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MUSIC SYNTHESIZERS

A MANUAL OF DESIGN & CONSTRUCTION

BY DELTON T. HORN



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Introduction

When electronic music synthesizers started to become popular in the 1960s, it was widely claimed that these wondrous instruments could create any sound—real or imagined.

Curiously, this rather extravagant claim is both true and not true. In theory at least, any sound can be synthesized electronically (nobody has tried every possibility yet). Some sounds may be extremely difficult and, perhaps, impractical to electronically synthesize, but in principle they should be possible. On the other hand, no individual synthesizer can generate all possible sounds.

Electronic music synthesizers are made up of a number of circuits, or modules, which can create or influence a signal in various, specific ways. To create a truly infinite number of sounds, an infinite number of modules would be required.

For this reason, different synthesizers are more suited than others to creating specific types of sounds. There is no single ideal electronic music synthesizer. It all depends on just what you need.

A great many synthesizer models are commercially available today. Some are excellent instruments, but they may not be perfect for your needs. You may be able to find a commercial synthesizer that includes all the modules you want, but you may end up with a number of additional modules that aren't of much use to you. You have to pay for them anyway.

In this book, you'll learn how to build your own customized electronic music synthesizer. It will be far cheaper than a ready-

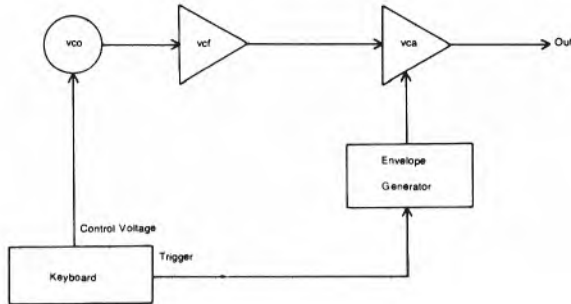
made commercial unit, and it will only have the modules you specifically need. Moreover, you can position and mark the controls in whatever manner suits you. You can also expand your synthesizer as your needs demand (or your budget allows). The only limit on the number of modules is how many you can build and afford.

In addition, you'll end up with a unique instrument, unlike any other in the world, and you can take pride in knowing it is your own handiwork.

This book includes dozens of circuits that can be mixed and matched to create hundreds of different synthesizer systems. Many tips on their use are also presented.

Some knowledge of basic electronics and the ability to work from schematic diagrams is assumed, but no previous experience or knowledge of electronic music and synthesizers is required.

Chapter 1



Introduction to Synthesizers

An *electronic music synthesizer* is, simply, a circuit or collection of circuits that generates, or in some way modifies, an electrical signal which will eventually be fed into a loudspeaker to produce a sound. Electronic music synthesizers may be used to simulate traditional instruments or to create new, never-before-heard voices. These sounds may be used in traditional or avant-garde musical compositions or as sound effects.

Most modern synthesizers rely heavily on the concept of voltage control. In a sense, this is letting electrical signals adjust the knobs and switches. Many effects that would be impractical or impossible to create by manually twiddling controls are a snap with voltage control.

No sound exists within the synthesizer. Only electrical signals flow through the circuits. There are two basic types of voltages used within a synthesizer. The distinction between the two is far from hard and fast. The two voltage types are signal voltages and control voltages.

Signal voltages are ac signals of various waveshapes. These signals will eventually be fed into the loudspeaker to be converted into sound.

Control voltages are used in voltage-controlled circuits to modify one or more specific parameters of a signal voltage, or, in some cases, another control voltage. Control voltages may be either dc or ac, depending on the effect desired.

A single voltage may be used as a signal voltage and a control voltage simultaneously. There is no fundamental difference between signal voltages and control voltages (although signal voltages must be ac signals within the audible range of frequencies). The distinction is only in how they are used at a specific point.

This may sound a bit confusing, but it will become clearer as you grow more familiar with synthesis techniques. This seemingly arbitrary distinction is extremely helpful in discussing sound synthesis.

Synthesizers are made up of a number of circuits, or modules, that may be interconnected in various ways. On some synthesizers, the interconnections are permanent or may be changed by switches. Other synthesizers allow the user to make any connection he desires with external patch cords.

In this book, you will learn how to build and use many synthesizer modules. The modules you select will determine the capabilities of your finished instrument.

Some synthesis modules are rather exotic and are used infrequently. Others are more or less standard in virtually every synthesized sound. Most electronic voices are generated by an oscillator (see Chapter 3) and modified by filters (Chapter 5) and amplifiers (Chapter 6).

It is also possible for a synthesizer to use external signal source such as a human voice (converted into an electrical signal by a microphone) or an electric guitar (using a pick-up). Any sound signal that can be fed through an amplifier can be used in an electronic music synthesizer (although the signal may have to be preamplified or attenuated to make sure it is at the proper level for the circuits involved).

POPULARITY

Electronic music synthesizers were once esoteric playthings of the avant-garde. Today, their popularity is immense and still growing. Many movie soundtracks and commercial jingles are produced on synthesizers, and these versatile instruments are becoming as common as the electric guitar in pop music bands.

There are many reasons for this growth in popularity. For one thing, in recent years the cost has come down and they have become easier to play. Early synthesizers were only practical in studio work, and were monophonic—that is, they could play only a single note at a time. Today many synthesizers are available for live

performance situations, and polyphony (the capability for playing chords) is becoming more and more common.

When the classic *Switched-On Bach* album by Walter Carlos (Columbia MS 7194) appeared in the late sixties, the synthesizer was an exotic novelty that few people had even heard of, much less seen. Today it is far more familiar and is accepted as a valid musical instrument by most people. Popularity begets greater popularity.

The synthesizer is a wonderfully versatile musical instrument. It can sound like dozens, hundreds, or even thousands of instruments—some recognizable, and others entirely new. A band using a synthesizer can play a wider variety of music. It would be impractical to hire a brass section for just one or two songs in a concert. Let the synthesizer play the brass parts. Many other sounds are only possible through sound synthesis.

In addition to this, the electronic music synthesizer opens up a whole new area of potential creativity for the musician. Besides just creating the notes, harmonies, and rhythms, the musician can create uniquely expressive voices for almost any desired effect.

COMMERCIALLY AVAILABLE MODELS

Many companies are manufacturing ready-made synthesizers today, and sales are booming. Moog is the most widely recognized name in this field, but there are dozens of others. A number of commercially available synthesizers are discussed in detail in *Electronic Music Synthesizers* (Tab Book No. 1167) and *The Beginner's Book of Electronic Music* (Tab Book No. 1438).

A few typical commercial models are shown in Figs. 1-1 through 1-3. Prices range from a few hundred dollars to several thousand dollars.

WHY DESIGN YOUR OWN?

