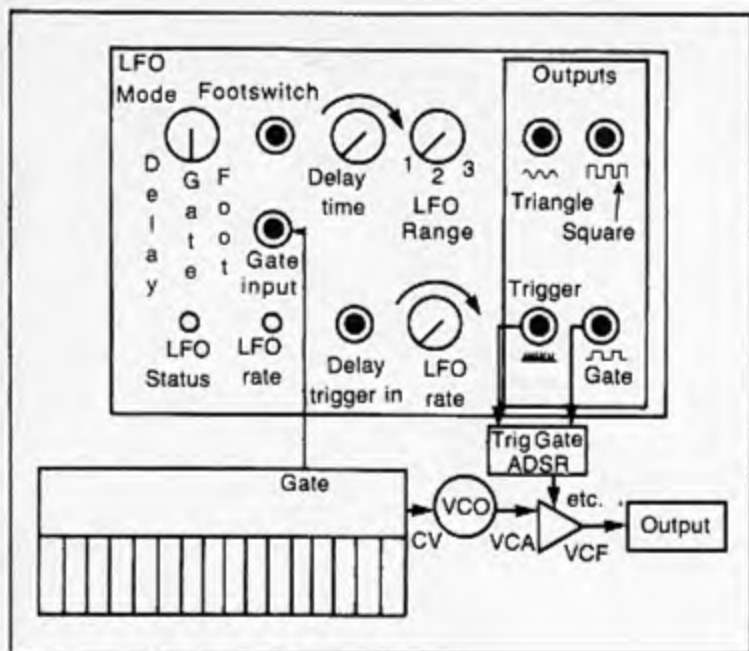


Fig. 9-5. Patch for the gated ADSR effect.

appear after the delay time has elapsed. This can be very useful when you want no vibrato until the last note of a passage.

In Fig. 9-5 a patch for a gated repeating ADSR effect is shown. In this patch the gate from the keyboard determines when the LFO is on, and gate and trigger signals from the LFO fire the ADSR. Push a key down and you get a series of repeating sounds; let up on the key and they stop. You can get some great, banjo-like plucking sounds with this patch.

For a super noise effect, try the patch in Fig. 9-6. The ADSR (besides modulating the VCA) also sweeps the noise source. You'll note that there isn't any filter in this patch, yet push a key down and what do you hear? A sound very much like a low-pass filter being swept! This is due to the special clock-sweep noise described above. Adding a filter to this patch really intensifies the sound. This patch can form the basis of some really far-out percussion effects.

One good use of the sample and hold is shown in Fig. 9-7. The noise source is sampled by the sample and hold unit, and the output is then sent to the VCO. The result is a series of random pitches, with the SCMs own LFO controlling the rate of these pitches. Add some *portamento* by turning up R60 and these random pitches become slurred.

It's up to you now to design some new patches. After only an hour of experimenting you'll be convinced that the Supercontroller Module is the brain of your synthesizer!

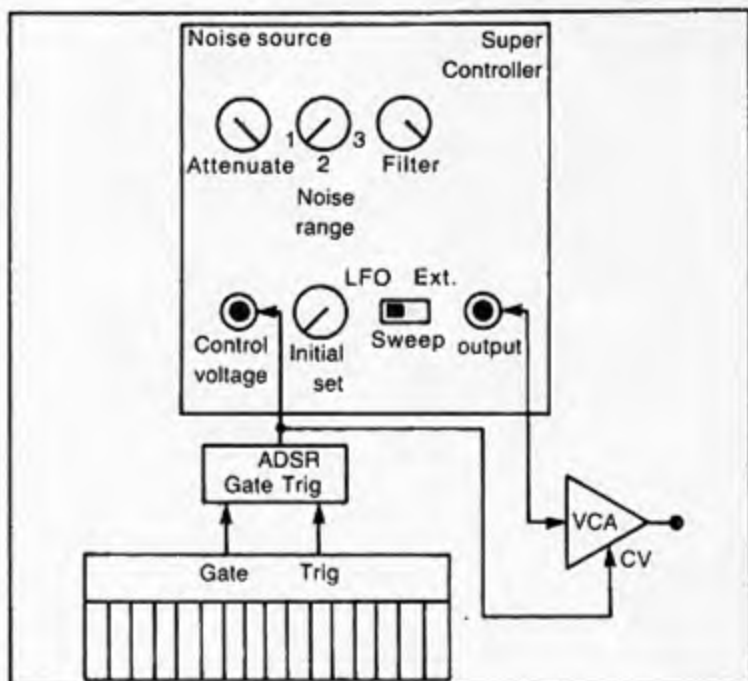


Fig. 9-6. Patch for the bizarre white noise effect.

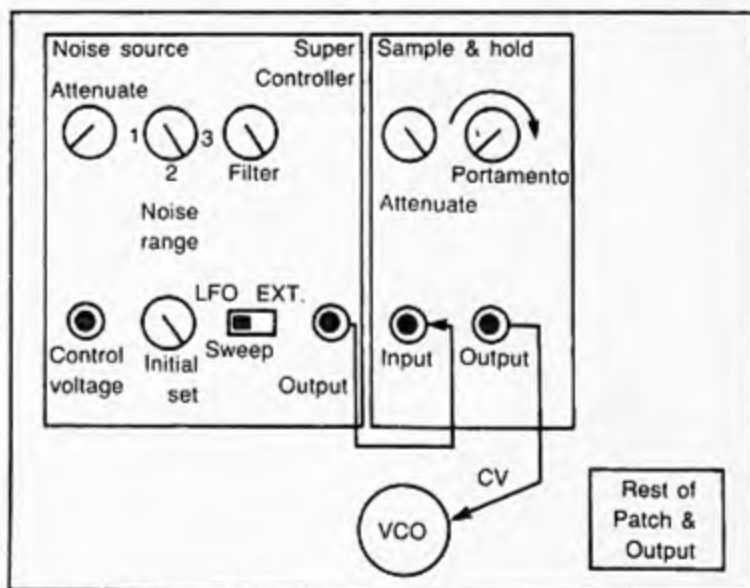


Fig. 9-7. A sample and hold patch.

Chapter 10

A Digital Keyboard

Synthesizer keyboards have always presented a problem to designers. Analog keyboards in particular have been tricky to deal with, since they have often suffered from reliability problems. Typically, contact bounce in the keyswitches, false triggering, and problems with sample- and hold-time constants have served to make such keyboards somewhat temperamental. And since most standard organ-type keyboards utilize a single-bus arrangement, it becomes triply hard to efficiently extract control voltages, triggers and gates simultaneously.

Since the keyboard forms the most intimate link between the performer and the electronics of a synthesizer, reliability is of the utmost importance. No other part of a modern electronic music system shapes a performer's technique as much as the keyboard. Therefore, you must maintain the highest standards, and as mentioned above, the analog approach seems to fall short of these.

In this chapter a digital keyboard is presented which overcomes most of these problems. Because the keyboard deals in binary numbers rather than control voltages, latches can be used, thus dispensing with the sample and hold. In addition, logic methods can be employed to generate the gate and trigger signals, and at the same time provide a certain measure of switch debouncing. Finally, the circuit is ideally suited for single-bus operation, thus simplifying wiring and reducing cost.

HOW THE CIRCUIT WORKS

Though the schematic looks forbidding, in actuality the mode of operation is quite simple. (Refer to the simplified block diagram in Fig. 10-1.) A standard 37-note organ keyboard is modified to conform to a matrix ar-

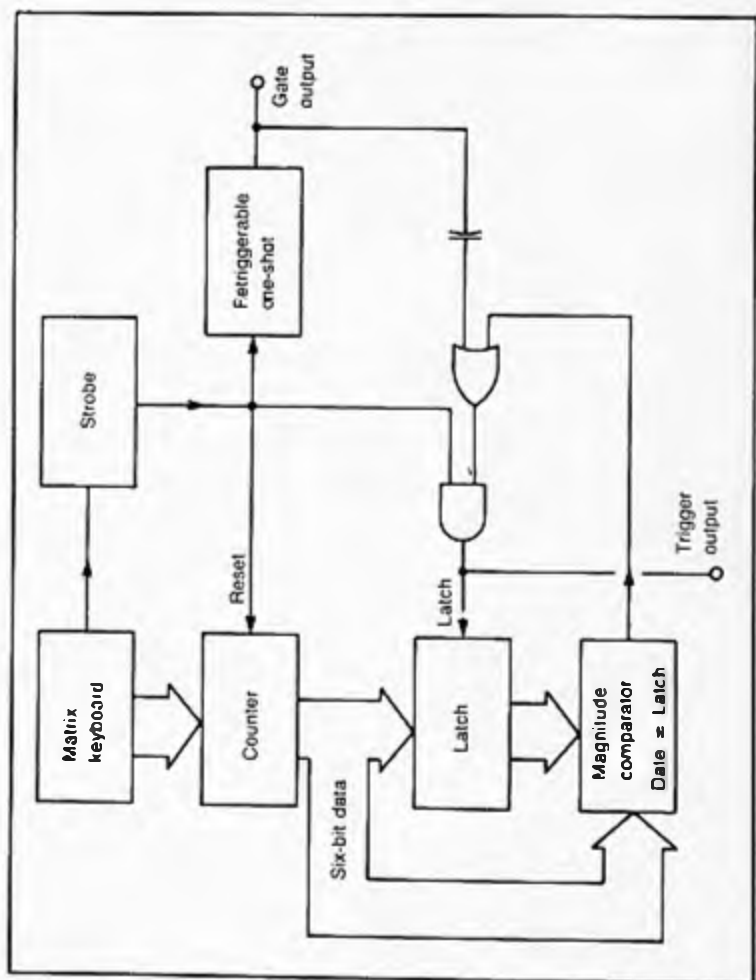


Fig. 10-1. Block diagram of the digital keyboard.

range. That is, starting at low C, the main bus bar is cut into groups of eight notes, with the last set of notes only containing five. A 37-note keyboard can be split up into four groups of eight notes with one group of five notes left over. The first note of each group is tied to the other first notes, the second notes to other second notes, and so on. The result is a matrix of keyswitches. Such an arrangement is similar to that commonly used in calculators. The matrix is continually scanned at a high frequency, and when a key contact is closed, a strobe signal is output. This strobe serves as an indicator that other circuitry should now set the latches, reset counters, etc.

The above description has been highly simplified and only takes into account the case of one switch being closed. What happens when more than one switch is closed? For that matter, what should happen? In general, custom has it that in this case only the lowest note depressed on the keyboard should be accepted. Since all higher notes are ignored, this scheme is typically called the "low note rule." This rule is certainly not the only one possible, but since most synthesists (who probably "grew up" with analog keyboards having the rule) have come to incorporate it in their technique, it is desirable to retain this feature.

Looking at Fig. 10-1 in greater detail, then, it can be seen that as the matrix keyboard is scanned by the binary counter, whenever a key is depressed a strobe is sent to the latches, indicating that the data on the data bus should be latched. Note that the six bits of binary data are sent not only to the latches, but also to a six-bit magnitude comparator. If the present data is found to not equal the previously latched data then the output of the comparator goes high, which in turn sends the OR gate high. The output of this gate is then ANDed with the strobe signal (which is also high at this point) and the new data is latched in. Simultaneously the counter is reset and the matrix scanning starts over again. It is clear, then, that only the lowest note is ever latched. Since the scan always starts at low C and scans upward, and since the scanner/counter is always reset by a strobe, only the lowest note depressed is registered. All higher notes are ignored.

The control voltage is derived by sending the latched data to a digital-to-analog converter. The trigger is taken right off of the latch signal. The gate signal is a little trickier to generate. Whenever a key is depressed and held, a pulse train is generated on the strobe line. This pulse train is far too "choppy" to be used as a gate signal. However, by sending the pulse train to a retriggerable one-shot, a suitable gate signal can be created. A standard one-shot goes ON for a specified amount of time, and then shuts down. A retriggerable one shot, on the other hand, is turned ON by a trigger, and then stays ON as long as triggers keep coming in at a specified rate. The output is a smooth ON or OFF.

The reader should convince himself that this system does indeed take into account all eventualities, and behaves exactly like a standard "low-note rule" keyboard with retriggering capabilities. By sketching out some

truth tables, you can get a clearer idea of the mode of operation.

Once the basic operating system is understood, it is a simple matter to put the scheme into practice using standard CMOS logic circuits. CMOS is chosen here since it doesn't impose noise or spikes on the supply lines and has a low-current drain.

Figures 10-2, 10-3 and 10-4 shows the complete schematic, while Fig. 10-5 is the parts list. The scanning circuitry is presented in Fig. 10-2. As mentioned, a standard single-bus keyboard is modified by cutting the bus bar into five sections, each section being eight keys long, except for the last which is five keys long. Interconnections are then made between the key switches as detailed in the schematic, and the result is a matrix keyboard.

Rows are selected one at a time by IC7, which is a one-of-eight switch. Likewise, columns are selected by IC8. IC6, which is a 4024 ripple counter, is the system counter. Pin 1 of this chip is driven by the system clock composed of NOR gates G1, G2, and G3 (each gate being one-quarter of a 4001). To best illustrate the operation, suppose the key at row two and column three is depressed. The system clock quickly causes IC6 to count up to the desired point. Pin 3 of IC7 is coupled to pin 14 (row two selected) and this is then sent via the matrix to pin 15 of IC8 (column three selected). Since pin 15 has been addressed by the data inputs, it is coupled to pin 3, which is at ground. The net effect is that pin 3 of IC7 has been pulled to ground, causing the output of inverter G4 to go high. A strobe is thus created. This strobe signals the rest of the circuitry that the number currently on the data lines is the number corresponding to the key depressed. These data output along with the strobe signal then go on to the rest of the circuitry in Fig. 10-3.

Before leaving Fig. 10-2, note that the strobe signal is also coupled to the reset line, pin 2, of counter IC6. As mentioned in the analysis above, it is necessary to reset the counter whenever a key has been found depressed. However, the reset should be delayed just a bit so that the latches have time to settle down. XOR gate X7 (1/4 of IC4, which is a 4070) is configured, along with R20 and C5 as a delay circuit, sometimes called a half-monostable. The delay time is on the order of half a millisecond.

Figure 10-3 contains the schematic for the latches, trigger and gate circuitry. The data enters at the bottom left of the drawing and is sent to the latches (IC9 and IC10) and the six-bit comparator composed of X1 through X6 (each 1/4 of IC3 and IC4, a 4070) and IC11. There is nothing fancy about the latches; they simply latch data when the two pin 5's are signaled by the trigger (to be discussed shortly). The latch outputs, L0 through L5, then go to Fig. 10-4.

The six-bit magnitude comparator, as mentioned above, is composed of various gates. Actually, integrated comparators exist (such as the 4063), but are relatively difficult to find on the open market. There really isn't much savings to be had by using these "all-in-one" magnitude comparators, however. If the latched data is different from the input data, at least

one of X1 through X6 will go high. This one high state (or up to six) at the input of IC11, an eight-input NOR gate, will cause the output of IC11 to go low. Since the logic of IC11 is "inverted," DeMorgan's Rule will have G5, (which is actually a NOR gate), act as an AND gate. To complete DeMorgan's Rule, the other input of G5 should also be inverted. This is done to the strobe signal by G6. The output of G5 tells the latches when to latch.

The trigger signal is derived from the latch signal by NORing it with the differentiated GATE signal (to be discussed shortly). This implies that a trigger will occur whenever a lower note has been found or whenever one goes from a no key depressed to a key depressed condition. The NOR function is performed by G7 and G8, which are wired in parallel for increased drive. This in turn fires the one-shot composed of IC15 and its associated circuitry. The trigger signal is +5V in magnitude and approximately one millisecond wide.

As described in the analysis above, the strobe signal is converted to a gate signal via a retriggerable one-shot. IC16 and IC17, both 555 timers, perform this function. Essentially, IC16 is never allowed to time out as long as strobe pulses are present at IC17. Note that C12 is the timing cap. and can be discharged in parallel by pin 7 of either timer. The configuration is quite unusual, so the reader is referred to Walter G. Jung's, *IC Timer Cookbook* (Indianapolis, Indiana: Howard W. Sams and Company, 1978), pp. 96-97, for more information on this circuit.

The remainder of the circuit is presented in Fig. 10-4. The latch data is sent to a binary adder composed of IC12 and IC13. By setting S1 to positions 1, 2, or 3, a numerical offset of 0, 12 or 24 is added onto the data. This, of course, allows for precise octave transposition. Thus the relatively small 37-note keyboard can be made to cover a five-octave range by switch manipulation. And of course, all tracking modules, VCOs, VCFs, etc., will automatically adjust to the new octave.

The output of the binary adder is made available to the user through buffers B0 through B5. The analog synthesizer user will probably not find these outputs useful, and may eliminate this front-panel feature. Computerists, however, will probably find these outputs handy for computer sequencer experiments.

The adder outputs also go to the digital-to-analog converter. IC18, which is a 5008, performs this function. Pin 14 is a reference voltage input, and A1 provides this voltage in a stable, low-impedance form. R12 is the volts-per-octave trim while R2 and C22 decouple this sensitive voltage control. A fancier voltage source is possible; however, this design has proven more than adequate.

The output of IC18 is a current and must be converted to a voltage. R9 and A3 (which is 1/4 of IC14, a 4136 quad op amp) perform this function. The output of A3 is then sent to a *portamento* circuit. R29 and C15 add in the necessary lag time, thus creating the *portamento* effect. Q1 and A4 form a high-input impedance buffer. Q2 is put in the feedback loop of

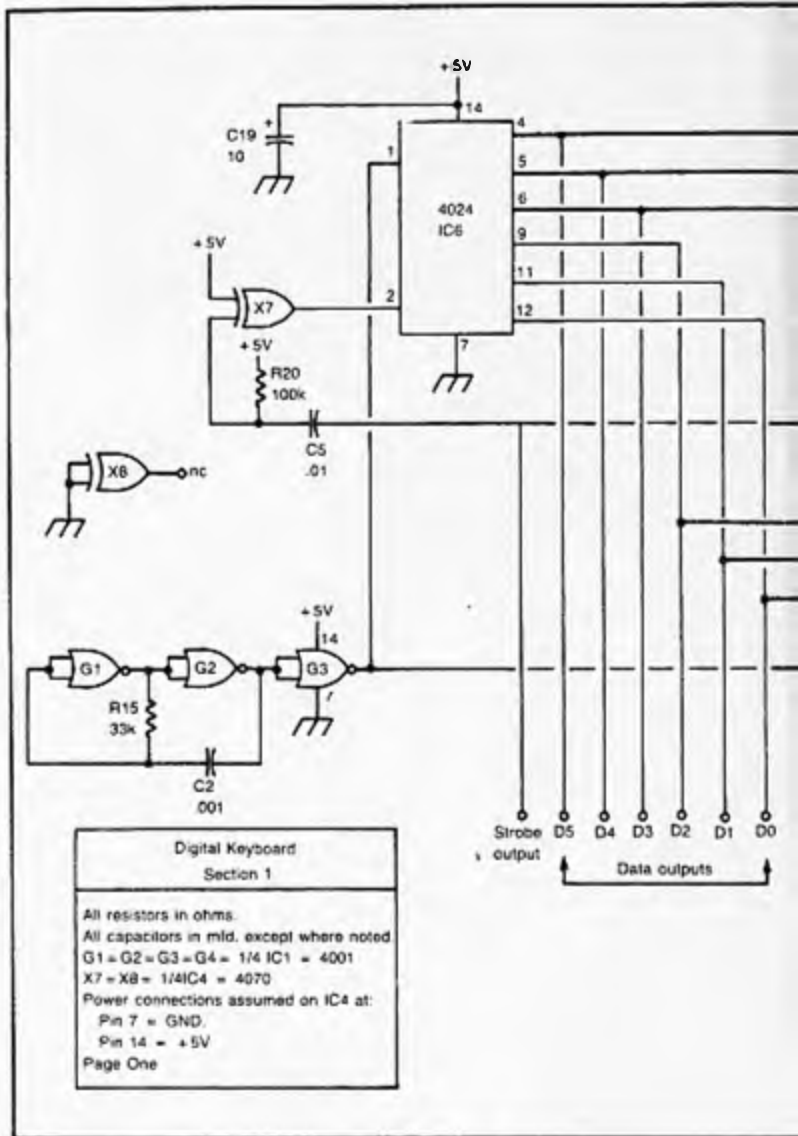
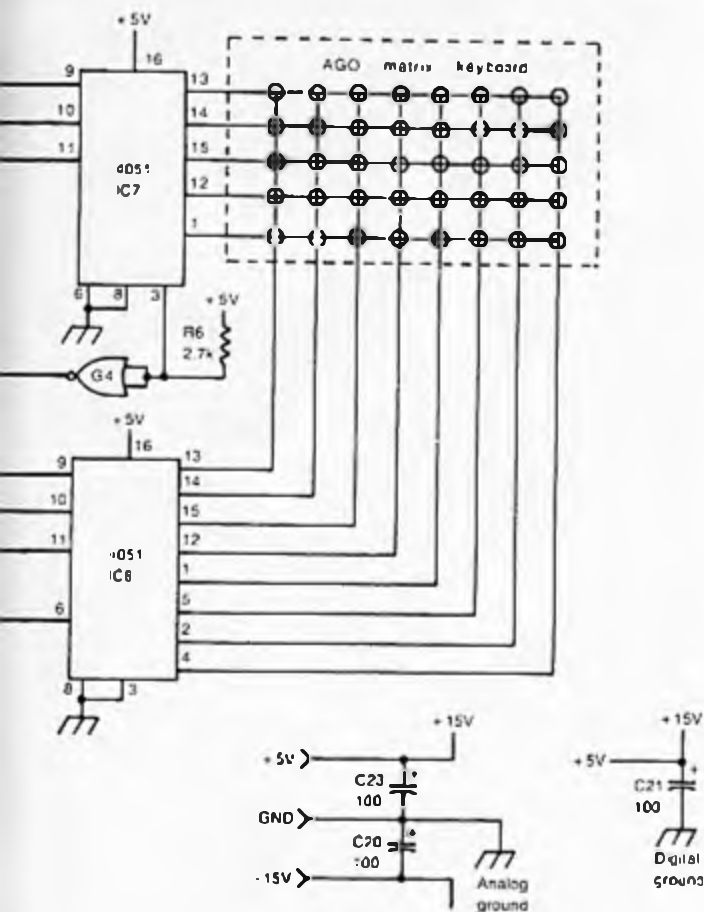


Fig 10-2 Schematic diagram of the digital keyboard (section 1).

A4 and thus cancels out any drift and offsets introduced by Q1. Essentially, an FET op amp has been created here.

A modulation input is made available at A2, along with a keyboard tuning control. This input allows the user to control the entire keyboard, thus modulating anything connected to it (via the control voltage). The tun-



ing control, R25, needs no further explanation except to mention that the prototype for the circuit used a five-turn front panel pot for this function.

This concludes the circuit description. Though it has taken quite a few words to explain the circuit action, conceptually the unit is quite simple and straightforward. The simplicity of the circuit, in part, accounts for the

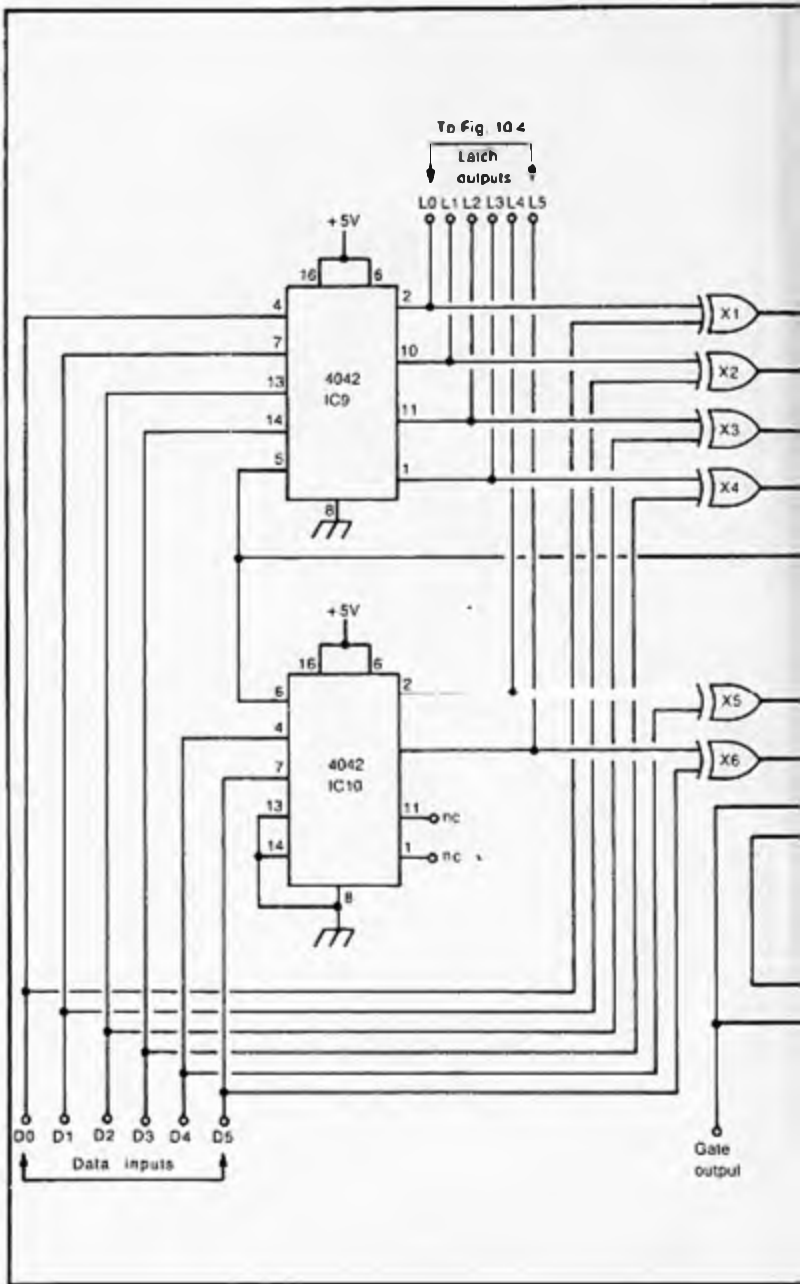


Fig. 10-3. Schematic diagram of the digital keyboard (section 2).

Digital Keyboard

Section 2

All resistors in ohms

All capacitors in mfd. except where noted

X1 - X2 - X3 - X4 = 1/4 IC3 - 4070

X5 - X6 = 1/4 IC4 - 4070

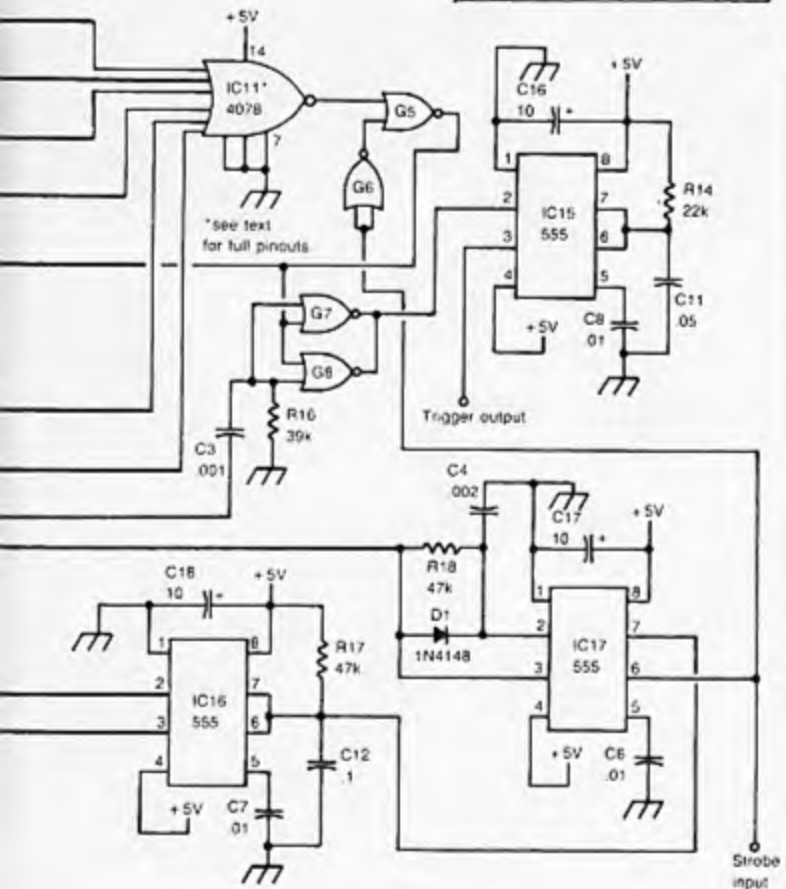
G5 - G6 - G7 - G8 = 1/4 IC2 - 4001

Power connections assumed on IC2.

IC3, and IC4 at:

Pin 7 - GND

Pin 14 = +5V



Digital Keyboard
(Section 2)

All resistors in ohms.
All capacitors in mfd. except where noted
Op amp power connections assumed at:
Pin 7 = -15V
Pin 11 = +15V
Page Three

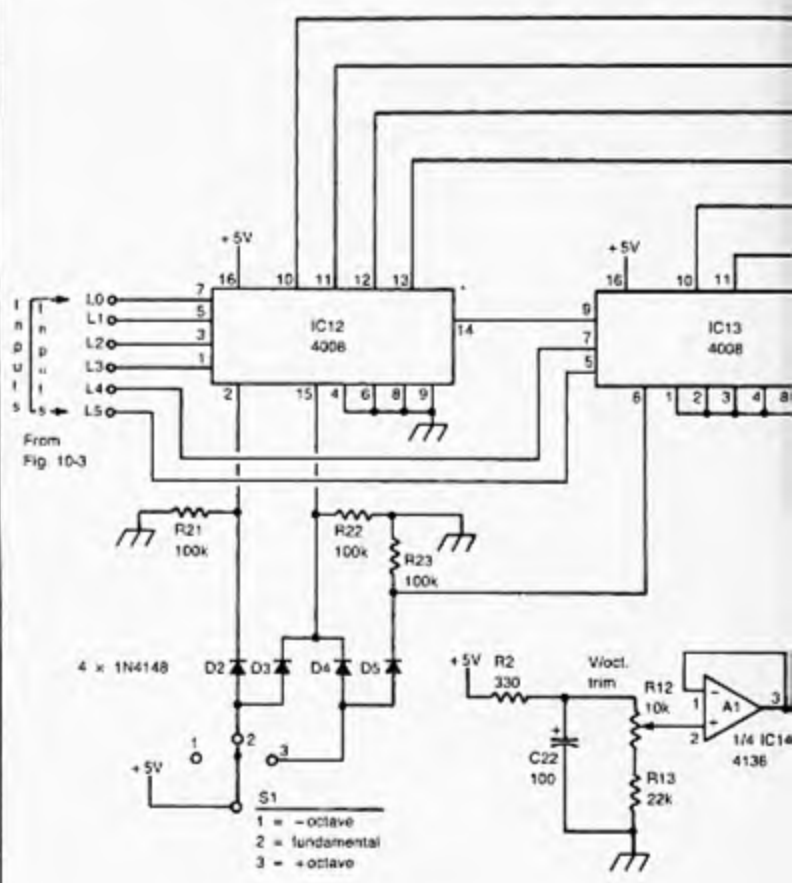
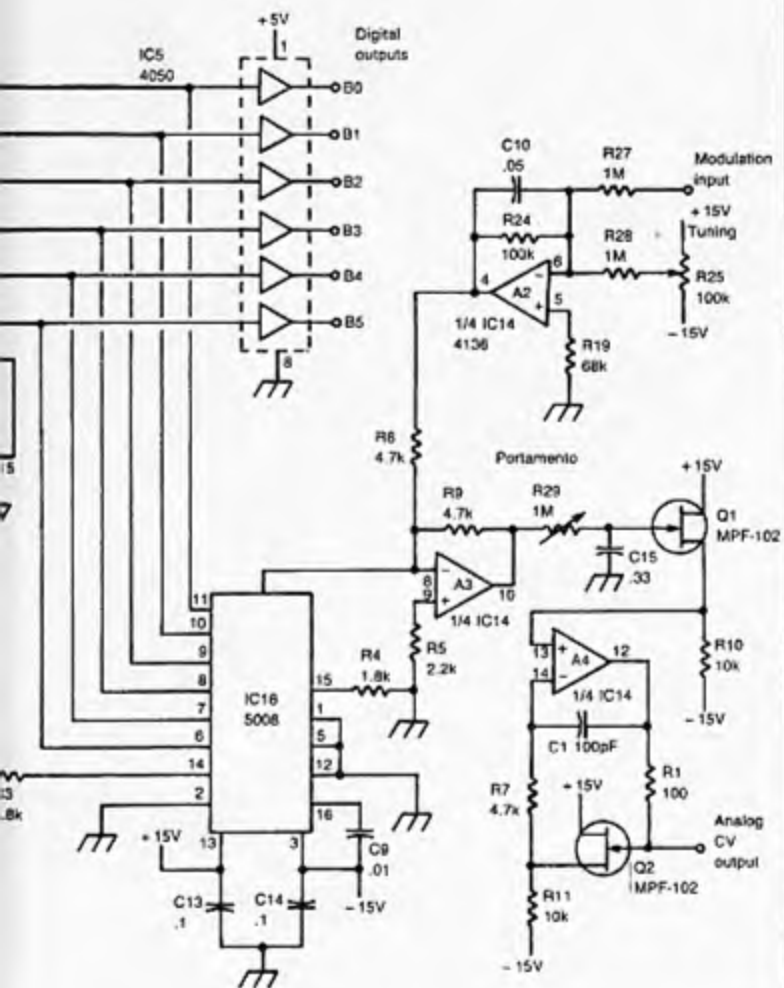


Fig. 10-4. Schematic diagram of the digital keyboard (section 3).



Resistors		Capacitors	
R1	100	C1	100 pF disc or poly
R2	330	C2, C3	.001 mfd. mylar
R3, R4	1.8k	C4	.002 mfd. mylar
R5	2.2k	C5-C9	.01 mfd. mylar
R6	2.7k	C10, C11	.05 mfd. mylar
R7-R9	4.7k	C12-C14	.1 mfd. mylar
R10, R11	10k	C15	.33 mfd. mylar
R12	10k trimmer	C16-C19	10 mfd. electrolytic
R13, R14	22k	C20-C23	100 mfd. electrolytic
R15	33k		
R16	39k		
R17, R18	47k		
R19	68k		
R20-R24	100k		
R25, R26	100k potentiometer		
R27, R28	1M		
R29	1M potentiometer		
Semiconductors			
IC1, IC2	4001 quad NOR gate (G1 through G8)		
IC3, IC4	4070 quad EXOR gate (X1 through X8)		
IC5	4050 hex buffer		
IC6	4024 seven-stage ripple counter		
IC7, IC8	4051 one of eight switch		
IC9, IC10	4042 quad latch		
IC11	4078 eight-input NOR gate		
IC12, IC13	4008 four-bit full adder		
IC14	4136 quad op amp (A1 through A4)		
IC15-IC17	555 timer		
IC18	5008 digital-to-analog converter		
D1-D5	1N4148 switching diode		
Q1, Q2	MPF-102 FET		

Fig. 10-5. Parts list for the digital keyboard.

good results which may be achieved.