Building your own synthesis equipment has many advantages, not the least of which is cost. A homebrew unit will tend to be considerably less expensive than a comparable commercial model. The ready-made unit might look a bit snazzier, depending on your skills, but the actual electronics can be just as good.

By building your own customized synthesizer, you are not limited to what somebody else considers a good compliment of modules. You can include as many modules (including duplicated modules) as you like, without having to pay for modules that are of little or no interest to you.



Fig. 1-1. The Emulator by Emu Systems.

The system can be set up for your own personal convenience. The control panel can be arranged in a manner that is perfectly comfortable for you.

Building your own equipment improves your understanding of the principles involved. This means you'll know your instrument better, allowing you to be more adventurous and creative in its use.



Fig. 1-2. The Crumar General Development System.



Fig. 1-3. The Realistic/Moog Concertmate MG-1.

In addition, you won't be reliant on someone else if something goes wrong. Synthesizers are notoriously difficult to find good service for (as a whole—there are a few exceptions).

The pride you'll feel at playing an instrument of your own handiwork is another advantage that should not be brushed aside. The work can be a lot of fun too.



Fig. 1-4. Experimenter's breadboard by Heathkit.

Of course, some people don't want to get involved in the circuitry. They'd prefer to just buy their instruments. After all, how many guitarists build their own guitars? For these people, a commercially available system is certainly the way to go.

However, an electronics hobbyist has a big advantage when it comes to electronic music synthesizers.

A RECOMMENDATION

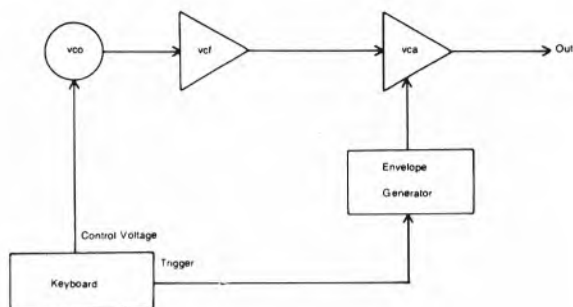
It is recommended that you read through this entire book before starting to build anything. Determine what modules you want to build, and how they will be inter-connected and packaged (especially see Chapter 10), before you start hooking up circuits, otherwise, you might find much of your effort wasted, and your modules may not fit together properly.

Don't rush. Make sure you know what you're doing at every point. An electronic music synthesizer is a complicated project, and expensive and time-consuming errors will catch you if you're careless. If you work slowly and carefully, preplanning your system, and thinking out each step ahead of time, you should have no real problems.

It's a good idea to temporarily hook up each circuit on a breadboard (see Fig. 1-4) before soldering anything. This will allow you to make sure the circuit does what you want it to do (it's just not possible to describe sounds on paper). An experimenter's breadboard will also allow you to make circuit modifications and try out various component values without much trouble.

Now, let's get down to the proverbial brass tacks and get to work on our customized electronic music synthesizers!

Chapter 2



The Power Supply

We have to start out with the most mundane, but absolutely essential portion of our synthesizer—the power supply. Nothing will work without a power supply.

Synthesizer modules usually draw too much current to make battery power economically feasible. In almost all cases, an ac to dc power supply circuit will be required.

Most synthesizers utilize a single power supply (often with multiple voltage outputs) for all of the modules, although a very large systems may include more than one power supply circuit to ease the current drawn through each.

Switching the system on and off is a bit of a problem in a synthesizer. No ideal situation exists, so a compromise is necessary. An individual power switch for each individual module clutters up the control panel, and makes the instrument awkward to operate. It's too easy to forget to turn a few switches on or off.

Most commercial Synthesizers have a single master power switch that turns on all of the modules. This makes the unit easier to use, and neater looking, but, since not all of the modules are likely to be used at once, the current drawn by the system is greatly increased. Still, this is usually the most practical approach.

Designing a power supply for a synthesizer system is somewhat paradoxical. You can't work on the other modules until the power supply is up and running, but you can't know what kind of power supply you'll need until you know the requirements of all of

your system's modules. Fortunately, you can make some reasonable estimates.

REQUIREMENTS

First off, different modules will probably require different voltages. As a rule of thumb, we can assume a low positive voltage (5 volts is typical), and a higher positive/negative voltage (± 12 volts is a good choice). If intermediate voltages are required, they can be tapped off with a simple voltage-divider resistor network.

A synthesizer power supply should be capable of passing a lot of current. Each individual module probably won't consume very much power, but the cumulative effect can add up very quickly. Ten modules consuming 100 mA each works out to a full ampere.

A one amp power supply is just about the minimum for a small music synthesizer system. Remember, some modules may consume much more than the 100 mA mentioned in the previous example. A more practical value would be from 2 to 3 amps. A large system might require as much as 4 or 5 amps.

You don't have to go overboard with a 10 amp power supply, but try to allow for your maximum planned system. It's better to err on the side of too much power than too little. A rough rule of thumb would be one amp of power supply current for every 6 to 8 active circuit modules. If you intend to have more than two dozen modules in your finished system, you probably should consider multiple power supplies, perhaps with a common ganged switch.

Be sure to include a fuse in your power supply circuit. Far too many experimenters try to save a few cents by omitting fuses. Don't! Mistakes and accidents happen to the best of us. Without a fuse, many modules (including some expensive components) could be damaged, or even ruined. Moreover, there is always the risk of fire and/or injury. A fuse is cheap and simple insurance. Admittedly, sometimes a fuse won't blow in time, but in most cases it will.

Use only fast blow type fuses—not the slow blow type. If you find you are repeatedly blowing out fuses, **do not** substitute a higher value or a slow blow type—find out what is making the fuse blow and repair the problem.

Keep a good supply of spare fuses on hand. You could accidentally cause a momentary short somewhere in the system, or a high voltage transient on the ac power lines might pop a fuse. Having a stock of the right fuse on hand will prevent the temptation to substitute (even temporarily) a different value or (worse yet!) to bridge the fuse holder with a piece of wire.

Since fuse holders are often sold in pairs at electronics stores such as Radio Shack, you can mount the extra holder in some convenient, out of the way place on your synthesizer, and store a spare fuse there.

Always down-grade your power supply's current handling capability when selecting the fuse size. Leave yourself a margin of error. For example, if your power supply has a nominal rating of 3 amps, it should be fused at 2.5 to 2.75 amps. Don't beg for trouble—not all components meet their nominal ratings exactly. Assume they're a little low.

There are many different types of power supply circuits, and several of these will be discussed in the pages to follow. Some modules discussed in this book are quite uncritical, and can be powered with the rippling dc from a simple rectifier type power supply. Other circuits are more critical (you will be told when this is the case) and will require voltage regulation. In virtually any circuit including integrated circuits (ICs), you can assume some degree of voltage regulation will be required. This is especially true of digital ICs.

CIRCUITS

Now let's examine some common types of power supply circuits and try to determine what will be suitable for a synthesizer system.