CONSTRUCTION

This is one circuit that is actually easier to build without using printed circuit board techniques. Like most digital circuits, there are a few passive components, but many parallel data lines. This being the case, a printed circuit board would probably have to be double sided, and this would in-

roduce an undesired level of complexity. An easier approach is to hand wire the unit.

An op-amp prototype circuit board, such as any Radio Shack stocks, affords a good mode of construction. All wiring can thus be done by hand. Actually, there are so many ICs in this project that two of these prototype boards will have to be used. To keep things neat, ribbon cable should be used to connect the two boards, and also to connect the circuit boards to the keyboard. The keyboard proper can be a standard, 37-note, single-bus unit. Keyboards like these often show up on the surplus market, so keep your eyes open. If you can't find one this way, then try going right to the manufacturer. Pratt-Read is one of the world's largest makers of these organ-type keyboards, so you might want to order one directly from them. (See Appendix A for the address.) The entire keyboard and circuitry can be housed in a road case, and this not only makes an attractive unit but adds to the durability as well. You can either build your own case, or buy one ready made. For example, PAIA Electronics, a manufacturer of electronic music kits, stocks a sturdy and attractive road case for a 37-note keyboard. (See Appendix A.)

Three of the controls, (*portamento*, octave switching and tuning) should be mounted right on the keyboard panel, while the rest of the controls and jacks can be mounted on a rack panel at the synthesizer. A pair of D-25 plugs and a length of ribbon cable complete the connection between the keyboard and the interface panel. Note that the ribbon cable, besides carrying the control voltage, gate the trigger signals to the panel, can also pick up the power supply lines from the synthesizer and run them to the keyboard.

All of the CMOS circuitry requires a +5V supply. IC18 and IC14 (the analog circuitry) require a bipolar 15V supply. One very important point is that the digital ICs should have their own ground, and likewise the analog ICs should have their own ground. The two grounds should connect at one point only! This will prevent any possible interaction. The two grounds are clearly indicated on the schematics by using the "chassis ground" symbol for the digital ground, and the "earth ground" symbol for the analog ground.

To simplify the schematic, the pinout is not shown for IC11. The inputs are at pins 2, 3, 4, 5, 9, 10, 11, and 12. Since only six of these inputs are used, the two unused pins should be grounded. The output is at pin 13. Pin 14 is +5V and pin 7 is ground. Pins 1, 6 and 8 are no connection.

One subtle point about the trigger circuit is the fact that occasionally a trigger two milliseconds wide will occur. This is due to the "count up to the not depressed and reset" nature of the system counter. That is, if the counter happens to start at a number higher than the key depressed, it will take somewhat over one complete count cycle (counting from 0 to 36) before the correct note is latched and the trigger is sent. Since the system clock operates at a very high frequency, this all happens so fast that the human ear will never know that anything unusual has happened. In

general, this will have no effect upon the circuits driven by the keyboard trigger.

CALIBRATION

The only adjustment that needs to be made is to the volts-per-octave trimmer, R12. Calibration is easy; while monitoring the control voltage output with a good voltmeter, adjust R12 so that a one-octave jump on the keyboard causes a one-volt jump. You will find that a digital voltmeter with a high input resistance makes the most sense here, so if you need to, see if you can borrow one for the purpose of calibration. After setting the trimmer, dab a little fingernail polish on it to hold it in place.

USING THE DIGITAL KEYBOARD

If you have used analog keyboards in the past, then this one will really surprise you. It's almost impossible to cause it to "glitch," and is well suited to a broad range of playing styles. In short, the digital approach is more than justified, if for not other reason than that it provides the most reliable operation around. An added advantage is that the digital outputs are available if and when you decide to go with a computer interface. Of course, making a sequencer through a digital approach is almost trivial and very large patterns can be entered through the keyboard and played back as desired. Chapter 16 has some suggestions along these lines.

But for right now, enjoy your new keyboard. Get familiar with the controls and jacks. Remember, you have three sorts of outputs: the control voltage (typically used to control VCOs), the gate and the trigger. These last two signals, of course, can be used to fire an ADSR.

On the input end of things, you can offset the entire keyboard with the master tuning control, R25, or modulate it with the input at R27. Finally, you can add in *portamento* for slurred effects.

Chapter 11

The Potpourri Module

Here's a rather unusual module for your synthesizer that's guaranteed to add greater versatility at a reasonable cost. It's called the Potpourri module because it adds six new functions all at once. These new functions are a lag processor, a buffer, a control voltage inverter, an audio inverter, a two-in one-out mixer and a comparator. You'll see some ways to use the unit a little later, but for right now let's dive in and see how it works.

HOW THE CIRCUIT WORKS

The whole Potpourri module is configured around a TL084 quad op-amp package. The amplifiers in this IC are FET input types, meaning (among other things) that they have a very high input impedance. This high-input impedance is exploited in the lag processor, so I'll consider it first. Refer to the schematic in Fig. 11-1. (The parts list is in Fig. 11-2). The lag-processor input is at J1. The signal is buffered by A1, which is set up as a voltage follower and because of R6, the input impedance here is 100k. The output of A1 feeds to capacitor C1, but only after passing through R13, the lag time control. There are several different ways to think about this structure; it can be considered to be a lowpass filter with an extremely low cutoff frequency, or you can think in terms of the capacitor charging at a certain time constant. Either way the result is the same. Any input voltage event will be slowed down, and that is the purpose of a lag processor. R13 allows selection of variable lag times.

C1 must be buffered well and op-amp A2 does this job. Now you can see why the TL084 was selected for this module. The very high impedance of the noninverting input of A2 guarantees that the charge on C1 will remain uncorrupted.

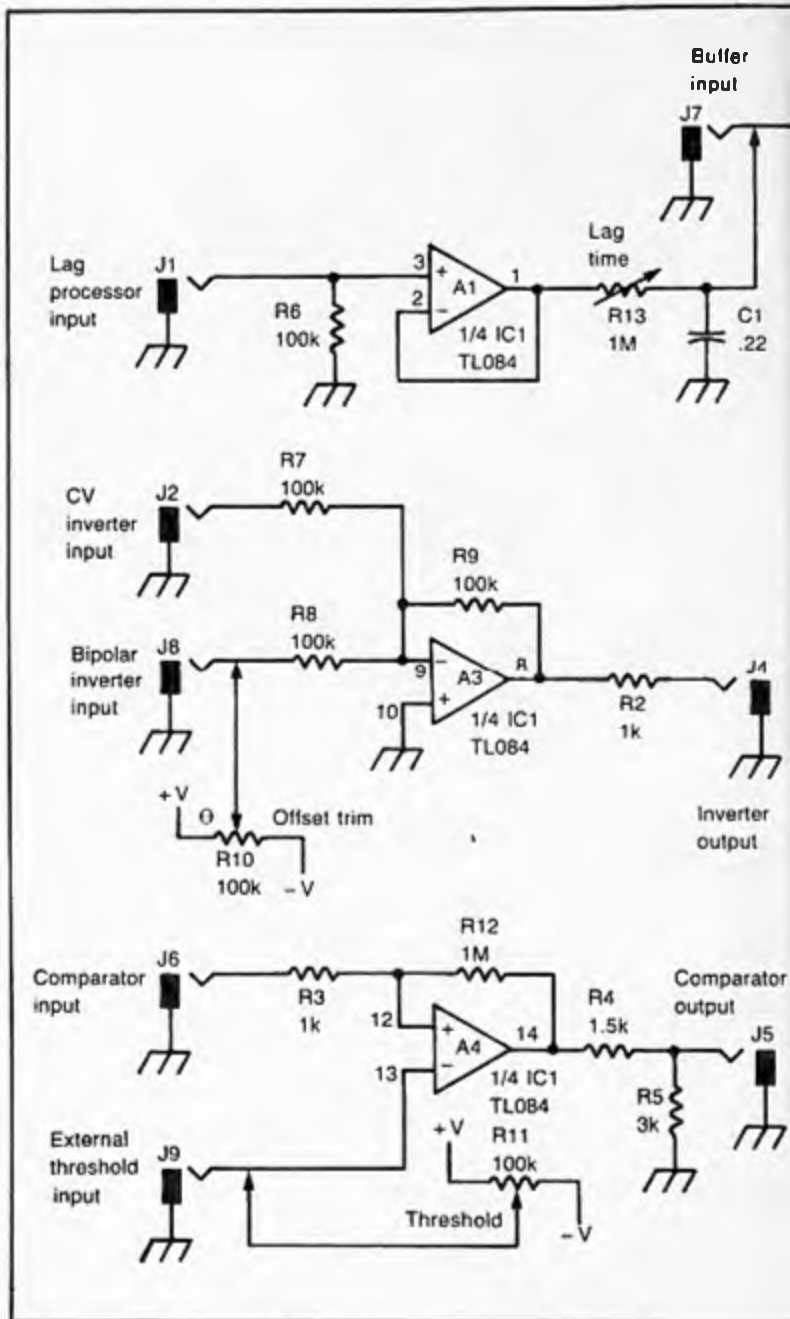
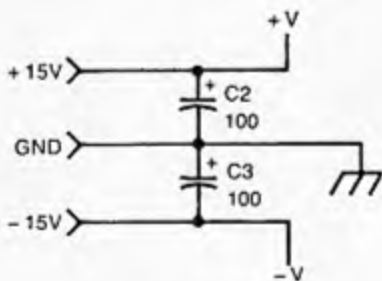
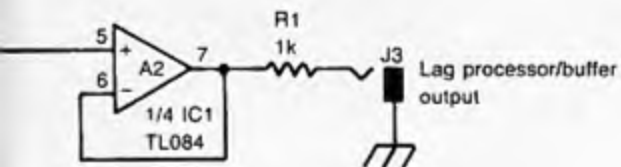


Fig. 11-1. Schematic diagram for the potpourri module.



Potpourri Module

All resistors in ohms.

All capacitors in mfd.

Power supply connections to op amps
assumed at:

IC1: Pin 4 = +15 V

Pin 11 = -15 V

Resistors		Capacitors	
R1-R3	1k	C1	.22 mfd. mylar
R4	1.5k	C2,C3	100 mfd. electrolytic
R5	3k		
R6-R9	100k		
R10	100k trimmer	Semiconductors	
R11	100k potentiometer	IC1	TL084 quad op amp
R12	1M		
R13	1M potentiometer	Miscellaneous	
		J1-J6	open circuit phone jack
		J7-J9	closed circuit phone jack

Fig. 11-2. Parts list for the potpourri module.

Jack J7 is a switching jack. When nothing is connected to it, A2 buffers C1, as mentioned above. However, by inserting a jack into J7, the lag processor network is isolated from A2 and you can now use it as a general-purpose buffer. Hence, you get a buffer function for the price of a single jack!

A3 and its associated circuitry form the two inverter options. Consider the input at J8 first. This is the audio or bipolar inverter input. When a plug is inserted into this jack, R10 is isolated from the circuit. The signal is sent to A3 by way of R8 and since the feedback resistor, R9, is equal to R8, the gain of the amplifier is unity. However, note that the signal has been inverted in the process. Thus the circuit has acted as a bipolar inverter. Whatever was +5V is now -5V and vice versa.

What if jack J2 is used instead? To answer that, suppose an envelope signal with a 0 to +5V swing is introduced into this jack. Trimmer R10, which has been tweaked to produce a -5V bias is now connected to the circuit. This voltage will then add in with any voltage present at J8. Suppose J8 is at 0V. Zero volts added to the -5V bias is -5V, which when inverted by A3 yields +5V. Now suppose J2 is at +5V. Plus 5V added on to -5V is 0V. As you can see, then, J2 can take a control voltage of 0V to +5V and convert it to a voltage with a swing of +5V to 0V.

Now suppose that both J2 and J8 have plugs inserted in them. In this case, the two signals are summed together and J4 yields the mixed result. That takes care of the mixer function!

Look at A4 and the surrounding circuitry. This part of the op-amp package is set up as a comparator. R3 is the input resistor, while R12 adds some positive feedback or hysteresis. Hysteresis of about .1% (1k divided by 1M) helps give the comparator a little "snap." The output of A4 will swing from about -15V to +15V, which is a trifle stout for a typical synthesizer system. R4 and R5, then, chop this signal down to a standard value of -5V to +5V, or 10V peak-to-peak. Not only do they attenuate this voltage somewhat, but they also create an output impedance of 1k, which is another synthesizer standard.

The comparator input is at J6. A signal present here will be compared to the reference voltage at the inverting input of A4. The reference voltage can be generated in two ways. If no plug is inserted into J9, then the voltage comes from R11, which is a front panel control. Otherwise an external reference voltage can be presented by way of J9, which is a switching jack.

Having described the individual functions of the Potpourri module I can now make a few general comments. In keeping with modern synthesizer standards, note that all signal levels are 10V peak-to-peak, while all control voltage levels are 0V to +5V. All input impedances are 100k, while all output impedances are 1k. The power supply is a plus and minus 15V. Thus, the circuit is suitable for use with any of the other modules described in this book.

CALIBRATION

Calibrating the Potpourri module is a snap. Simply ground input J2, and while monitoring J4 with a voltmeter, adjust R10 so that the output voltage is +15V. That's all there is to it!

USING THE POTPOURRI MODULE

There are obviously hundreds of ways to use these new functions. Here are just a few. A good patch for the lag processor is to split an ADSR signal so that it feeds a voltage-controlled bandpass filter and the lag processor. The output of the lag processor then controls another voltage-controlled bandpass filter. Now when the envelope is fired you will hear two filter sweeps, one lagging behind the other by the amount of time set by R13.

And here's an idea for the inverter. As you did before, split the ADSR signal and have one side of the split feed the VCF. Now take the other side of the split and feed it to the control-voltage inverter. J2. Take the output, at J4, and feed it to another VCF. Now when the ADSR is fired you will hear one filter sweep upward, while the other sweeps downward. This is a very nice sound, and is especially interesting at that point where the two filters pass each other since all sorts of phase cancellations and additions occur.

The comparator has many uses, especially in the area of percussion synthesis. But perhaps the most interesting application is that of phase-modulated sync. This is a fairly detailed patch, so you are referred to the article, "The Use of Sophisticated Modifications in Creative Synthesis," by Ken Perrin, (*Polyphony*, Sept./Oct., 1979, pp. 27-31), which describes phase-modulated sync in some detail.

So there you are! Six additional functions for your synthesizer at a very low cost. While the Potpourri module is perhaps not the most exciting module in a system, it is probably one of the real "workhorses." Give it a try and you're sure to find that it makes possible the creation of very subtle and complex sounds.

Chapter 12

A Quadrature Function Generator

What's a quadrature function generator? To answer that you must first be familiar with a quadrature oscillator. An oscillator is said to have *quadrature* outputs if it is capable of producing sine, cosine, negative sine and negative cosine waves simultaneously. Such an oscillator can be created quite easily by setting a four-pole lowpass filter resonating and tapping the required outputs off of the four consecutive stages. (For a detailed explanation of this, see J. Patchell, "Build a Voltage-Controlled Quadrature Oscillator," *Polyphony*, Nov./Dec. 1980, pp. 26-27. Refer to Appendix B for the address of *Polyphony*.) If trigonometry alarms you, just consider the outputs to be four sine waves, each one ninety degrees out of phase with the previous one.

Even though many interesting patches are possible with this type of oscillator, it does have one main drawback. Sine-wave oscillators are fairly touchy and oftentimes you will find that as the frequency is swept over a broad range, the amplitude will change as well. In addition, under some conditions the oscillator may fail to oscillate, or on the other extreme, may go into hard clipping and produce some undesirable distortion.

That's where the quadrature function generator comes in. You can throw out the oscillator completely and replace it with a function generator. Oscillators are reactive; they depend on a resonating RC network. Function generators are nonreactive (in the sense that they don't resonate); they depend only on the charge and discharge of a capacitor. As you will find if you build this circuit, the outputs are very stable in amplitude and purity over a very wide range. Finally, one more distinction between oscillators and function generators is that the former outputs sinusoids, while the latter has triangle waves (or sometimes ramp waves) for outputs. Fig-

ure 12-1 shows the four outputs possible with this circuit.

Since this is an unusual circuit (no commercial synthesizers have them), we probably should take a quick look at what it may be used for. The quadrature function generator produces four triangle waves, each one ninety degrees out of phase with the previous one. That suggests a good application for quadraphonic systems. Imagine you are sitting in the middle of four speakers, one to the north, one to the south, and a pair east and west of you. Now imagine that a sound going to each speaker is being controlled by a separate VCA, and each VCA is being controlled by an output of the quadrature function generator. As the north speaker is decreasing in volume, the east one is increasing. Then the east speaker decreases and the south one increases, and so on. You've just created circular sound! The sound will go around and around, and the psychological illusion is quite convincing; you will either feel like the sound is moving around or that you are. This is an example of a little explored area in electronic music known as *location modulation* (LM). In LM, you use various circuits to give the illusion that sound is moving in some fashion.

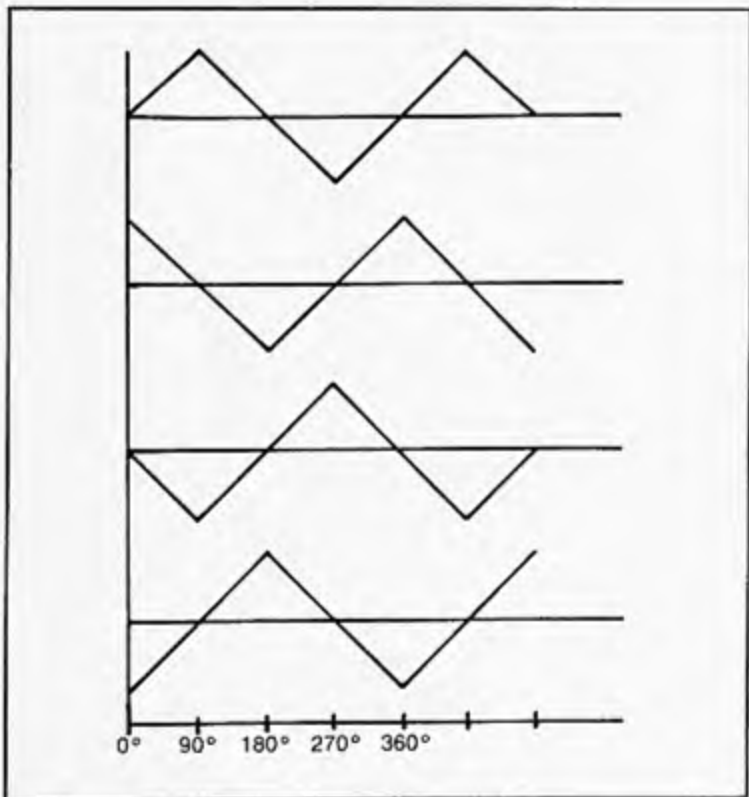


Fig. 12-1. The four outputs of a quadrature function generator.

A little later in the chapter, you'll see some other uses for the circuit, but this is enough to get you going.

HOW THE CIRCUIT WORKS

Now that you know the basics of what a quadrature function generator does, let's see how it works. To fully understand the circuit required some mathematics which unfortunately is beyond the scope of this book. If you're interested in the full mathematical derivation, though, refer to my article, "A Function Generator With 'Quadrature' Triangle Outputs," *Electronotes* #122, pp. 13-20, which gives a rigorous treatment of the circuit. (See Appendix B.) Actually, the version presented here is more compact and uses fewer parts than the one described in the article, but the circuit action is very similar.

Even though I can't do the mathematics here, you can get an intuitive feel for how the circuit works. Stop and think about triangle waves. A triangle is made up of two line segments, one going "up" and one coming "down." To build another triangle wave ninety degrees out of phase with the original, you need only to pick up parts of these two line segments and invert, level shift and mix them in the proper order and at the proper time. To drive this point home, sketch two triangle waves, with a ninety-degree phase difference, and try to deduce how one can be derived from the other just by moving the segments around. This basic concept is quite easy to understand, even though the mathematics may get a little complex in a formal derivation.

This, then, is really all you need to know to get a feel for how the circuit works. Just remember, line segments are picked up and rearranged at the proper moments, and result in the desired waveform.

Now refer to the schematic in Fig. 12-2. Op-amps A1 and A2 form a Schmidt trigger-integrator function generator, a very standard application for op amps. C3 sets the basic frequency range, with R21 allowing for an adjustable frequency output. It is important to note that, like all Schmidt trigger-integrator function generators, the output of A1 is a triangle wave and the output of A2 is a square wave. The triangle wave is sent directly to the Primary Triangle Output at J1. The output here is 10V peak-to-peak in amplitude, and so may be used for any ordinary LFO application.

Recall that various line segments of the triangle wave will be used to construct a new one ninety degrees out of phase with the first. So besides going to J1, the triangle also feeds to other circuitry via A3 and A4. Now, it's important that the circuit knows when to grab the various segments of the triangle wave and that's where the square-wave output of A2 comes in. This output provides the timing information for the rest of the circuit.

IC3 is configured as an XOR (exclusive-OR) gate. It may seem strange to use an entire package of NOR gates to accomplish this function, but actually it's more cost effective to use the inexpensive 4001 for this, rather than using one gate out of a quad XOR package. Note the unusual power

supply hookup for the chip. Since this part of the circuit drives the gate of an n-channel FET, you need a swing from -15V to ground.

A4, Q1 and associated circuitry comprise a sign changer. A sign changer is a two-input, one-output device. One input is a control input and the other is the signal input. When the control input (the gate of Q1, in this case) is at ground, the signal input (to R13 and R14) is passed unaltered. But if the gate is at -15V , then the signal is sent to the output inverted. It is the duty of the sign changer to invert segments as required to create the new triangle wave.

Note that so far all of the op amps have each been one-quarter of a TL084 quad FET op-amp package. This particular chip must be used (in lieu of 741s, for example), since an extremely high slew rate is needed. You want all of the switching to be as clean as possible and this will insure that when the segments of the new triangle wave are "glued" back together they will connect in a smooth fashion.

Now I'm going to reverse this philosophy and specify as slow an op amp as possible for A5 and A6! A5 sums together the various line segments to form a new triangle wave, ninety degrees out of phase with the original. I suggest a low slew rate for this amp, since any of the discontinuities in the derived triangle will be masked by the op amp's inability to slew fast enough! In addition, A6 is set up as a lowpass filter, and this will help smooth the new triangle out as well. The cutoff frequency of this two-pole filter is at 1.5 kHz.

Admittedly, without getting into a rigorous treatment of the circuit, it is difficult to believe that it really works. But the hints given above should aid you in developing an intuitive feel for how the quadrature function generator is implemented.

CONSTRUCTION

Any method of construction is fine for this project because no audio or radio frequencies are involved. Pick whatever method you want. As usual, hang the completed circuit board behind the front panel on small angles and use #4 hardware to affix in place. This will minimize the length of wires running between the circuit board and the panel. Figure 12-3 shows the complete parts list for this project.

To round out the circuit you will probably want to provide inverted versions of the "Primary" and "Derived" triangle outputs. This will give you a total of four outputs, each one ninety degrees out of phase with the previous.

If you like flashing lights on your equipment, you might want to add some LED displays to the various outputs. For a really colorful light show, use four tri-colored LEDs!

CALIBRATION

Adjusting the trim pots is fairly easy. To simplify the process, tem-

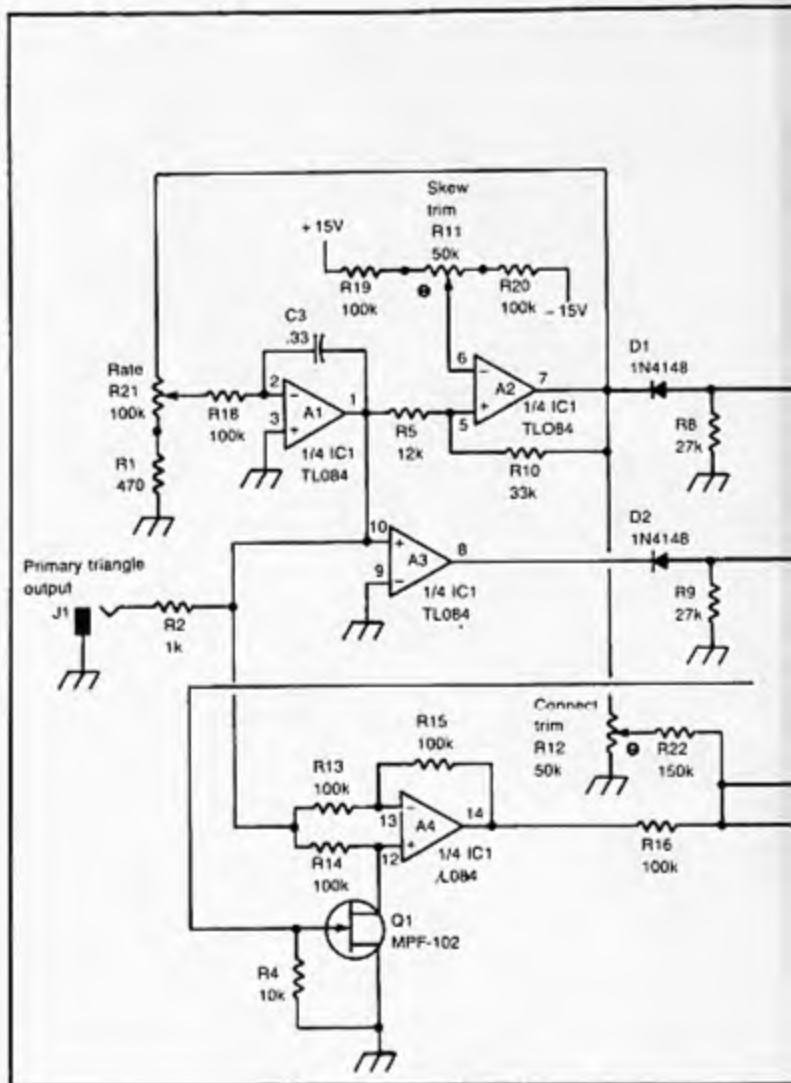


Fig. 12-2. Schematic diagram for the quadrature function generator.

porarily replace C3 with a 0.05 mfd. capacitor. This will move the frequency of the function generator up into a range which is easier to observe. Now monitor the "Derived Triangle Output" at J2 on an oscilloscope. While watching the waveform, go back and forth between trimmers R11 and R12 until the waveform "comes together" and connects to form a smooth triangle. The process seems almost magical; the sight on the 'scope will really amaze you! If you have a dual-channel 'scope, compare the two waveforms

Quadrature Function Generator

All resistors in ohms.

All capacitors in mfd.

Power supply connections assumed on op amps:

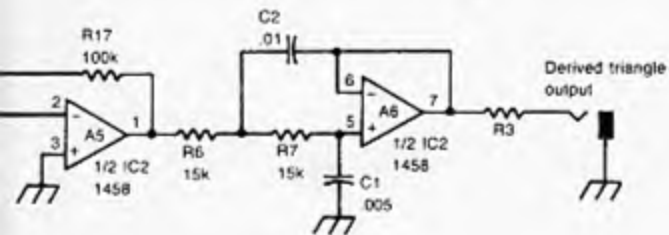
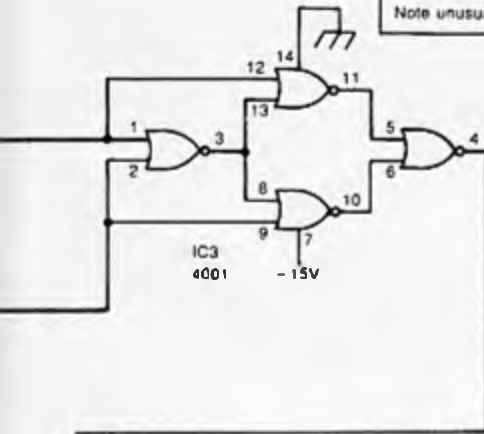
IC1: Pin 4 = +15V

Pin 11 = -15V

IC2: Pin 4 = -15V

Pin 8 = +15V

Note unusual power connections to IC3.



("Primary" versus "Derived") and confirm that they are indeed ninety degrees out of phase.

USING THE QUADRATURE FUNCTION GENERATOR

Now what shall we use the quadrature function generator for? How about, for starters, automatic quadraphonic panning? This was described

Resistors		Capacitors	
R1	470	C1	.005 mfd. mylar
R2, R3	1k	C2	.01 mfd. mylar
R4	10k	C3	.33 mfd. mylar
R5	12k		
R6, R7	15k	Semiconductors	
R8, R9	27k	IC1	TL084 quad op amp
R10	33k	IC2	1458 dual op amp
R11, R12	50k trimmer	IC3	4001 quad NOR gate
R13-R20	100k	D1, D2	1N4148 diode
R21	100k potentiometer	Q1	MPF-102 FET
R22	150k		

Fig. 12-3. Parts list for the quadrature function generator.

at the start of the chapter; simply gang the inputs of four VCAs together, and control each one by a different triangle wave. Send each output of the VCAs to a different channel in your quad system. The result is circular location modulation.

What if you don't have a quad system? Don't worry, you can use the

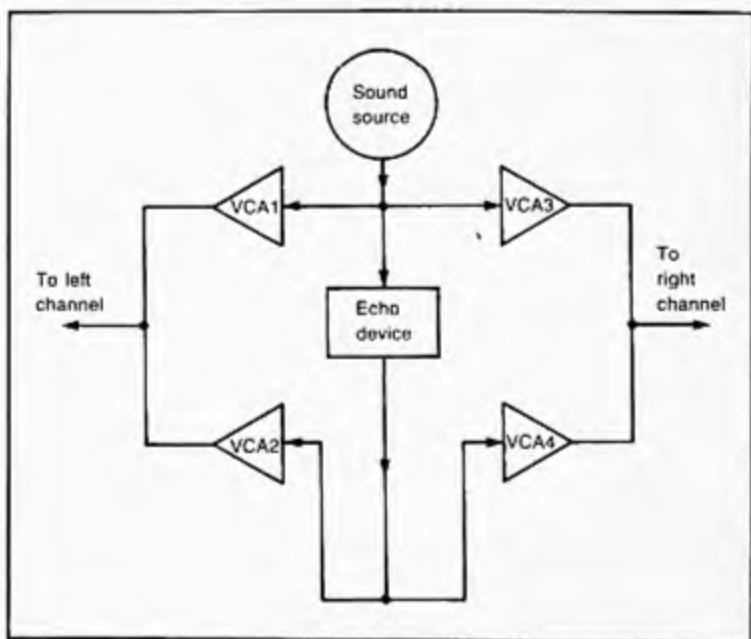


Fig. 12-4. Each VCA is controlled by a different output of the quadrature function generator.

quadrature function generator for some neat stereo effects. For a really bizarre sound, try the patch in Fig. 12-4. Gang the inputs of VCAs 1 and 3. Apply a dry signal to these VCAs. VCA1 should feed the left and VCA3 the right channel. Now gang the inputs of VCA2 and VCA4. Apply an echoed signal to these inputs. VCA 2 should mix into the left channel, while VCA 4 mixes into the right. Now start the quadrature generator turning and crank up the delay time and feedback of the echo unit. Hit some staccato notes. The sound should be quite convoluted, and may actually make some listeners seasick!

If you don't have a stereo rig either, then it's harder to think up patches, but try this one. Apply an audio signal to four different flangers, each one set for a slightly different initial delay time. Then send each flanger output to a different VCA, and finally sum the VCAs to a monaural output. Set the function generator going; you should hear an extremely dense and lush sound. This is especially good for waveforms with lots of harmonics—pulse waves in particular.

If you don't have a monaural system (or even a synthesizer), it's still possible to have fun with this circuit! You can create some great Lissajous figures on an oscilloscope screen, and the circuit could very easily serve as the basis for an electronic art contraption.

Chapter 13

A Balanced Modulator

The balanced modulator, as I explained in Chapter 2, is responsible for creating complex timbres out of relatively simple ones. Recall that the balanced modulator has two audio inputs and one output. If sine waves are pumped into the inputs, then the output will consist of new sine waves. To be more explicit, the output will consist of two frequencies: the sum of the input frequencies and the difference of them. These new frequencies are called *sidebands* and in general are not related to the inputs in any harmonic manner. Thus, a balanced modulator is used to create the so-called "clangorous" sounds, sounds that approximate the tinkle, crash or clang of metal pieces smashing together.

Early electronic music was characterized by lots of balanced-modulator effects. Because the sound is disconcerting when overused, many innocent listeners drew erroneous conclusions concerning the nature and potential of electronic music. Just like any module, when overused or not used in an artistic fashion, the sounds thus created tend to cloy. So be forewarned! Your system should have a balanced modulator, but use it in a meaningful way.

Incidentally, besides clanging noises, with the proper adjustment of the controls a balanced modulator may used be to create a fairly convincing calliope! You'll see how a little later.

As mentioned earlier, the first balanced modulators were built with transformers and a ring of diodes. Of course, such a unit is essentially passive and consequently suffers from signal loss. Other drawbacks as well make the old-fashioned ring modulator approach undesirable.

A little later, in the early 'seventies, a balanced-modulator IC became available. It had the generic number 1595 and tended to show up in many

commercial and home-built designs. While active in nature (and hence immune to loss of signal) and definitely a big improvement over the ring of diodes approach, it tended to be a little sensitive and touchy to adjust. Since that time, the 1595 has almost dropped out of sight, and is now rarely seen on the open market.

The design approach in this chapter is to take a standard 3080, operation transconductance amplifier (OTA) and configure it as a balanced modulator. Normally an OTA is a *two-quadrant multiplier*, meaning that one input may be bipolar in nature while the other must be unipolar. Of course, both inputs of a true, balanced modulator must be bipolar, so if an OTA is to be used, some sort of adjustment must be made to the unipolar input. As it turns out this is quite easy to do.

There are two real advantages to using the 3080 in this design. First, unlike the 1595 which is quite difficult to get, the 3080 is available from lots of dealers. And the price is around \$1, putting it well within everyone's reach. Secondly, the 3080 actually simplifies the design of a balanced modulator somewhat, making the resulting circuit easy to build and very reliable as well.

As you might suspect, this application for the 3080 is quite nonstandard. You might wonder where the original idea came from. Well, the idea has been around for some time and seems to have appeared originally in an application note for the LM13600, another OTA made by National Semiconductor. The idea lay fallow, until Bernie Hutchins, the editor of *Electronotes*, ran across it and realized that the circuit is well suited for electronic music applications. You can see his treatment of a similar circuit in *Electronotes* #107, pp. 13-15. (See Appendix B for the newsletter's address.)

A full understanding of the circuit requires quite a bit of time spent in setting up equations based upon Ohm's Law and Kirchoff's Laws, then eliminating equations as needed. The mathematics required is fairly simple, but the messiness of the equations puts it beyond the scope of this book. If you would like to see the equations (and they are quite interesting for those with a mathematical bent) then refer to the above article for more information. For the purposes of this book, a more intuitive treatment is sufficient.

A number of features have been added to the original circuit to make it more reliable and easier to use. In particular, impedance levels and voltages have been firmed up to make the unit compatible with the rest of the circuits in this book, and several input features have been added to make more complex patches easily obtainable.