Let me repeat that you should read through this entire book first to determine what modules you intend to build before you build anything—especially the power supply. The wrong power supply could prove to be a real white elephant.

Half-Wave Rectifier

Probably the simplest ac to dc power supply is the simple half wave rectifier circuit. This device is illustrated in the schematic diagram of Fig. 2-1.

This simple circuit works because of the basic action of a diode. A diode will allow current to pass through it in only one direction. A signal of the opposite polarity (reverse biased) will be blocked.

Since an ac voltage cycles back and forth through positive and negative instantaneous values, only half of the waveform signal can pass through the diode—hence the name, *half-wave rectifier*.

Figure 2-2 illustrates how the signal looks at the input and the output of a typical half-wave rectifier circuit. Notice that this is not a

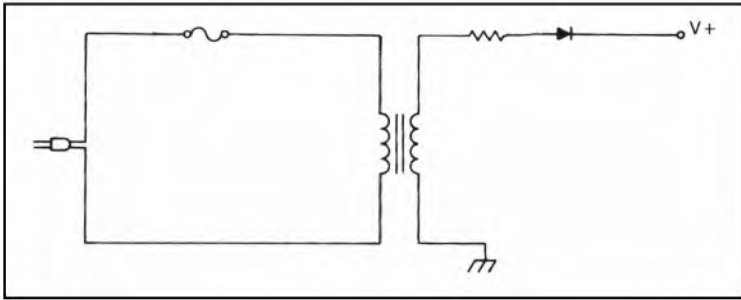


Fig. 2-1. Half-wave rectifier.

dc signal at the output. It is a constantly varying, pulsating waveform with a constant polarity.

Since only half of the input ac waveform is used, the half-wave rectifier wastes a lot of the potential energy. The wrong polarity voltages are discarded.

The output signal can be smoothed out by adding a capacitor after the diode, as shown in Fig. 2-3. This capacitor should have a large value—typically a few hundred microfarads. Electrolytic capacitors are generally used in this application.

When the diode is forward biased (passing current), the capacitor is charged. On opposite half cycles, when the diode is reverse biased (blocking current), the capacitor is discharged. If the capacitor has a large enough value it will not be completely discharged by the time the next positive cycle recharges it. This produces a rippling dc signal, as illustrated in Figure 2-4.

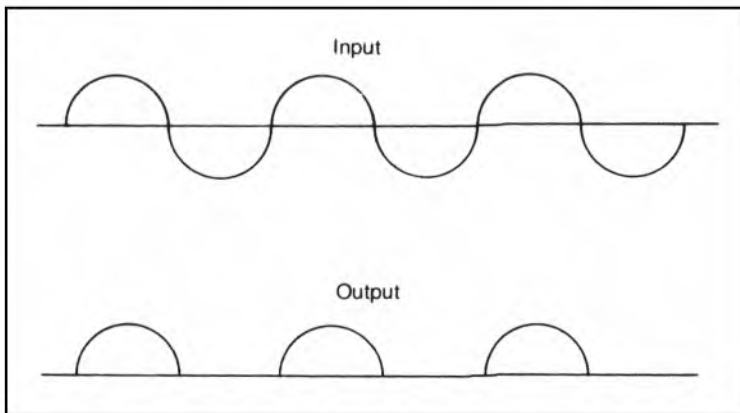


Fig. 2-2. Input and output signals for a half-wave rectifier.

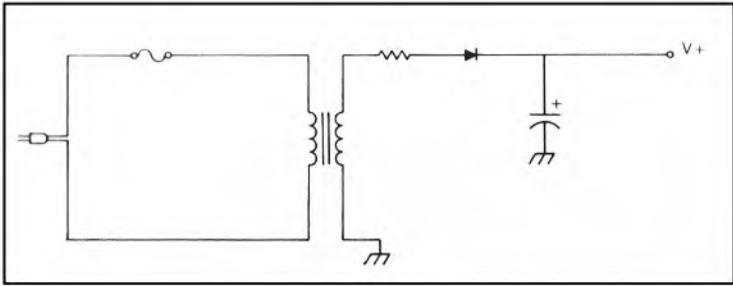


Fig. 2-3. Half-wave rectifier circuit with filter capacitor.

Full-Wave Rectifier

To make up for the wastefulness of the half-wave rectifier, a second diode can be added, as shown in Fig. 2-5. More commonly, you will see four diodes connected in the rectifier bridge arrangement of Fig. 2-6.

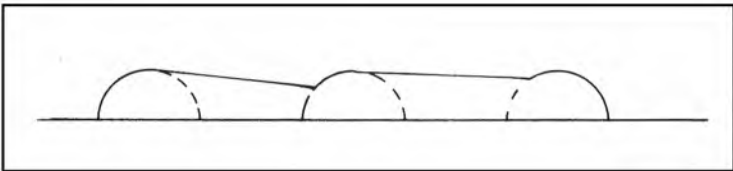


Fig. 2-4. The filtered output of a half-wave rectifier.

On one half of the ac cycle, the current flows through the circuit in the path shown in Fig. 2-7. On the other half cycle, the path shown in Fig. 2-8 is taken, so the final output signal is as shown in Fig. 2-9. Since the entire cycle is fed to the output, this more efficient power supply circuit is called a *full-wave rectifier*.

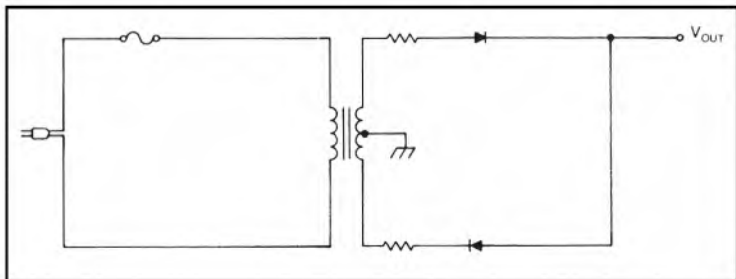


Fig. 2-5. Full-wave rectifier circuit.

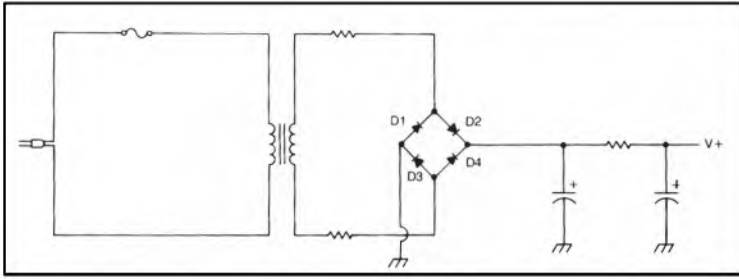


Fig. 2-6. Rectifier bridge circuit.