Before looking at the circuit, one small point should be clarified. Sometimes this circuit is referred to as a *balanced* or *four-quadrant multiplier*. If you are thinking in terms of dc levels, it's easy to see why the term "multiplier" is used. For example, put +2V on one input and -3V on the other, and the output will show -6V (scaled down by some multiplicative constant). Thus, it actually multiplies voltages! You can think of a balanced modulator as multiplying voltage levels, or creating sum and difference

frequencies. Either interpretation makes sense, and you'll see both as the need arises.

HOW THE CIRCUIT WORKS

As mentioned, a full understanding of the circuit requires a certain amount of mathematical sophistication. I'll skip the mathematics and try to give you a feel for how the circuit works in general terms.

Refer to the schematic in Fig. 13-1. The heart of the whole circuit is the 3080 OTA. This is very similar to other op amps, except for the inclusion of a special gain control pin. This is pin 5. By varying the amount of current which is withdrawn from this pin, the gain of an amplifier configured around the 3080 may be changed. Now note that the input to this amplifier is a differential stage and expects to see a voltage. The output is a current, and so must be converted to a voltage if you are to put the chip to good use. And as mentioned previously, control pin 5 expects to see a current as well. This can be summarized by saying that the 3080 is a voltage-in, current-out, current-controlled amplifier.

Keeping the 3080 in mind for a few minutes, let's backtrack to the input structure. Any balanced modulator has two inputs; for convenience we'll call these the x-input and the y-input. The x-input in this circuit is at jack J2 and the y-input is at J1. Note that we can have either ac or dc coupling by simply throwing a switch (S1 or S2). For most audio applications you will want to have ac coupling and in this way you can forget about any possible dc offsets. For some applications, though, you might want direct or dc coupling; in this case simply close the desired switch.

As with all modules in the synthesizer, attenuators are provided on the inputs (R15 and R17). These, of course, make it easier to deal with mixed signals which may exceed the required 10V peak-to-peak and additionally allow you to set the amount of modulation desired on either input.

Normally you will want the unit to be fully balanced, and by this it is meant that you won't want either of the original input signals to leak through. But some sounds, the calliope most notably, do have some of the original frequencies bleed through. For times when you need this intentional "bleed-through," control R16 allows you to offset or unbalance the input. When you don't want this effect, just turn the control down.

The x and y signals are multiplied and the product is sent to the output, pin 6 of IC2. As mentioned, this is a current output and should be converted to a voltage. Op-amp A3 accomplishes this and also scales the voltage as well. If you stop to think about it, the output must be scaled. Suppose you have a +5V dc level at the x-input and a +5V dc signal at the y-input as well. After the two signals are multiplied by the circuit, theoretically the output should be +25V! Of course, since this is a 15V power supply, it's impossible for the output to swing this far, and the result will be a clipped waveform. But if the output is scaled or divided by a constant, this can be avoided. Taking into consideration the input to 3080 and the effects

of A3, the final output of this circuit is scaled down by a factor of 5. This puts the output into a range suitable for the rest of the synthesizer.

CONSTRUCTION

This circuit can easily be built on a printed circuit breadboard affair. Alternately, perfboard and flea-clip methods are suitable as well. In either case, be sure to use shielded cable for the audio inputs if the connections from the circuit board to the front panel run more than a few inches. And as usual, beware of ground loops. When using a shielded cable, be sure to ground only one side of it.

Be tidy with the layout of the circuit. IC1 and IC3 are FET op amps and can give you troubles with supersonic oscillation if you aren't neat in your wiring. Avoid the "rat's nest" approach and you'll do just fine.

Don't forget C3 and C4. These decouple the circuit from other modules in your unit and prevent undesired interaction through the power supply lines.

Figure 13-2 shows the parts list for this project. As you can see, the parts are all easy to find and fairly inexpensive. In fact, this design is probably the least expensive balanced modulator around and yet has quite a good sound.

CALIBRATION

Unlike some balanced modulators, this one is quite easy to calibrate and will tend to stay fixed once set. Apply a 1 kHz tone of 10V amplitude peak-to-peak to the y-input. (Use your VCO for this.) Now ground the x-input at J2. Monitor the output with an oscilloscope or a sensitive ac voltmeter and adjust the y-trim, R18, for minimum deflection at the output.

Now apply the signal to the x-input and ground the y-input. Adjust the x-trim, R4, for minimum deflection. Repeat the calibration several times until you are satisfied with the results. By the way, if the y-trim seems to be having negligible affect, try replacing the 3080 with another. It seems that not all 3080s have the same offset characteristics. And don't expect too much; the carrier and signal rejection will never be much below -40 dB. This figure, while rather poor for laboratory work, is nonetheless still quite good for synthesizer applications.

USING THE BALANCED MODULATOR

Your first experiments with this module should be conducted with sine waves for inputs. In this way you will become familiar with the simplest sonic effects possible. Only after you have a feel for simple waveforms (like the sine wave) should you try using more complex tones. In general, pulse and ramp waveforms tend to sound rather harsh and brittle, while sine waves have a soft, "outer space" quality.

We've mentioned the calliope example several times. Give it a try. The

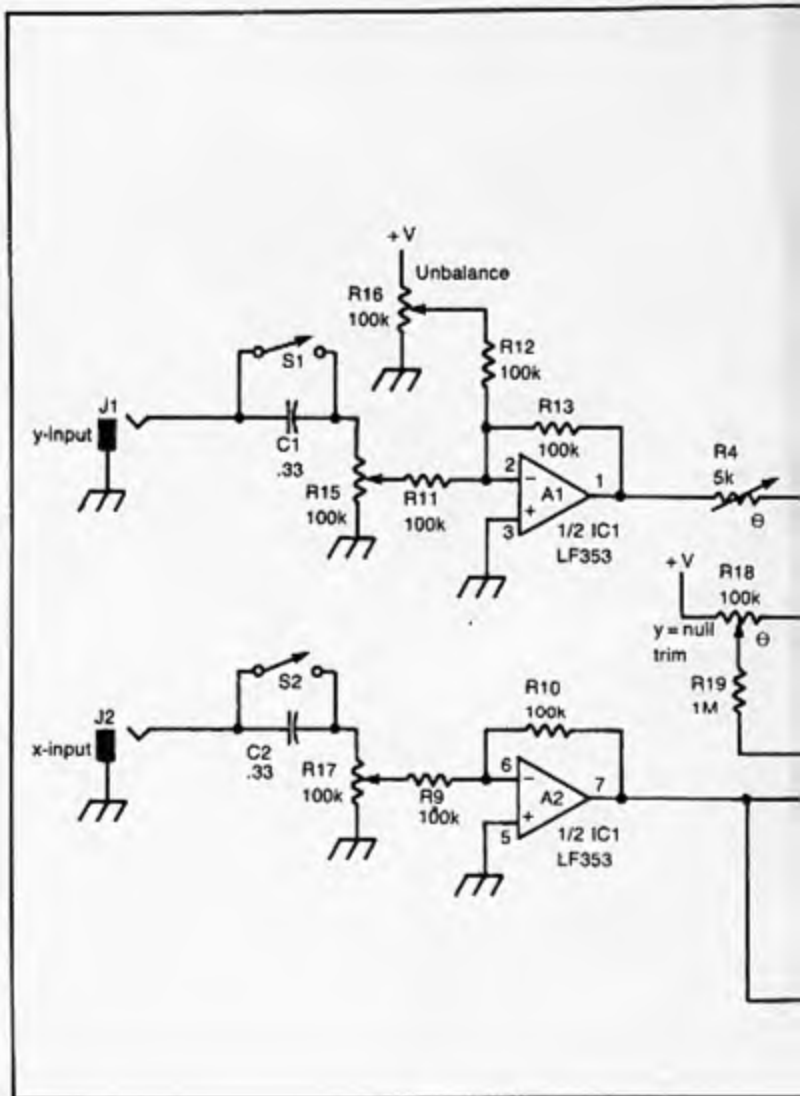


Fig. 13-1. Schematic diagram of the balanced modulator.

trick is to use a couple of sine waves or triangle waves for the input, and then deliberately unbalance the affair, thus letting some of the original note creep in. Play with R16 until just the right amount of the original sound is included. Now add in some white noise to imitate the sound of air rushing through a pipe. Finally, experiment with a bandpass filter to add a little resonance to the noise source. This isn't an easy patch, but makes a

Balanced Modulator

All resistors in ohms.

All capacitors in mfd.

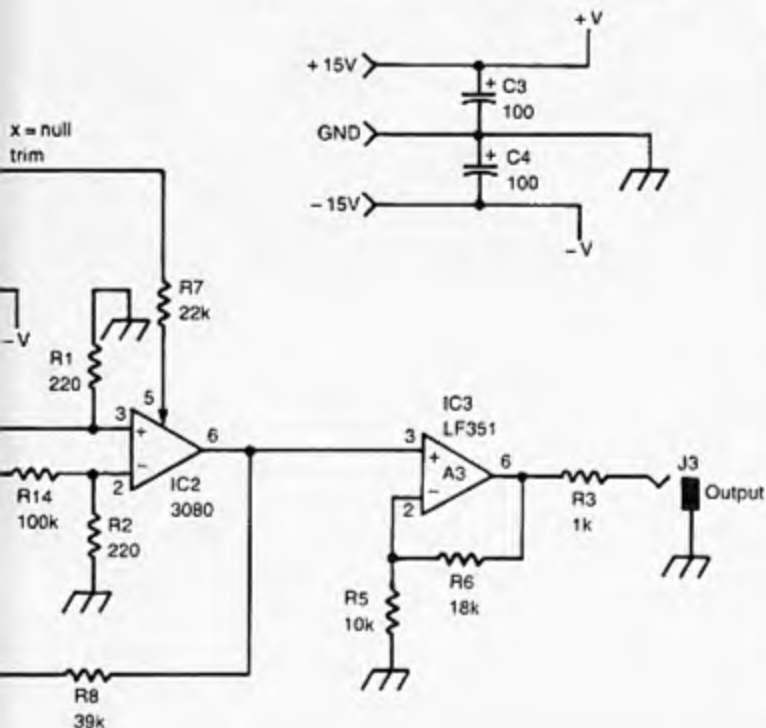
Power supply connections to opamps
assumed at:

IC1: Pin 4 = -15 V

Pin 8 = +15 V

IC2, IC3: Pin 4 = -15 V

Pin 7 = +15 V



fine challenge for your synthesizer.

One important point that you will want to keep in mind when you use this module is that static inputs tend to create very boring sounds. Always try to vary the frequency of one of the inputs in some manner. A very good example is to use an ADSR to sweep one of the VCOs acting as an input. This will help to create a very ghostly sound, a kind of whistling. The name

Resistors		Capacitors	
R1, R2	220	C1, C2	.33 mfd. mylar
R3	1k	C3, C4	100 mfd. electrolytic
R4	5k trimmer		
R5	10k	Semiconductors	
R6	18k	IC1	LF353 dual op-amp
R7	22k	IC2	3080 OTA
R8	39k	IC3	LF351 op-amp
R9-R14	100k		
R15-R17	100k potentiometer	Miscellaneous	
R18	100k trimmer	J1-J3	open circuit phone jack
R19	1M	S1, S2	SPST toggle switch

Fig. 13-2. Parts list for the balanced modulator.

of the game is experiment. The balanced modulator, more than any other module, takes the most rehearsal and practice to tame it into a proper musical instrument!

Chapter 14

The Power Supply

The power supply is one of the most important modules in the synthesizer in the sense that if it doesn't work right, then none of the other modules will either. Fortunately, building a good power supply is quite easy nowadays due to the advent of IC voltage regulators. Even though these ICs simplify things considerably, you will still want to take your time and plan your power supply carefully.

All integrated circuits in a modern electronic music synthesizer need a steady source of dc voltage. As discussed earlier, a bipolar 15V supply and an optional +5V supply can take care of all needs of the system. Standard 110V ac wall current is clearly unsuitable, so the power supply must convert this large-voltage, alternating-current source to a low-voltage, direct-current source. This is the main duty of the power supply. One additional requirement is that the dc voltage be well regulated. This means that no matter what the load placed on the supply, the voltage remains completely stable. Varying amounts of current may be withdrawn from the source, but the voltage always remains fixed.

Now for a few warnings. Remember, although the dc voltage present at the output of the power supply is quite low, the input voltage is 110V and this can be exceedingly dangerous if incorrectly approached! Wall current can kill you if you're not careful! This being the case, use extreme caution when dealing with the ac end of the power supply. Use good wire with heavy insulation, electrical tape, heat-shrink tubing, or whatever it takes to completely insulate the dangerous voltage from the rest of the synthesizer. Remember at all times that synthesizers (especially road units) tend to be used in uncontrolled environments and any extra protection that can be built in is well worth it.

How much current should the supply be capable of handling? Well, this depends entirely upon the rest of the synthesizer. A big system clearly needs a big supply, whereas a road unit might be able to get by with much less. A good rule of thumb is to build the supply for as much current as possible, and in this way you will be well prepared for later expansion. One amp on each side of the supply should hold out for at least twenty-five to thirty-five modules under most circumstances. This is about as big as you can build it, because the limiting factor is the regulator. Most commonly available regulators are good for about one amp, but only if well heat-sunk, so plan accordingly.

Having described the basic design problems and made the appropriate warnings, let's move on and see about the details of the circuit.

HOW THE CIRCUIT WORKS

Figure 14-1 shows the schematic for a very complete power supply, while Fig. 14-2 gives the parts list. A bipolar 15V supply is shown, along with a +5V one. If you have no need for this latter supply, simply eliminate it. But before you do so, be sure to think about future expansion. Remember, if you start to get into computer interfacing at a later date, that +5V supply will come in mighty handy!

Let's look at the bipolar supply first. A 35V center-tapped transformer reduces the 110V ac to a lower source of alternating current. Because this is a 35V transformer and because it is center-tapped, each side of it is capable of putting out about 17.5V. Note that transformers are measured in rms, so the peak voltage put out by each side is actually about 24V (without a load). You can always find the peak voltage by multiplying the rms value by the square root of two. Twenty-four volts is fairly high, but the regulator will chop this down to size a little later on.

The low voltage ac is converted to dc by the four rectifiers. Note that one leg of the ring of rectifiers puts out a positive voltage while the other puts out a negative voltage. The center tap of the transformer establishes the ground level.

The positive and negative legs each go to their own filter capacitor (C7 and C8). It is the duty of these capacitors to filter the ripple out of the full-wave rectified signal. Note that the ripple won't completely vanish (at least while the circuit is under load), but the regulator takes care of this later on. By the way, C7 and C8 should be good quality capacitors rated for 35V or more. Do not skimp on these parts since so much depends upon them!

Let's trace the positive leg of the circuit now. The rectified and filtered dc is presented to IC1, which is the positive regulator. This regulator clamps the voltage to +15V no matter what the input voltage and no matter what the output load conditions. C1 and C2 are compensation capacitors and immunize the regulator from transients, spurious oscillation and other noise

problems. These capacitors must be soldered right to the pins of the regulator in order for them to do their job! Don't eliminate them, thinking that they're superfluous. They are critical for making the power supply work right.

The output at this point should be a rock solid +15V. All regulators vary a little bit, so the output might be a rock solid +14.75V or +15.35V or some other value! This doesn't matter much for the purposes of the synthesizer, as long as the value is fixed and reasonably close to +15V.

Besides providing power to the rest of the system, note that a variable bias voltage can be output via R2. R2 and its associated jack should be front-panel features. Sometimes you will want a variable bias voltage (to create offsets, etc.) and that is the purpose of this control. It isn't crucial to the power supply, so if you feel that you don't need it, feel free to eliminate it from the design.

The negative leg follows the same pattern as the positive. C4 and C6 stabilize IC2, and the output is a steady -15V. Note that the capacitor values are a little higher than those on the positive side. This is because the 7915 tends to be a little more prone to problems than the 7815 positive regulator. However, with the addition of these capacitors, the unit is quite reliable.

D9 is an LED, and it would typically be mounted on the front panel to provide an indication that the circuit is turned on. R3 provides a source of negative bias, and can be used in the same way as the positive source mentioned above.

Let's back up for a moment and look more closely at the ac input to the power supply. In Fig. 14-1 you will note that the entire unit is fused. Fuses are often neglected on home-built units, but there's no excuse not to include one. They're inexpensive, easy to install and provide a margin of safety as well. It can't be emphasized enough that a synthesizer should be made as safe as possible and a fuse will go a long ways towards protecting you and your equipment. Get a panel mount unit and install it somewhere handy.

The convenience outlets shown in the schematic aren't strictly necessary, but do make systems interconnections considerably easier. For example, you might be using an echo unit with your synthesizer; rather than running in another extension cord to power it, you could simply plug it in to the convenience outlet. Since power supplies don't tie up very much panel space, you might as well throw these in. Note that they are fused as well, so be sure not to exceed the current rating. Further note that these outlets are switched, meaning that the main power switch may be easily used to shut down everything on the same line.

The +5V supply is essentially just a copy of the +15V supply, the main difference being in the type of regulator used. If you can find a transformer with an additional winding that puts out ten to twelve volts, use it. Otherwise, simply employ a separate transformer for this supply. See

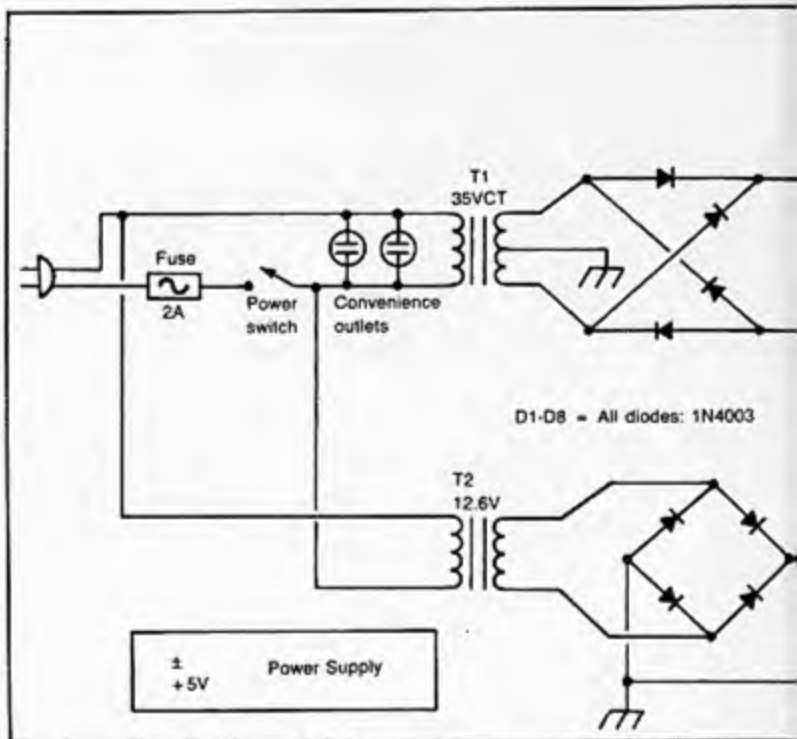
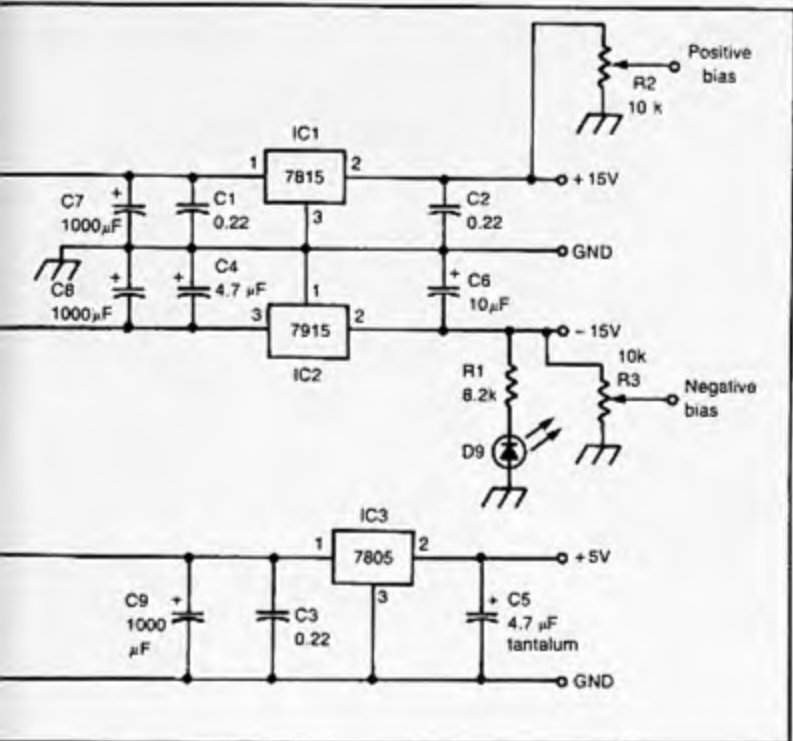


Fig. 14-1. Schematic diagram for the power supply.

Resistors		Capacitors	
R1	8.2k	C1-C3	22 mfd. mylar
R2, R3	10k potentiometer	C4	4.7 mfd. electrolytic
		C5	4.7 mfd. tantalum
		C6	10 mfd. electrolytic
		C7, C8	1000 mfd. 35V electrolytic
Semiconductors			
IC1	7815 +15V regulator		
IC2	7915 -15V regulator		
IC3	7805 +5V regulator		
D1- D8	1N4003 rectifier		
D9	LED		
Miscellaneous			
T1	35V center-tapped transformer		
T2	12.6V transformer		
	2A fuse, convenience outlets, switch, power cord, etc.		

Fig. 14-2. Parts list for the power supply.



below for more help in selecting a transformer.

CONSTRUCTION

Building the power supply is easy, if you take a great deal of care in the layout. Remember, dangerous voltages are involved so insulate any exposed wires. When stripping wires for this project, be sure you don't cut too deeply into the insulation and damage the wire, and use high-quality wire throughout.

You can build the power supply on a printed circuit board or a prototype board. Wire-wrap and perfboard methods are undesirable since the wire used in these is far too slender to pass the high amounts of currents involved. When building the unit on a circuit board, watch for solder bridges. With other modules, solder bridges usually just keep the circuit from working, but in a power supply, they can cause sparks to fly! Exercise extreme caution.

Selecting the right transformer for the job is most important. For the +5V supply any transformer that puts out 10V to 18V is acceptable. Don't use 6.3V filament types since they only put out about 8V peak which is

too "borderline" for the regulator. In general you need an input voltage to the regulator of at least 10V for it to work correctly. On the other hand, don't go too wild. An input of 30V, for example, will cause the regulator to run extremely hot and may cause damage.

For the bipolar supply, a transformer of 35V to 40V ratings is recommended. Twenty-four volt or 25V filament transformers are not suitable, being far too low in voltage to work correctly. And be sure that the transformer selected is rated for at least 1 amp per side, or else it will run hot and cause all sorts of problems.

If you are only taking 100 mA or less from the regulators, then no heatsinks are required. However, currents in excess of this will necessitate the inclusion of adequate heat sinking. There's no reason to buy these, if you are money conscious. Simply cut some stock aluminum down to size, drill mounting holes and spray-paint with flat black and you'll have a more than adequate heatsink for just pennies. If you are pushing the supply to the 1A limit, the heatsink should have about 5 to 10 square inches of surface. After making the heatsink, squirt some silicone heatsink compound between it and the regulator. This will improve the thermal bond considerably.

As mentioned, the power supply doesn't consume much front panel space. If you build the full version, you will need room on the panel for the line cord, fuse, switch, bias controls, convenience outlets and the LED indicator. These will all fit comfortably behind a standard 1 3/4-inch by 19-inch rack panel with room left over. Use the extra room for a one-by-four multiple if you wish.

When installing the power supply, be sure to watch the orientation of the transformers. Keep them away from any sensitive audio circuitry and you won't run into any hum troubles. Also, watch the wire routing. Use cable clamps, tie-bands, heatshrink tubing or whatever it takes to keep things neat and orderly. And when you are wiring the 110V lines, you must be sure to insulate all exposed wires!

USING THE POWER SUPPLY

You don't really "use" this module in the regular sense of the word. Essentially, you just turn it on and it supplies power to the system. However, you do need to know how to hook up the other modules to the unit, and this is covered in Chapter 15.

The power supply is often considered to be "too basic" to merit much consideration and is often essentially an afterthought. This simply shouldn't be the case. You need to think about it just as carefully and spend just as much time in coming up with a good design as you would with any electronic circuit. After all, if the power supply isn't reliable, then neither is the whole synthesizer!

Part 3

Putting It All Together



Chapter 15

Assembly

Well, you've come a long way now. By this time you should have some modules together, a rack mount selected, hardware organized, and are ready to put everything in place. This is the stage where things get exciting and consequently you'll be most tempted to cut corners and rush. But whatever you do, don't make haste! It's very important to take your time and plan everything out. Patience is the name of the game now. There's nothing quite so satisfying as when the modules slip right into place and everything goes exactly as planned. But that won't happen unless you have a master plan in mind. This chapter will guide you in forming that master plan.

A concept that was talked about in an earlier chapter, but is important enough to bear repeating, is that of "Psycho-Cybernetics." This was a craze a number of years ago and was a technique devised for improving your outlook on life, your personality, etc. What does it have to do with electronic music synthesizers? Well, consider the basic principles of Psycho-Cybernetics in terms of a concrete example. Suppose you're going to apply for a new job tomorrow. Psycho-Cybernetics states that the night before, you should visualize in your mind the entire interview. You'll try to see the new boss, the offices, your co-workers, in short anything that you're apt to see at the actual interview. The key is to let your imagination run wild and try to see every detail in your mind. And you shouldn't just use your mental "eyes" either. Use your "ears" and try to imagine the boss's voice asking you various questions and your responses.

The point of all this should be obvious. If you go through a "mental" interview the night before, then the actual one the next day, should be a breeze. After all, you've already been through the interview; the real one will just be a matter of going through the motions!

This concept is easy to apply to building synthesizers. When getting ready for a large project, you should plan for it well in advance by "building" the synthesizer in your mind. In this way you will know what pitfalls to avoid, what features to add and the proper order in which to do things. Remember, if you're building a one-transistor metronome circuit and goof up, you can probably recoup your losses by gerry-rigging it. But a synthesizer is so monstrous compared to this that there simply isn't any room for major slip-ups. You have to do it right the first time!

Your principle should always be to spend one night (or more) visualizing exactly what you plan to do the following day. Use whatever crutches you need. For example, get a big piece of butcher paper and make sketches of the whole system. Use colored pens, erasers, templates, whatever it takes to organize everything. Take notes, use a dictaphone, round up magazine articles, buy tools; you get the idea. Go at this like a passion!

If you follow this principle then your projects will probably go together easier, look nicer and perform better than you could ever have hoped for. So remember "Psycho-Cybernetics!"

POWER SUPPLY CONNECTIONS

Well, that was pretty philosophical! Let's come back down to earth with some pointers on how to wire up the final unit. It's assumed that you have the power supply and other modules all ready to go and that they're mounted in the cabinet. All that remains to be done is the final wiring.

Rule number one for wiring a synthesizer states that each and every module must have its own power supply run. That is to say, don't "daisy chain" the power supply connections. Make sure that you have a separate group of wires running from the power supply to each module. This reduces the chance of interaction and is especially important where "hot" signals, such as pulse waves, white noise, etc., are involved. You don't want any garbage being imposed on the power supply lines and creeping back into other circuits.

If you've followed the directions in Chapters 5 through 13, then you will already have installed decoupling capacitors at each module. If you haven't, do so now. Some manufacturers skimp in this area; there is no excuse for you to. Use 100 mfd. capacitors throughout and be sure to watch the polarities carefully.

When routing the wires through the cabinet to the various modules, use cable clamps and tie-bands. These help organize the wires into neat bundles and in general help to reduce hum and noise. Troubleshooting is made considerably easier too. Figure 15-1 shows the inside of a homemade synthesizer; note the use of clamps to keep the "trunk lines" neat and orderly. Incidentally, since I've just mentioned troubleshooting, this is a good time to point out that whatever you put together should be built in such a way that it can be taken apart again if needed. Modifications, improvements and other changes may occur to you at a later date. By making the

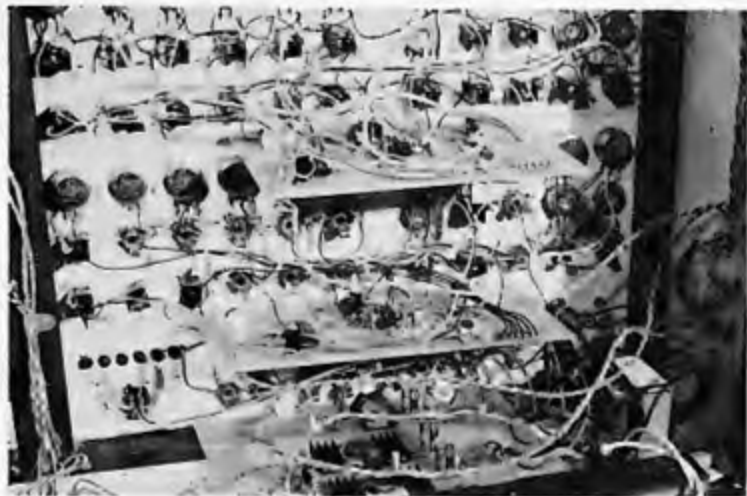


Fig. 15-1. Power supply bus connector.

unit easy to take apart, you stand a good chance of rendering these changes. Remember, you're not making a magician's box! The layout should be so neat and orderly that it's immediately obvious how to take the unit apart if needed.

Use color-coded wires for the power supply connections. One good scheme is to use red for +15V, green for ground, yellow for -15V and orange for +5V (if needed). Use the electric drill method mentioned in Chapter 4 to create some twisted pairs, and the results will look very professional.

One idea that might not have occurred to you is a method for distributing the power via some bus bars. Figure 15-2 gives the general idea. You need two phenolic terminal strips, each one with four solder lugs on it. Screw these down to the cabinet about two inches away from one another. Now slip a two-inch piece of bare bus wire through each pair of lugs and solder in place. You'll be left with four sturdy, parallel bus lines. Now solder the +15V supply line to one of the bus bars, the -15V to another, ground to another and +5V to the last. Whenever you want to hook up a new module, simply solder the power supply connections to a free space on the bus wires. As you can see, you can do this without disturbing other modules' power supply connections and even better, you can disconnect any module without interference to another.

You're probably getting sick of hearing it, but again be very careful with the 110V ac side of the power supply. This is especially true if you're using a metal rack cabinet. Be certain that every wire is well insulated and that the entire system is fused. Heat-shrink tubing is exceptionally handy for securing everything and making it safe. Simply slip a piece of this tubing over the offending wire and heat the joint with a blow drier. If you're careful, you can use a match to shrink the tubing, as long as you keep it

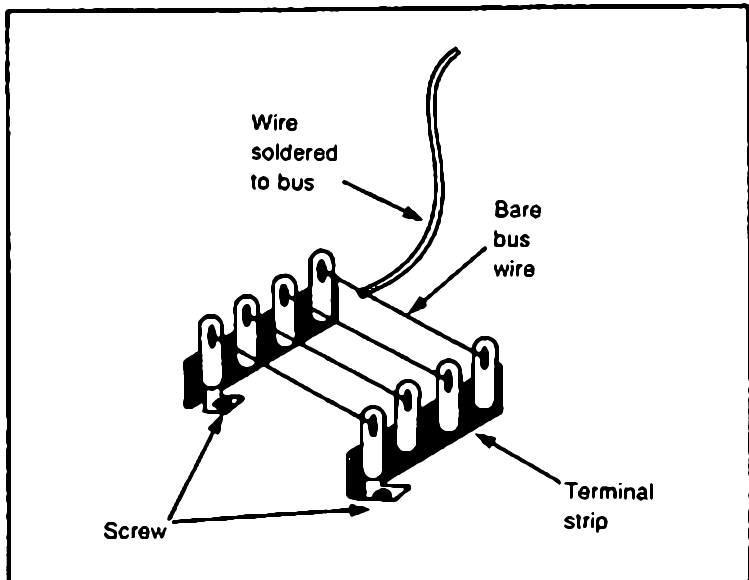


Fig. 15-2. Running trunk lines inside the synthesizer.

moving and don't let the heat build up on any one spot.

Remember that power supplies can generate a fair amount of heat, especially near the heatsinks of the regulators. This being the case, you will want to be certain that no wires are draped across the supply. Also, one point that might not have occurred to you is that a VCO shouldn't really be mounted right on top of the power supply. Even though all modern VCOs are compensated for temperature effects, why take chances? Because pitch is such an important parameter give the VCO every chance to work correctly. Every other module in the synthesizer runs cool, so place the VCO next to one of these.

COMPLETING THE SYNTHESIZER

When installing the modules in the cabinet, follow some sort of logical order. For example, you might want to put the VCO below the VCF and this below the VCA. Since these modules are used in this order more often than not, it pays to mount them in the same way. This, of course, reduces the length of the patch cords needed to interconnect the modules. Likewise, it makes sense to organize all of the audio circuits in one area and the controller circuits in another. Then it will be easy to trace patch cords down simply by noting the general area from which they are emanating.

At this point you should have the modules installed and wired up. You will no doubt want to put some sort of a cover on the back of the unit and thus insulate it from moisture, dirt and temperature extremes. Use about

six bolts to affix the cover, because you may need to take it off again at a later date.

The last step is to throw on the frills. These include metal corners for the cabinet, handles, rubber feet, etc. The frills are the icing on the cake, but they really help give the final unit that polished look. It can't be emphasized enough that a musical instrument which looks nice puts you more in the mood for making music. So browse around the hardware store and see what sort of niceties they have which will make your synthesizer a work of art.

TESTING AND TROUBLESHOOTING

When you have the unit all put together, sit down and play it for a number of hours. This pays off in several ways. First of all, you'll become familiar with the effects of the various controls. This is very important, especially since a synthesizer has so many knobs, jacks and switches which may be confusing at first. Secondly, it is only after playing it for a while that you will start to discover what needs to be improved or worked on.

Malfunions will typically fall into three main categories. They are:

1. The module doesn't work at all.
2. It works, but not very well.
3. It works, but is plagued by hum and hiss.

A Module Doesn't Work at All. . .

Let's tackle each of these three types of malfunctions, one by one. The first category is actually the best sort of problem. Circuits that don't work at all are always easier to fix than ones which are intermittent. This is because the symptoms are very obvious and stay that way. The first things you should suspect are a misplaced wire, a broken wire or a bad solder connection. For example, if the power supply connections are reversed, or the lead-in wires broken, then the circuit will clearly not perform correctly. Likewise, a poor solder connection to an input jack could cause total circuit failure. So, for starters, check the power supply connections and the input/output wiring.

What sort of test equipment is needed to make this analysis? Your best tools are your eyes. Simply check everything over by inspecting the circuit carefully under a good light. Believe it or not—almost ninety percent of all problems can be resolved by a careful inspection.

A simple voltmeter is also handy to chase down this first kind of malfunction. Check that power is indeed present at the circuit board, and if necessary check the power pins of each IC. Confirm that all is as it should be.

If this doesn't get you anywhere, try rapping the various parts of the circuit with a pencil or some other insulated probe. Flex the circuit board carefully, or jiggle the connecting wires. In general, always suspect something mechanical first.