Once again, a filter capacitor can smooth out the output voltage, as illustrated in Fig. 2-10. Notice that the full-wave rectifier circuit has inherently less ripple (pulsating dc) in its output than a comparable half-wave rectifier circuit.

Voltage Regulation

The rippling dc voltage output from half-wave and full-wave rectifier circuits is fine for powering some circuits. Others require a more stable and accurate supply. In cases like this, some form of voltage regulation is necessary.

The simplest form of voltage regulation is to use a zener diode, as shown in Fig. 2-11. A zener diode is a special type of diode that has a unique reaction to a reverse biased voltage. Below a specific critical level, a reverse biased voltage is blocked, just as with an ordinary diode. Once the applied reverse biased voltage equals or exceeds this critical level, or avalanche point, the diode begins to conduct heavily. This is shown in the graph of Fig. 2-12.

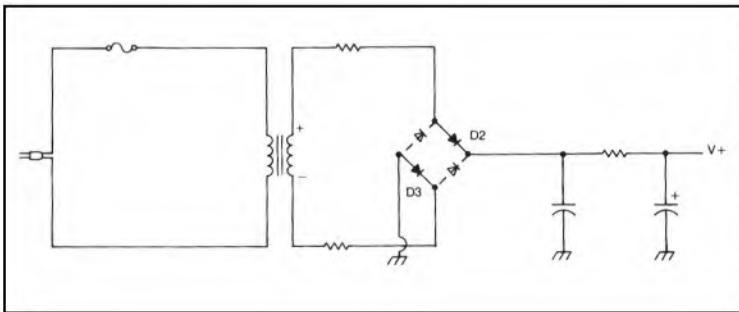


Fig. 2-7. Current path of rectifier bridge circuit during first half cycle.

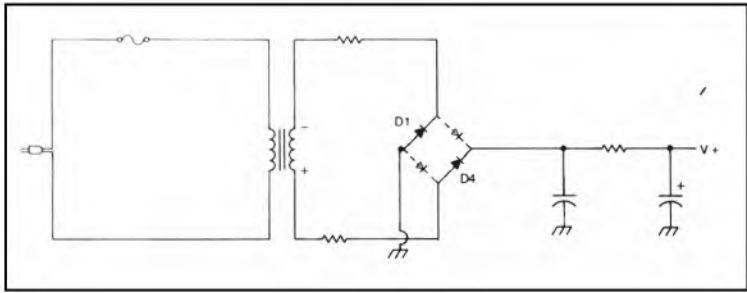


Fig. 2-8. Current path of rectifier bridge circuit during second half cycle.

This special property of zener diodes allows for a simple form of voltage regulation. In the circuit of Fig. 2-11, if the voltage exceeds the diode's avalanche point, the excess will be bled off to ground. Resistor R is used to prevent the zener diode from con-

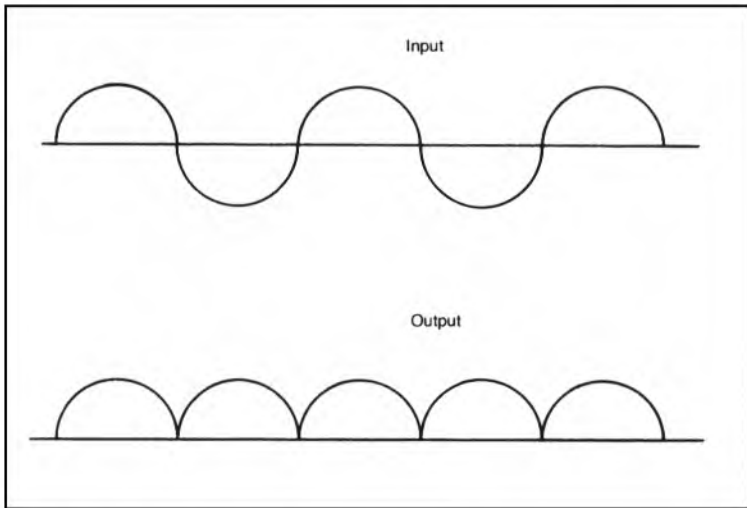


Fig. 2-9. Output signal of a full-wave rectifier.

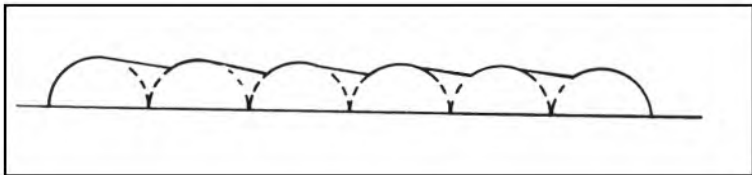


Fig. 2-10. Filtered output signal of a full-wave rectifier.

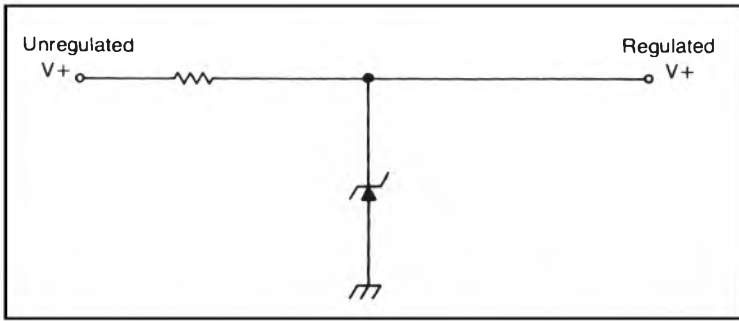


Fig. 2-11. Using a zener diode for voltage regulation.

ducting too much current and overheating and quite possibly burning out.

Zener diodes are available in a wide variety of voltages. The specified voltage is the nominal avalanche point of the zener diode (there is some variation between units—the voltage is not precise).

Much greater precision can be achieved with IC voltage regulators. These devices are usually sold in three lead packages, as shown in Fig. 2-13. One lead is the input, the second is the output, and the third is the common. Voltage regulator ICs will usually accept a wide range of input voltages with little or no fluctuation at the output. They are available in many common voltages (5 V, 10 V, 12 V, etc.). Positive and negative voltage models are available.

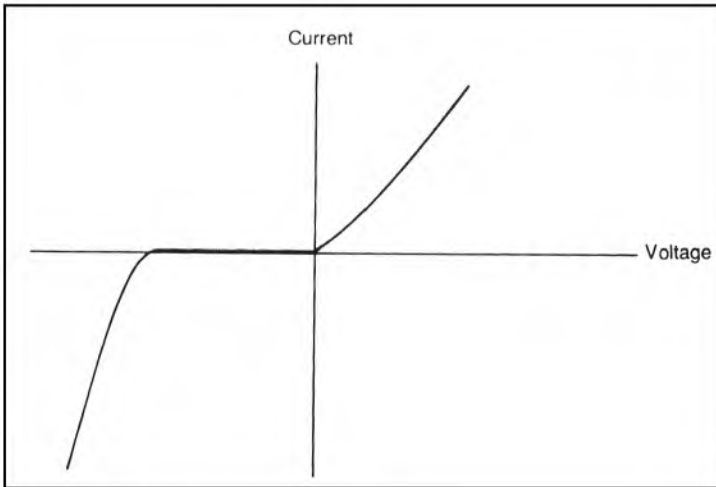


Fig. 2-12. Graph of zener diode action.

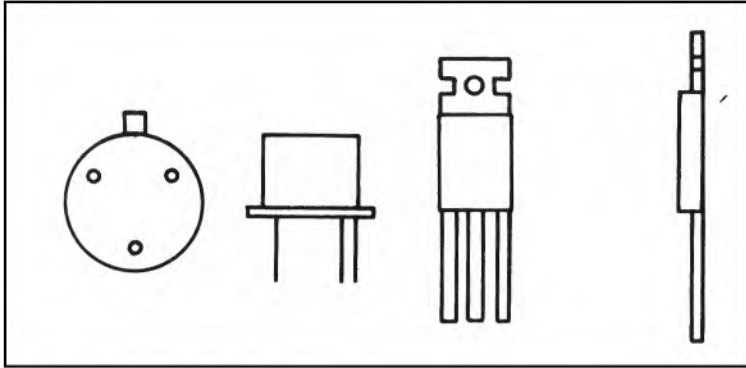


Fig. 2-13. Typical IC voltage regulators.

Figure 2-14 shows how voltage regulator ICs are generally used in a power supply circuit.

Voltage Dividers

Often it is necessary to tap off a number of different voltages from a simple power supply. This is usually accomplished with a resistor network called a *voltage divider*. This little trick is based on Ohm's Law.

Figure 2-15 shows a simple voltage divider circuit. If we assume the battery is putting out 3 volts, and the resistors each have a value of 100 ohms, there will be a total of 300 ohms in the circuit. According to Ohm's Law, current equals voltage divided by resistance ($I = E/R$). In this case, we find that $I = 3/300 = 0.01$ amp, or 10 milliamps. The current will be the same through each of the resistors. At point A, we can tap off the full 3 volts, of course. Before the voltage reaches point B, it is dropped by R_1 . The voltage

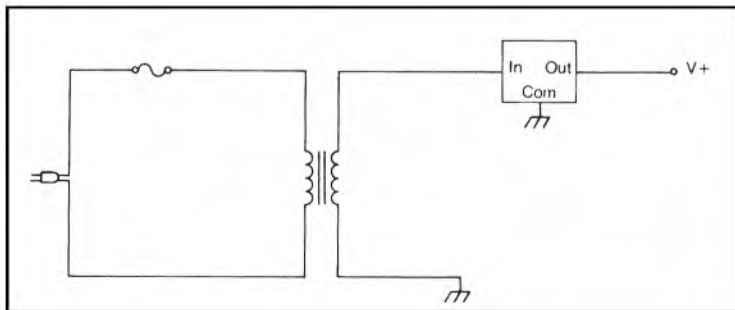


Fig. 2-14. Power supply using an IC voltage regulator.

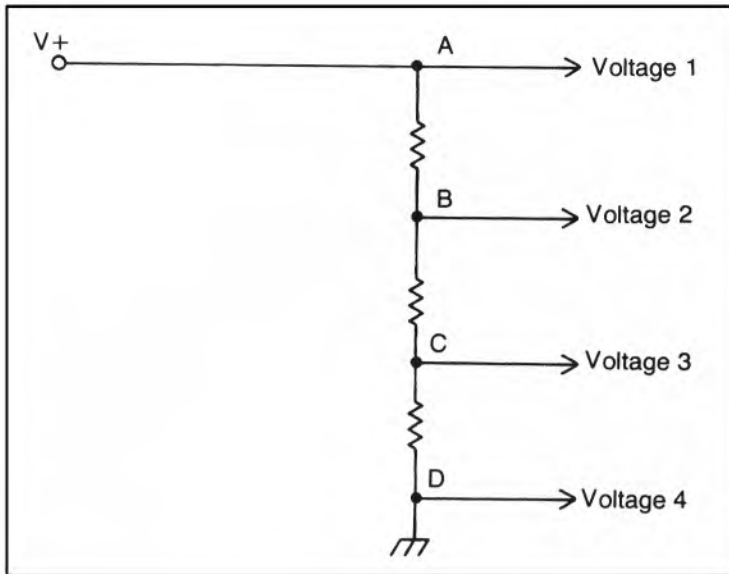


Fig. 2-15. Simple voltage divider circuit.

drop across R_1 can easily be calculated by another form of Ohm's Law (voltage equals current times resistance, or $E=IR$). Here we have $E = 0.01 \times 100 = 1$ volt. Therefore, we should be able to measure 2 volts at point B. Similarly, resistor R_2 would drop another volt, leaving 1 volt at point C. R_3 drops the remaining volt, so point D is at 0 volts, just as it should be.

The resistances in a voltage divider do not have to be the same values. This time let's say the battery is putting out 10 volts, R_1 is 100 ohms, R_2 is 220 ohms, and R_3 is 47 ohms. The total resistance in the circuit is $100 + 220 + 47$, or 367 ohms. The current flowing through the circuit is equal to E/R , or $10/367$, or about 0.027 amps (27 mA). The full 10 volts would reach point A, of course. R_1 drops 0.027×100 , or 2.7 volts, so 7.3 volts would be found at point B. R_2 has a drop of 0.027×220 , or about 5.95 volts, leaving 1.35 volts at point C. If you calculate the voltage drop across R_3 , you'll get 0.027×47 or 1.269. The error of 0.081 volt comes up because we rounded off the current value. This will rarely make a significant difference. Generally, the tolerances of the resistance values will account for larger errors anyway.

A voltage divider resistor network of this type is usually not very precise, but it is generally sufficient for most purposes, and is widely used in power supply circuits.

Remember that the circuit being powered is essentially a resistance in parallel with the voltage divider. The load resistance could throw the voltages way off in some cases. This can often be prevented by keeping the voltage divider resistances relatively small. If the voltage divider is 100 ohms and the load resistance is 10,000 ohms, the parallel combination will have a value of :

$$\frac{1}{(1/100 + 1/10000)}$$

or $1/(0.01 + 0.0001) = 1/0.0101 = 99.00990099$. That is so close that the difference should scarcely matter.

VARIABLE POWER SOURCE

Since electronic music synthesizers depend heavily on voltage control, a manually variable voltage source is usually extremely helpful, if not absolutely essential.

A variable voltage supply can be used to set voltage-controlled modules to fixed levels. It can also be used to bias an effect. This essentially sets the range of the module. If a negative constant bias voltage is applied to a voltage-controlled module, a second control voltage would have to exceed that preset level before it will have any effect on the signal.