Next look for components that are obviously damaged. Are there any broken leads on the capacitors? Are the diodes in good shape? Are the ICs seated in their sockets correctly? Also be sure to check for polarized components which may be installed backwards. These include electrolytic capacitors, diodes, transistors, and ICs.

The very last thing you should suspect is a defective IC! For some reason this is the part that most hobbyists suspect first and yet statistically the IC is the least likely to cause problems. Always suspect yourself first (misplaced wire, poor solder joint, etc.), a mechanical problem next (shorted wire on the panel) and the ICs last.

The Module Doesn't Work Very Well

This type of problem is harder to chase down. If the circuit basically works, but not very well, then you've got to get sneakier in your approach. Start by writing down everything that the circuit can do, then jot down what it can't do. Now, while looking at the schematic, apply logic and form a hypothesis that covers the facts. Think of an experiment which you can perform to test this hypothesis.

Writing down the symptoms forces you to see things clearly and logically and tends to fix the facts in your mind. And since there is no such thing as magic in electronics, logic is your only weapon in the fight against bugs!

Try to narrow down what circumstances lead to the malfunction. Is it related to temperature, moisture, or does the problem only occur when the module is used with another particular module? A good way to chase down temperature disorders is with some spray coolant. Simply spray the suspected part with the coolant and watch for any possible effects. On the other extreme, you could blow hot air from a hair drier on the circuit and watch for results.

You shouldn't rule out any possible method when it comes to troubleshooting. Above all, remember to be orderly. The cardinal rules are to apply logic, think of experiments to prove or disprove theories and draw conclusions based upon the results of these experiments. Also, you need faith! Remember that any problem can be solved, if only you stick with it. To help you along in this department, you can be assured that every project in this book was built, tested and debugged before being included. You can make them work too.

Problems with Hum and Noise

If the circuit is subject to hum and noise (problem [3] from above) then you should suspect bad solder joints right off the bat. A cold solder joint acts just like a resistor, and placed in the wrong part of the circuit could introduce either hum or noise. Another possibility is interaction between several wires. To see if this is the case, try moving the connecting wires to and fro and see if you note any difference.

Hum is usually caused by improper grounding techniques. For example, check to see that all ground runs return to one point only. We all know that a ground wire is supposed to be at zero potential and hence passes no current, yet in practice it turns out that large ground runs with multiple returns often do transmit small currents. The result is a ground loop which more often than not can cause large amounts of hum. So check to see if you have a ground loop anywhere. Pay special attention to the front panel connections, since a ground loop is most apt to occur here. Also, if you have used shielded cable to interconnect pots and jacks to the circuit board, make certain that only one side of the shield is grounded. Under most circumstances you will ground the shield at the circuit board end of the wire and let the panel end float.

Troubleshooting: Case Histories

This covers the most common sources of problems and how they may be dealt with. To tie these principles together, here are a few case histories that illustrate the procedures to be followed when troubleshooting electronic circuits.

CASE #1. A flanger all of a sudden broke into a weird sort of oscillation and whistling. The first step in finding the problem was to open the unit up and put it through a simple visual inspection. This didn't show anything obvious, so the next step was to think and ask questions. What could possibly cause an oscillation in the unit? The original instrument sound was getting through, but the oscillation was much louder. More inspection still showed nothing obvious in terms of faulty panel connections. Then all of a sudden an idea struck. What would 60 Hz hum sound like if it were flanged? Ordinary hum is obvious, but what about flanged hum? After listening to the output again, it could be confirmed that something (possibly hum) was actually being flanged. A test capacitor (a 1000 mfd., 50V capacitor on alligator clips) was hooked across the dc output of the power supply and the problem stopped! Looking very closely at the big power supply capacitor it then became obvious that a printed circuit board pad had lifted up and fallen to pieces. After several minutes, the connection to the capacitor was reconstructed and the unit proceeded to work like a charm. The moral of this story is that even ordinary hum can sometimes sound different. This being the case you must be willing to let your imagination run wild at times.

CASE #2. A ten-band graphic equalizer was acting very erratic. As long as it wasn't touched or moved it worked just fine. But the minute that one of the pots was adjusted, it began to crackle and cut out. As usual, the first thing done was to open it and simply put it to a visual inspection. No obvious problems were detected. So the next step was think about it—still no luck. A third technique was employed; using the eraser end of a pencil, the unit was tapped. This wasn't done at random; rather, only those areas with potential "loosen-ups" were tapped. This gave some results.

It wasn't possible to identify the exact circuit board pads which were giving trouble, but the problem area was narrowed down. At this point a magnifying lens and a bright light were used and a thick layer of rosin between the pads and the lugs of the potentiometers was detected. A touch-up with the soldering iron was all it took to get the EQ back on the road again! The moral of the story is obvious: solder rosin is an insulator! How many times have you read this fact and just passed it over as being very unimportant?

CASE #3. A balanced modulator was not working correctly. The carrier signal was coming through loud and clear, but the signal wasn't being passed at all. This symptom narrowed things down somewhat. Since there was output of some sort, the problem must lie with the input. A quick visual inspection showed that a pad on the circuit board, right on the input capacitor, had lifted up. Thus the diagnosis was correct. The moral here is to always think functionally. Don't worry about details until you've found the basic area where the problem is occurring.

Summary of Troubleshooting Techniques

Using these three case histories as examples, the basic principles of troubleshooting can be summarized:

1. Always look before you leap. Ninety percent of all problems can be resolved by simply inspecting the circuit.
2. Next think.
3. When you do think, think in terms of functions or building blocks because this helps isolate the problem. When you've narrowed things down to a particular function, it's much easier to find individual component problems.
 4. Always be prepared for the unexpected (as in CASE #1, above).
 5. Make a test capacitor. Power supplies can sometimes be a problem.
 6. Tap with something insulated to find intermittent problems.
 7. And remember, rosin is an insulator!

Using these basic principles you should be able to troubleshoot any problem with your new synthesizer. Actually, this entire chapter has been quite alarmist; in all actuality, if you have taken your time to be neat and orderly, then the circuits will probably work right off the bat. Experience has shown that carefully planned projects stand a good chance of working perfectly.

USING YOUR SYNTHESIZER

So you've got the unit up and running, now what? The answer is simple: make music! But you're in a unique position since you can be both musician and scientist. This means that you can approach any musical prob-

lem in a scientific manner. You will no doubt want to get a notebook and keep copious notes on any new patches you've come up with. Listen to other people's music and try to figure out how they get a particular sound. Or you might try to imitate nature. Try making some rainstorms, surf sounds or tornados. What about traditional instruments; how would you go about "imitating" a saxophone, a flute, or an electric bass? Try all of these.

Don't expect anything dramatic at first. You should simply try to get some experience in analyzing sounds and seeing if you can duplicate them. If you are successful, then you have understood the basic principles involved.

At this point you are thinking on a microcosmic level. That is, you will be most interested in individual sounds and won't concern yourself with whether they are useful or not. Only after you feel at home with sound and the synthesizer should you start to think macrocosmically and try to compose entire songs based upon your previous work.

You will want to fill your notebook with sample patches. These samples can serve as starting points for more useful and serious work later on. Two magazines, *Polyphony* and *Electronotes* often publish sample patches which can be modified for use on your home-built synthesizer. (See Appendix B for information on these magazines.) A number of books also give some sample patches. For example, Delton T. Horn's *The Beginner's Book of Electronic Music* (Blue Ridge Summit, PA: TAB Books, 1982), Appendix D, gives quite a few patches that can be used with your homemade gear. Another possibility is *The Source*, (Oklahoma City, OK: Polyphony Publishing Company, 1978) which is a compendium of patches submitted by readers of *Polyphony*.

If you have built all of the circuits in this book, then you have probably spent a great deal of time in the lab. Shut the door and lock it! It's time to be a musician for a while. Spend the next several months being musical. And remember, your synthesizer is the best around. By now you should agree that you can always build them better than you can buy them!

SUMMARY

In this chapter the last steps needed to complete the synthesizer were covered. You learned how to supply power to the entire system and found out that it indeed makes a difference how the power is distributed. Separate runs to each and every module insure that interaction between the various circuits is minimized. Next the basic principles of troubleshooting were expounded. Good troubleshooting can only be learned with experience, but as a start you saw that looking, thinking and tapping can solve many problems. Finally some suggestions for starting a "patch" notebook were offered.

Since your synthesizer is all done and working like a champ, you may think that your job is done. Far from it! The next chapter offers some suggestions for future projects, including how to come up with a computer interface.

Chapter 16

Inventing New Modules

This book describes the main modules found in any synthesizer system. As such, they form the basic core and will find their way into just about every patch you come up with. But these modules certainly don't exhaust the list! Many other unusual circuits are possible, and by now you will have the experience needed to try your hand at designing some yourself. This chapter, then, gives some hints and tips to guide you along the path of inventing new modules for your synthesizer.

SUGGESTIONS FOR MORE ANALOG CIRCUITS

PHASE SHIFTERS. Phase shifters have been primarily used with electric guitars in the past, but can also create some excellent sounds when used with a synthesizer. In particular, when the input signal is a pulse/sub-octave mix, the output, after phasing, is an extremely lush, "swooshing" sound. Most guitar phase shifters simply phase back and forth under a fixed LFO control, but one built for the synthesizer is much more versatile in that any control voltage may manipulate the phase delay. For example, phasing controlled by the ADSR creates an excellent sound.

Phase shifters are very easy to make now, thanks to large-scale integration. For example, you might try the SSM2040 chip or alternatively the CEM3320. Check the application notes for these chips to see how to set them up as phase shifters. You'll be surprised to see how easy it is.

FLANGERS. Flangers create a sound similar to phasers but most people find it to be much more "intense." Once again, these are most commonly used by guitar players, but have much to offer to the synthesist as well. Many commercial recordings utilize this effect and the sound has been likened to a jet taking off.

Flangers are a little trickier than most analog circuits in that radio frequency clocks are involved. You must exercise extreme caution to keep that high frequency clock from feeding through to the audio paths. Also, noise (inherent to the chips used in flangers) can be troublesome. But, with care, it is possible to design much better flangers than you can buy. And once again, by making them voltage controlled you can get far more sounds with a home-brew unit than with the commercial ones.

The most commonly used chip for flangers is the Reticon SAD-1024. This used to be about the only delay chip readily available to the experimenter, but recently other units have started to show up on the hobbyist level. Try the SAD-4096, for example, for some very long delay times. If you can get them, the Panasonic chips, such as the MN-3005, are very easy to use and quite low in noise. For sample circuits, refer to the appropriate application notes.

ECHO UNITS. Echo units are built along the same lines as flangers and use very similar parts. To get longer echo times, usually several delay chips are wired in series. Two SAD-4096s or four SAD-1024s might be a good starting point. Of course, using more chips possibly means more noise, so some sort of noise-reduction scheme should be contemplated. The 571 compander chip (made by Signetics) is one good way to add noise reduction to an echo unit.

ENVELOPE FOLLOWERS. This module can be used to extract the envelope from a traditional instrument and apply it to a synthesized signal. They are very easy to make, being no more than a simple full-wave rectifier. However minimizing the ripple and maximizing the response time of the unit can sometimes be a little tricky. Your best bet is to breadboard the circuit and adjust components as needed to make it work right with the intended instruments.

Circuits for full-wave rectifiers abound. Try National Semiconductors Linear Databook for some ideas; another good source is *Electronotes* which has published a number of practical circuits over the past five years. (See Appendix B)

FREQUENCY SHIFTERS. This is a most unusual module. It will shift any input frequency up or down by a fixed amount in an arithmetic fashion. For example, if the input signal is a sine wave of 200 Hz and the shift index is set for 50 Hz, then the output will have a frequency of 250 Hz (assuming you're shifting in an upward direction). This simple example with a sine wave is hardly interesting, but consider what happens if the input is a waveform consisting of a fundamental of 100 Hz, with an octave harmonic of 20 Hz. Perform the shift. The output will be a tone of 150 Hz with another of 250 Hz added in. Note that the shift was arithmetic and that the harmonic relationship has been lost. In general, then, a frequency shifter is used to create nonharmonic effects. If you put a human voice through it, for example, the output is still intelligible to a degree, but doesn't really sound like a voice anymore!

Frequency shifters are very big circuits and incorporate lots and lots

of balanced multipliers. This is not a suggested project for beginners, but if you're really looking for an advanced, state-of-the-art project, check out Bernie Hutchins' home-brew frequency-shifter circuit in *Builder's Guide and Preferred Circuits Collection*, published by *Electronotes* (see Appendix B).

SEQUENCES. The sequencer is used to create complex patterns of notes, often in some sort of melodic passage. Essentially you program a melody through the keyboard and the sequencer plays it back at any speed. It used to be that sequencers were manually set affairs containing many potentiometers, but modern computer methods have now made the design and construction of these units far simpler. The digital keyboard described in Chapter 10 can serve as the start of a sequencer since there is a digital, six-bit output all set to go. Simply clock the data into some RAM (random access memory) to program a melody and read it back to a DAC (digital-to-analog converter) to play the melody. The 2112 RAM experiments, being both easy to find and inexpensive.

Well, that should get you started. As you can probably tell, there are all sorts of new analog modules that can be built to jazz up your home-brew synthesizer. To keep abreast of new developments in this area, you will want to be sure to subscribe to some of the electronic music synthesizer journals. *Electronotes* and *Polyphony* are two good ones and offer several new circuits, suitable for experimenters, each month. Also, don't neglect *Electronics*, *Radio-Electronics*, *EDN* and some of the more technical journals. These often publish circuits which, while not specifically for electronic music, can often be adapted. Check out Appendix B for the addresses.

MOVING TOWARDS DIGITAL SYNTHESIS

True digital synthesis is still far too expensive for the home experimenter to even consider (although that might change one day soon). Digital synthesis is a means whereby a complex waveform is built up point by point and the construction of such a waveform takes into account the frequency, amplitude and timbre of the desired sound. As you might guess, additive synthesis such as this requires an extremely large computer capable of running at speeds far beyond those of typical home computers.

But hybrid computer control is possible for the experimenter. In the hybrid approach a microcomputer is used to control a standard analog synthesizer using special digital-to-analog circuits to complete the interface. The computer thus sends a number of signals to the interface circuitry, where they are converted to regular control voltages, gates and triggers that the synthesizer can respond to.

Likewise, the computer can read the keyboard if it is digital in nature. The keyboard presented in Chapter 10 is suitable for this purpose, since besides putting out analog information it also has a six-bit digital output. You should be able to get just about any home computer to read this keyboard.

How do you go about getting into hybrid digital synthesizers? The first step is to think about getting a personal computer if you don't already have one. The good part about this is that you can still use the computer for whatever other operations you had in mind. Thus, besides using it for music applications, you will still be able to prepare income taxes, perform scientific calculations, play video games, and execute countless other tasks. The point is, an electronic music interface shouldn't conflict with any other uses for the computer.

After getting your hands on a personal computer, you will want to spend as much time as possible getting familiar with it, and this means both from the standpoint of hardware as well as the software. Obviously, if you're going to build an interface for it, you must know about the hardware operating system. You'll probably have to learn about RAM (random access memory), ROM (read only memory), PIAs (peripheral interface adapters), data buses, address buses, the microprocessor and a whole slew of other technical concepts. This is not meant to put you off; learning about these things is quite interesting and will probably suggest other nonmusic uses for your computer as well.

You have to know about software, since that's what any computer program is. After coming up with a hardware interface, you will need to write programs to "drive" the hardware. For example, a typical program might contain subroutines to scan the keyboard, latch a note, send the note to a DAC, keep track of polyphony, transpose passages, play back a sequence, etc. It pays to really learn the ins and outs of programming, since you will want to make sure that the computer is operating at the peak of its ability.

To accomplish all of these tasks, there is no doubt that you will have to learn to program in machine language. Most personal computers come equipped with BASIC as the fundamental language, but are able to do machine language as well. BASIC is certainly a pleasant language to program in, but unfortunately is far too slow to manage a complete synthesizer system. Just for comparison, a typical implementation of BASIC on a personal computer runs programs one hundred to one thousand times slower than an equivalent machine language program is able to run them!

To become an efficient machine-language programmer, you will need to accumulate books, magazines and charts illustrating the ways in which the microprocessor used in your computer may be programmed. In addition, you might need some extra tools, like machine-language monitors, assemblers, disassemblers, etc. Once again, mentioning all of this is not meant to discourage you; look at it as a challenge. Programming is actually quite fun, and is well within the capabilities of anyone who can think logically. Of course, you will have to log lots of practice writing simple programs first, but that's just part of the learning process.

COMPUTERS FOR ELECTRONIC MUSIC

Just about any home computer can be used for electronic music, but

there are two in particular that stand out as good choices. These are the Commodore VIC-20 and the Sinclair/Timex ZX-81. Why these? Well the main reason is price. The ZX-81 can be found in stores for under \$100, while the VIC-20 is now below \$200. Most home experimenters operate with a limited budget and these computers are certainly low enough in price to justify the expenditure, even if they're only used for music. (Of course they can be used around the home for many other things.)

Let's look at the ZX-81 first. This is a very small unit, in terms of size, but is actually a very complete computer. The BASIC interpreter is quite slow, though, since the Z80 MPU chip must keep track of a number of things including the video display. This being the case, you should expect to conduct all of your programming for musical purposes in machine language. And incidentally, the display (use an ordinary television for this) is actually quite good and very suitable for musical purposes.

Interfacing to the ZX-81 is fairly straightforward since every MPU line is brought out to a connector on the back side of the computer. The address lines, data lines, control lines, etc., are all there. Thus a ribbon cable may interconnect the computer to the synthesizer in a fairly straightforward manner. You shouldn't have to mar or alter the actual computer in any way.