As we start working with voltage-controlled modules in this book, we will look at how bias voltages can be used in specific situations.

Adding a manually variable voltage source to a power supply circuit is simple enough. It is simply a matter of tapping off another output from the main voltage line through a potentiometer. This is illustrated in the partial schematic in Fig. 2-16.

Positive and negative variable voltage sources are both useful.

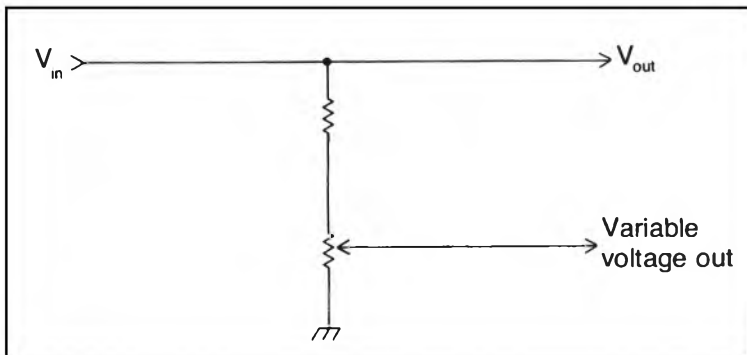


Fig. 2-16. A variable voltage source.

A SYNTHESIZER POWER SUPPLY

Figure 2-17 shows a schematic for a complete power supply that is appropriate for most of the circuits in this book. The parts list is given in Table 2-1.

This circuit uses three voltage regulator ICs for outputs at +5 volts, +12 volts, and -12 volts. Resistor voltage dividers are used to provide additional outputs at +9 volts, -9 volts, and -5 volts. Note that these outputs are not regulated. Loading effects of the circuits they power and resistor tolerances might cause some voltage fluctuation from the nominal values. However, because they are tapped off from a voltage regulator output, there should be little or no ac ripple superimposed on any of the outputs.

This power supply circuit also features a positive and a negative variable voltage source. The range for each variable voltage source is from 0 to 10 volts (at the appropriate polarity, of course). Standard potentiometers can be used because high degrees of precision and accuracy will almost never be required.

The main fixed voltage outputs should be connected to terminal strips or buses so the various modules can be permanently connected to the appropriate voltages. The variable voltage sources should be fed out through banana jacks on the front panel, so whatever temporary connections are needed may be easily and quickly made or changed with patch cords. (If you are building a hard wired system, see Chapter 10.)

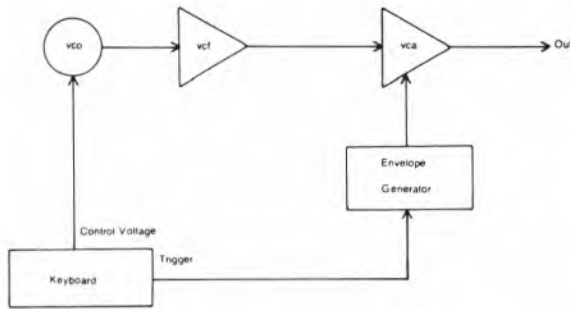
The potentiometers controlling the variable voltage sources and the power switch should also be mounted on the main control

Table 2-1. Parts List for Fig. 2-17.

R1	680 Ω resistor
R2, R7	2.2 k Ω resistor
R3-R6	1 k Ω resistor
R8	2.5 k Ω potentiometer
C1, C2	1000 μ F 35 V electrolytic capacitor
C3-C6, C9, C10	0.3 μ F capacitor
C7, C8	220 μ F 25 V electrolytic capacitor
D1-D4	1N4001 (or similar)
IC1	7805 5 V regulator IC
IC2	7812 12 V regulator IC
IC3	7912 -12 V regulator IC
T1	power transformer
	primary —120 V
	secondary —centertapped
	—at least 25 V
	—at least 3A
F1	3 A fuse and holder (do not substitute a higher value!)
S1	SPST switch

panel, of course. The power indicator lamp Π is optional, but handy. It reminds you to turn off your synthesizer when not in use and assures you that it is getting power when it is turned on. If it is used, the lamp should obviously be mounted near the power switch. If you prefer, an LED may be substituted for the lamp. It should be powered by the regulated +5 volt line.

Chapter 3



Signal Sources

Electronic music synthesizers can manipulate signals from many different sources. Guitar synthesizers are common. They use the signal generated by the pick-up of an electric guitar. Any sound that can be picked up by a microphone may also be fed into a synthesizer for manipulation. Most synthesizers, however, are entirely self-contained. That is, they have circuits for generating the primary signal. This type of circuit is usually an *oscillator*.

An oscillator is a device for generating an ac signal. Its output is a specific changing voltage pattern that is repeated at a definite rate, or frequency.

The voltage pattern generated by an oscillator is called a *waveshape*, or *waveform*. Each complete pattern is called a cycle. Different waveshapes have different characteristics and, if fed through a loudspeaker, different sounds.

A theoretically infinite number of waveshapes are possible. Fortunately, most sounds can be simulated by grouping and manipulating just a few basic waveshapes. The *sine wave* (shown in Fig. 3-1) is the simplest. Other common waveshapes are the *triangle wave* (sometimes called a *delta wave*) (Fig. 3-2), *sawtooth waves* (Fig. 3-3), and *rectangle waves* (Fig. 3-4). Rectangle waves are sometimes called *pulse waves*, especially when they are very narrow as in Fig. 3-4A. An important special form of the rectangle wave is the *square wave* (Fig. 3-4C). Each of these waveshapes and their characteristics will be discussed in this chapter.

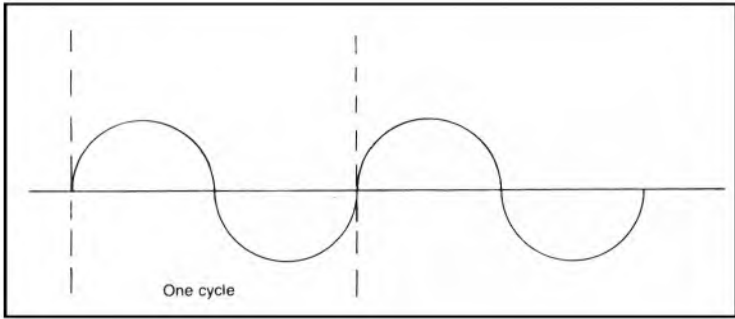


Fig. 3-1. Sine wave.

The frequency of an ac signal is measured in cycles per second, or hertz (abbreviated Hz). Obviously this simply means how many times a second the voltage pattern is repeated. A low frequency is heard as a low pitch, while a high frequency is heard as a high pitch. The human ear, depending on the individual, can hear from 20 Hz to 50 Hz at the low end of the spectrum to 12,000 Hz to 20,000 Hz at the high end.

To be useful in an electronic music synthesizer, an oscillator should have the capability to vary its output frequency. Many oscillator circuits can simultaneously generate multiple wave-shapes too. Of course, the more versatile an oscillator circuit is, the more useful it is likely to be.

In electronic music synthesizers, oscillators are usually used as signal sources, but they may also be employed as periodic control

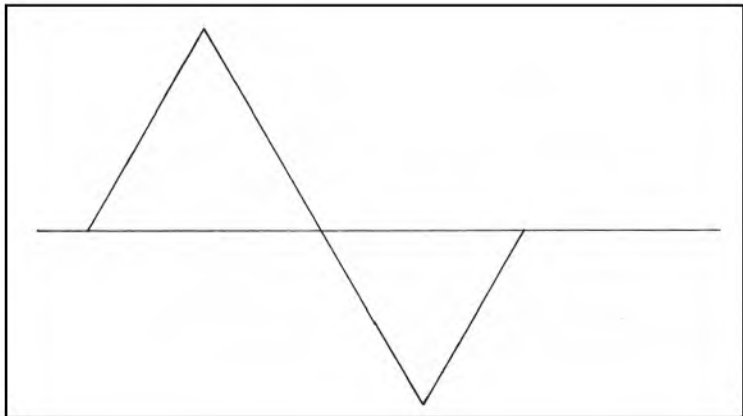


Fig. 3-2. Triangle wave.

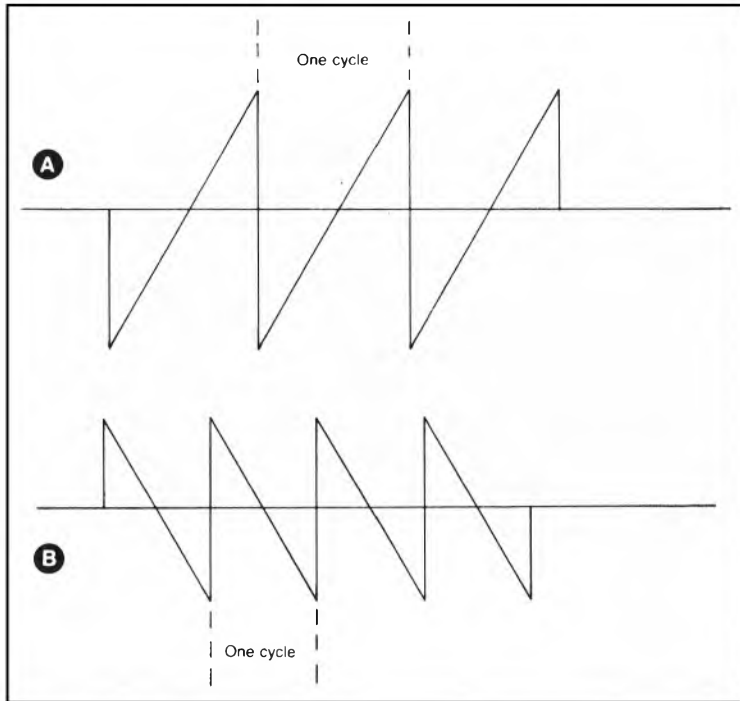


Fig. 3-3. Sawtooth waves, ascending (a) and descending (b).

voltage sources, especially at relatively low frequencies. An oscillator with a frequency of about 4 to 10 Hz can be used to create tremolo (fluctuating volume) and vibrato (fluctuating pitch) effects. Using an oscillator with a frequency in the audible range as a control voltage source leads to modulation effects and sidebands (discussed in detail in later chapters). The two most common forms of modulation are *amplitude modulation* (AM) and *frequency modulation* (FM).

BASIC OSCILLATOR CIRCUITS

In the next few pages, we will explore a number of circuits that will generate various common waveforms. In most cases, there are several methods for obtaining a specific signal. You may include any or all of these oscillators in your synthesizer. Bear in mind, however, these are manually controlled oscillators, so many effects are not convenient or possible unless these circuits are modified. Later, the concept of voltage controlled oscillators will be presented.

Manually controlled oscillators are usually fine for use as periodic control voltage sources.

Sine Wave Oscillators

The simplest waveform possible is the sine wave. This waveshape is illustrated in Fig. 3-1. If a sine wave signal is fed through a loud-speaker it produces a very pure, and somewhat piercing tone. An unmodified sine wave can be very unpleasant to listen to for extended periods.

Often the term oscillator is assumed to imply a sine wave oscillator. Circuits that produce other waveshapes are frequently referred to as waveform generators.

In electronic music, sine wave oscillators are generally employed as control voltage sources. This waveshape is ideal for creating smooth tremolo and vibrato effects.

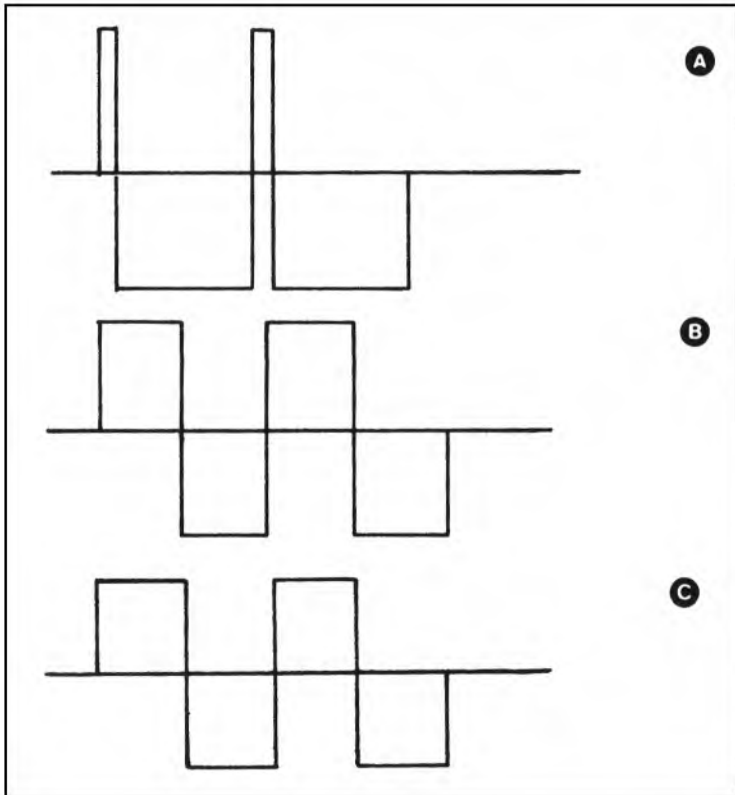


Fig. 3-4. Rectangle waves, pulse (a), rectangle (b), square (c).