The VIC-20 is even easier to interface with, because you have access to an uncommitted VIA (versatile interface adapter) on the back side of the computer. You can think of this as a "port" with which you can interface just about any electronic circuit you may have in mind, and the port may be configured and reconfigured simply by changing the software. Once again, a ribbon cable and connector is about all it takes to hook something up to the port.

The VIC-20 has an extremely powerful operating system and is quite easy to program for musical purposes. Besides the uncommitted VIA port, mentioned above, you have access to a number of other connectors, powerful ROM routines and a very reliable program tape save mechanism. In general, the VIC-20 is much easier to program, and faster than the ZX-81, but then again the price is almost double. As with all things in life, you get what you pay for. Choose the computer that suits your needs and pocketbook and you won't go wrong.

If you are buying a computer especially for electronic music purposes then you might want to consider either the VIC-20 or the ZX-81 since these are the lowest priced personal computers around and are more than powerful enough for this application. (Other computers will work equally well, including ATARI, OSI, TRS-80, or any other personal computer. Thus, if you already have a computer, put it to work with your synthesizer!)

Before you dive in though, be sure to accumulate stacks of manufacturer's broadsheets, books, magazines and any other literature that will guide you in the selection and purchase of a computer. Then keep reading so that you will learn everything you can about your new computer. Finally, join local user groups in your area and try to meet other people in-

terested in your projects. User groups are a great way to expand your contacts and often times when you're stuck on some problem, you will meet another member who has already encountered and solved it!

So become an information exchange. To keep up in this quickly changing field you will simply have to absorb every piece of information that you can!

ON TO THE FUTURE!

You've come a very long way now since you started to learn about the three parameters of sound! If all has gone well, you should be the proud owner of a synthesizer far better than any you can buy. You're probably starting to think about future projects, including new analog circuits as well as digitally controlled units. Whatever you do, keep track of the information flow and share your results with others. This is the start of the golden age of electronic music, and you're now a part of it!



Appendices



Appendix A

Suppliers

This list of suppliers is far from complete, but should stand you in good stead for quite some time. Although all of these suppliers have provided good service in the past, I give no warranty in supplying this list; it is for reference only.

Each of these places has stocked electronic parts of interest to synthesizer builders at one time or another. Of particular interest are PGS Electronics (which stocks a complete line of the SSM music chips and many other synthesizer parts), PAIA Electronics (which stocks electronic music kits and the CEM line of music chips), Pratt-Read (which stocks keyboards suitable for synthesizers), and Sound Logic (which handles a broad range of parts for *Electronotes* projects). Write to these places to secure catalogs and ordering information.

All Electronics Corporation
905 S. Vermont Avenue
Los Angeles, CA 90006

Bullet Electronics
P.O. Box 401244F
Garland, TX 75040

Chaney Electronics
P.O. Box 27038
Denver, CO 80227

Diamondback Electronics
P.O. Box 12095
Sarasota, FL 33578

Digi-Key Corporation
P.O. Box 677
Thief River Falls, MN 56701

Jameco Electronics
1355 Shoreway Road
Belmont, CA 94002

Mouser Electronics
11433 Woodside Avenue
Santee, CA 92071

PAIA Electronics
1020 W. Wilshire
Oklahoma City, OK 73116

PGS Electronics
P.O. Box 749-C
Terre Haute, IN 47808

Poly Paks, Inc.
16-18 Del Carmine Street
Wakefield, MA 01880

Pratt, Read and Company
Ivoryton, CT 06442

Sound Logic
P.O. Box 49331
Austin, TX 78765

Star-Tronics
P.O. Box 683
McMinnville, OR 97128

Appendix B

Sources

Books and Manuals

Anderton, Craig. *Electronics Projects for Musicians*. New York, New York: Music Sales Corporation, 1975. (More than any other book, this one contains the clearest examples of how to actually build something. Although most of the projects are geared to a guitar-playing audience, many of the circuits may be adapted to synthesizer without much problem. There are lots and lots of excellent tips in this book.)

Anderton, Craig. *Home Recording for Musicians*. New York, New York: Music Sales Corporation, 1978.

Brown, Robert and Olsen, Mark. *Experimenting with Electronic Music*. Blue Ridge Summit, Pennsylvania: TAB Books, 1974.

Chamberlin, Hal. *Musical Applications of Microprocessors*. Rochelle Park, New Jersey: Hayden Book Company, Inc., 1980. (If you're thinking of computer control, then this is the one book you must have!)

Drake, Russell and Herder, Ronald. *How to Make Electronic Music*. Pleasantville, New York: Educational Audio Visual Inc., 1975.

Douglas, Alan. *Electronic Music Production*. Blue Ridge Summit, Pennsylvania: TAB Books, 1982.

Friend, David, Pearlman, Alan R. and Piggott, Thomas D. *Learning Music with Synthesizers*. Winona, Minnesota: Hal Leonard Publishing Corporation, 1974.

Horn, Delton T. *The Beginner's Book of Electronic Music*. Blue Ridge Summit, Pennsylvania: TAB Books, 1982.

Horn, Delton T. *Electronic Music Synthesizers*. Blue Ridge Summit, Pennsylvania: TAB Books, 1980.

Hutchins, Bernie. *Musical Engineer's Handbook*. Ithaca, New York: *Electronotes*, 1975. (If you can only get one book, make it this one. This is the single most important book ever published on design and construction of synthesizers! You will find a wealth of circuits and design information on every conceivable topic.)

Hutchins, Bernie. *Electronotes Builder's Guide and Preferred Circuits Collection*. Ithaca, New York: *Electronotes*, 1980. (This is another important work and contains dozens of practical, tested circuits.)

Jenkins, John and Smith, John. *Electric Music: A Practical Manual*. Bloomington, Indiana: Indiana University Press, 1975.

Polyphony, The Source. Oklahoma City, Oklahoma: Polyphony Publishing Company, 1978.

Strange, Allen. *Electronic Music*. Dubuque, Iowa: Wm. C. Brown Company Publishers, 1972.

Ward, Brice. *Electronic Music Circuit Guidebook*. Blue Ridge Summit, Pennsylvania: TAB Books, 1975.

Wells, Thomas. *The Technique of Electronic Music*. New York, New York: Schirmer Books, 1981. (This book is a perfect example of the excellence of work which can be done by a person knowledgeable in both electronics and music. It probably has the best illustrations of any electronic music book around.)

Periodicals

Journal of the Audio Engineering Society, published ten times yearly by the Audio Engineering Society, 60 East 42nd Street, New York, New York. (You will have to be a member of the AES to get this journal, but it's worth the effort. Though articles on electronic music synthesizer design only show up several times a year, it's generally true that they're among the most important published. For example, one of the earliest accounts of a voltage-controlled synthesizer, by Robert Moog, appeared in this journal.)

Common Mode, published six times yearly by PGS Electronics, P.O. Box 749-C, Terre Haute, IN 47808. (This newsletter is filled with many useful tips, circuits and projects for electronic music experimenters. Among the

items featured are applications for the new SSM line of music chips.)

Electronotes, published by Bernie Hutchins, 1 Pheasant Lane, Ithaca, New York 14850. (*Electronotes* publishes a monthly newsletter, applications notes, and several other "extended service" publications. You should subscribe to them all, since you simply won't find any other publication like this! *Electronotes* has consistently published the most important articles in the field of electronic music synthesizers and has set an unequalled standard of excellence. You simply can't afford to do without *Electronotes*.)

Polyphony, published bimonthly by Polyphony Publishing Company, P.O. Box 20305, Oklahoma City, Oklahoma 73156. (*Polyphony* is a very slick magazine featuring articles by some of the best writers in the field. There are usually two or three excellent construction projects each month.)

Synthesource, published irregularly by Curtis Electromusic Specialties, 110 Highland Avenue, Los Gatos, California 95030. (This newsletter is filled with high-quality circuits exploiting the Curtis line of music ICs. In the short time that it has been in existence, it has published many important design articles.)



Index



Index

A

- ADSR generator, 16
 - construction of, 82
 - one-chip, construction of, 80-84
 - operation of one-chip, 80
 - output of, 17
 - parts list for one-chip, 83
 - schematic for one-chip, 81
 - use of one-chip, 83
- amplitude, 3-6
- amplitude modulation (AM), 19
- AR generator
 - input of, 15
 - output of, 16
- assembly and modification (Part III), 139-153
- attack time, 14

B

- balanced modulator, 21
 - calibration of, 127
 - construction of, 127
 - operation of, 126
 - parts list for, 130
 - schematic diagram for, 128-129
 - troubleshooting, 146
 - use of, 127
- balanced modulator, 124-130
- breadboard, 46
- bus connector, 141

C

- capacitors

- disk, 29
- electrolytic, 29, 30
- Mylar, 29
- polanized, 30
- polystyrene, 29
- selection of, 29
- tantalum, 30
- chromatic scale frequencies, equally tempered, 8
- circuit board, 46
 - attaching the, 47
- circuit construction, 45
- construction projects (Part II), 57-136
- control voltages, 27
- controller, 22
 - envelope extractor, 23
 - pitch extractor, 23
 - pressure sensor, 23
 - rotary, 23
 - theremin, 22

D

- decay or release time, 14
- decibel, 9
- digital keyboard
 - block diagram for, 98
 - calibration of, 110
 - construction of, 108
 - operation of, 97
 - parts list for, 108
 - schematic diagram for, 102-107
 - use of, 110
- digital synthesis, 150

dynamic range, 9

E

echo units, 149

electrical standards, 25

electronic music computers, 151

envelope followers, 149

envelope generator, 14

 attack-decay type (AD), 15

 attack-decay/sustain-release
 (ADSR), 16

 attack-release type (AR), 15

 patching, 18

equalizer, erratic operation in, 145

F

FET input amplifier, 32

final decay, 17

flanger oscillation, 145

flangers, 148

flea clips, 46

four-quadrant multiplier, 125

Fourier analysis, 3

frequency, 3-6

frequency modulation (FM), 19

frequency shifters, 149

G

glissando, 23

H

harmonic content, 3-6

hum problems, 144

Hutchins, Bernie, 71

I

initial decay, 17

input structures, 28

integrated circuits (ICs)

 FET input amplifier, 32

 op amps, 32

 quad op amps, 33

 selection of, 32

J

jacks, 33

K

keyboard, 12, 22

 digital, 97-110

L

light box, 46

location modulation (LM), 117

low-frequency oscillators (LFO), 19

M

multiple (jacks), 27

N

noise

 source of, 19, 20

 white, 20

noise problems, 144

O

op amps selection guide, 34

operational amplifier (op amp), 32

output structures, 27

P

patch-over hardware, 48-52

 hold scheme for, 52

 master scheme for, 50

 mixer for a VCO, 51

perforboard, 46

Perrin, Ken, 115

phase shifters, 148

pitch, 4-6

pitch extractor, 23

portamento control, 91

potentiometer, 31

potpourri module, 111-115

 calibration of, 115

 operation of, 111

 parts list for, 114

 schematic diagrams for, 112-113

 use of, 115

power supply, 131-136

 connections for, 140

 construction of, 135

 operation of, 132

 parts list for, 134

 schematic diagram for, 134-135

 use of, 136

psycho-acoustics, 5

Q

quad op amps, 33

quadrature function generator,
116-123

 calibration of, 119

 construction of, 119

 four outputs of, 117

 operation of, 118

 parts list for, 122

 schematic diagram for, 120-121

 use of, 121

 VCA control in, 122

R

rack mount enclosures, 37

- rack panels
 - construction of, 39-45
 - plans for, 41
 - sizes and mounting holes for, 42
- resistors
 - carbon composition, 30
 - carbon-film, 30, 31
 - metal-film, 31
 - selection of, 30
- ring modulator, 22

S

- semitone, 8
- sequences, 150
- sidebands, 124
- signal levels, 26
- slew rate, 32
- sound
 - accuracy of, 7
 - chromatic scale frequencies of, 8
 - components of, 3, 6
 - decibels of, 9
 - dynamic range of, 9
 - precision of, 7
 - semitones in, 8
 - volume of, 9
- supercontroller (SN76477)
 - calibration of, 91
 - construction of, 91
 - front panel layout for, 93
 - introduction to, 85
 - operation of, 86
 - parts list for, 92
 - patch for bizarre white noise with, 96
 - patch for delayed vibrato effect in, 94
 - patch for gated ADSR effect with, 95
 - sample and hold patch for, 96
 - schematic diagram for, 88-89
 - use of, 94
- suppliers, 157
- supply voltages, 25
- synthesizers
 - assembly of, 139
 - circuit construction for, 45
 - completion of, 142
 - definition of, 3
 - digital, 150
 - Introduction to (Part I), 1-53, 3
 - inventing new modules for, 148
 - modules of, 11
 - obtaining parts for, 34

- planning, 36
- power supply for, 24, 131-136
- selecting components for, 28
- testing and troubleshooting, 143
- using, 146

T

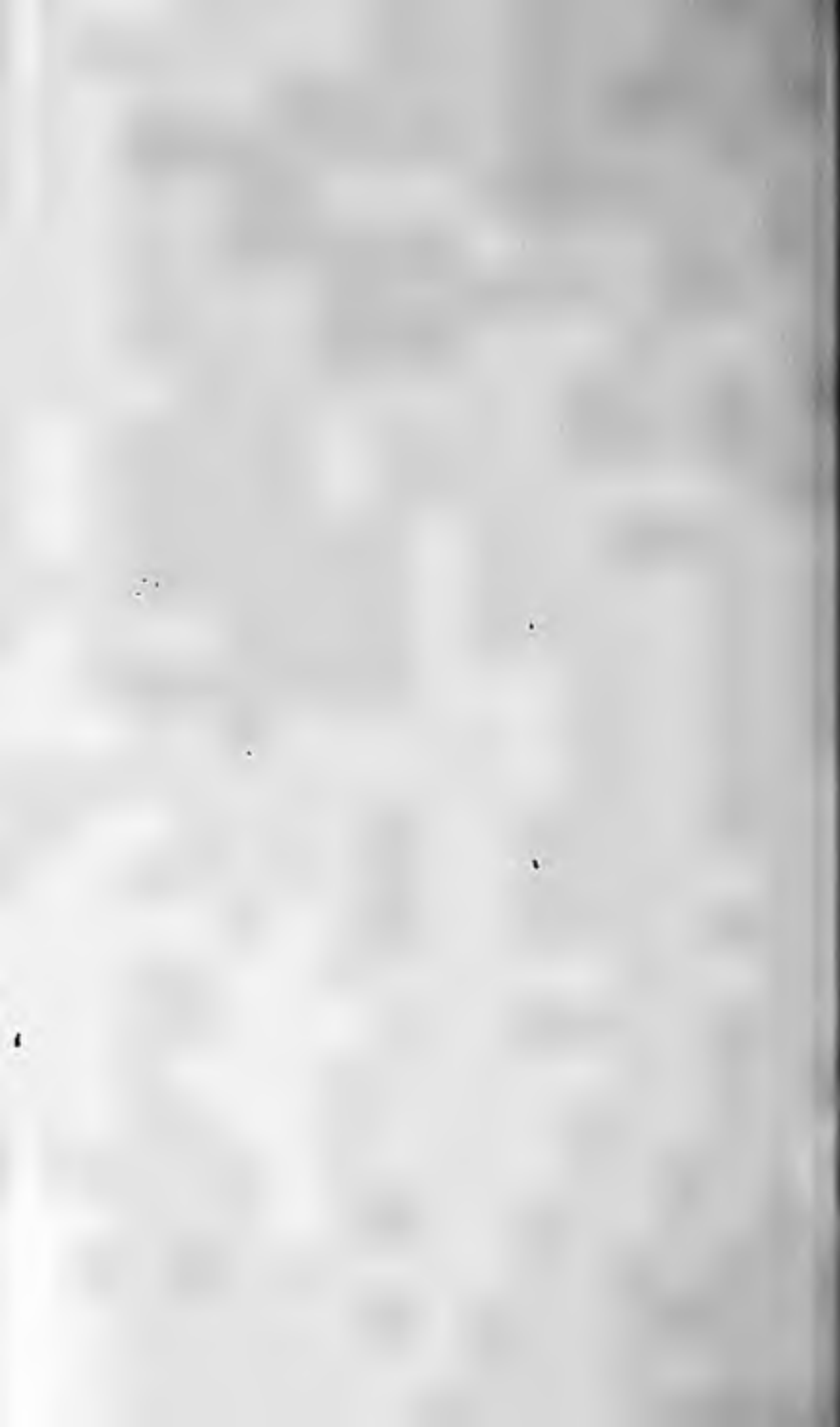
- theremin, 22
- timbre, 4, 6
- tracking, 12
- tremolo, 19
- trigger signals, 28
- trunk lines, 142
- TTL ICs, 26
- two-quadrant multiplier, 125

V

- vibrato, 19
- voltage-controlled amplifier (VCA), 13
 - calibration of dual, 78
 - construction of dual, 75
 - dual, construction of, 73-79
 - operation of, 74
 - parts list for dual, 78
 - schematic diagram for, 76-77
 - use of dual, 78
- voltage-controlled filter (VCF), 13
 - calibration of four-pole lowpass, 70
 - construction of four-pole lowpass, 67
 - features of four-pole lowpass, 66
 - four-pole lowpass, 65-72
 - operation of, 66
 - parts list for four-pole lowpass, 70
 - schematic for four-pole lowpass, 68-69
 - troubleshooting the four-pole lowpass, 71
 - using the four-pole lowpass, 72
- voltage-controlled oscillator (VCO), 7, 11
 - calibration of, 63
 - construction of, 57-64
 - operation of, 57
 - parts list for, 62
 - specifications for, 58
 - using the, 64
- voltage-controlled oscillator (VCO), 60-61
- volume, 4-6, 9

W

- windstorm effect, 21





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