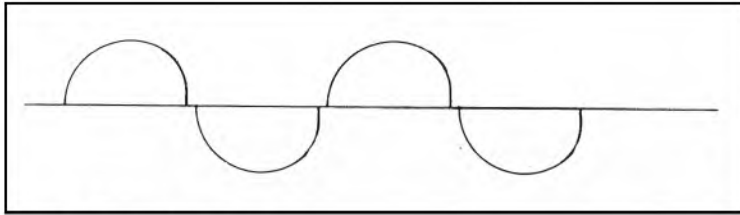


Fig. 3-5. Distorted sine wave.

A number of sine wave oscillators are often used together for additive synthesis, which will be explained in the next chapter.

Most practical circuits have some degree of distortion, or variation from the ideal waveshape. This concept is illustrated in Fig. 3-5. While this distortion adds some overtones (additional frequencies above the nominal, or fundamental frequency), it is usually very slight unless the distortion is severe. It is not difficult to build a circuit for generating a sine wave that is pure enough for electronic music needs. Any of the circuits presented in this section should do the job just fine.

Because of their relative purity, sine waves are often used as a test signal to check amplifiers and other circuits for distortion. Obviously a very good (low distortion) sine wave oscillator would be required for this type of application. Figure 3-6 shows the basic setup for this kind of testing. The oscilloscope is used to compare the output signal with the input signal. Any differences (other than amplitude) can be assumed to be due to distortion. The advantage of a dual trace oscilloscope should be obvious here. Both waveforms can be viewed at the same time.

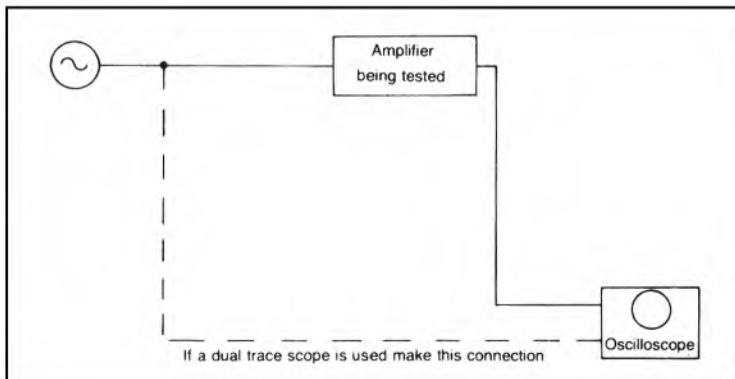


Fig. 3-6. Using a sine wave oscillator to test an amplifier for distortion.

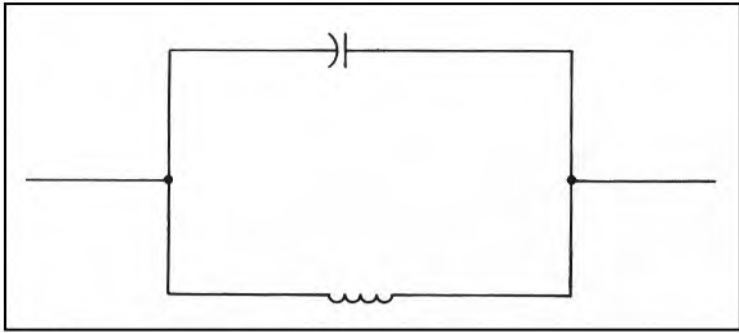


Fig. 3-7. Simple parallel resonant LC circuit.

Early sine wave oscillators were constructed around what is called a parallel resonant LC circuit. L is used to represent the inductance of a coil and C identifies the capacitance of a capacitor, so an LC circuit consists of a coil and a capacitor. A simple parallel resonant LC circuit is shown in Fig. 3-7.

The impedance (ac resistance) of a coil increases as the applied frequency increases, while the impedance of a capacitor decreases with increasing frequency. At some specific frequency, the impedance of the coil will equal the impedance of the capacitor. This is called the resonant frequency. In a parallel resonant LC circuit, the impedance is at its maximum level at the resonant frequency.

The Principle of Oscillation. To understand the principle of oscillation, let's consider the simplified circuit shown in Fig. 3-8. (This is not a practical circuit—it is included here for demonstration purposes only.) As long as the switch is closed, nothing much happens. When the switch is suddenly opened, however, the current through L1 starts to drop. If you are familiar with basic electronics theory you should recall that a changing current through a

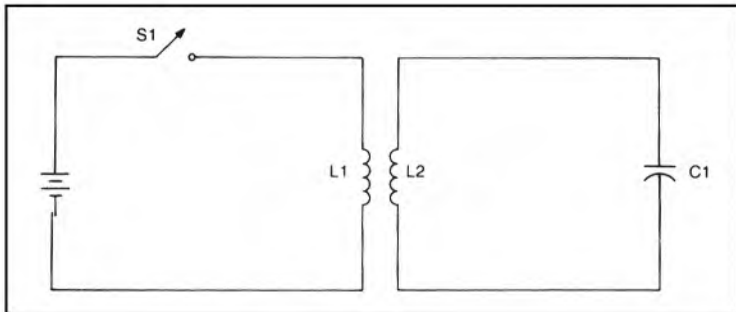


Fig. 3-8. Simplified LC oscillator.

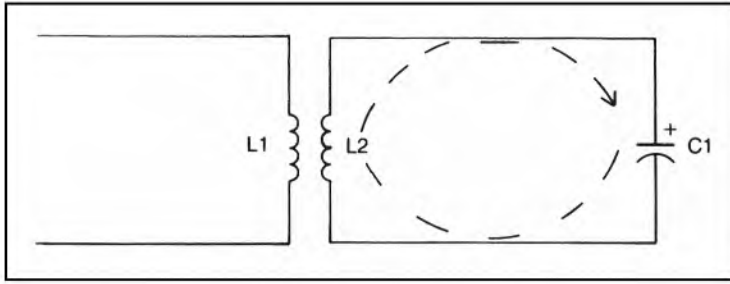


Fig. 3-9. Current flow during first half cycle.

coil creates a magnetic field, which will induce a voltage in the nearby L2. This induced voltage will charge the capacitor as illustrated in Fig. 3-9.

Before long, the current flowing through L1 will drop to zero, of course. The magnetic field will collapse, and no further voltage will be induced in L2. This allows the capacitor to begin discharging through L2, as shown in Fig. 3-10. This current flowing through the coil will start up more mutual induction between L1 and L2, and the capacitor will be recharged with the opposite polarity. When this happens, current stops flowing and the entire cycle is repeated.

This process (theoretically) generates a very pure sine wave signal in the parallel resonant LC circuit. Of course, resistances within the circuit will gradually reduce the amplitude of the signal. Each cycle will be somewhat lower, until the signal dies out entirely. This effect, referred to as damping, is illustrated in Fig. 3-11.

The frequency of the oscillations will be determined by the

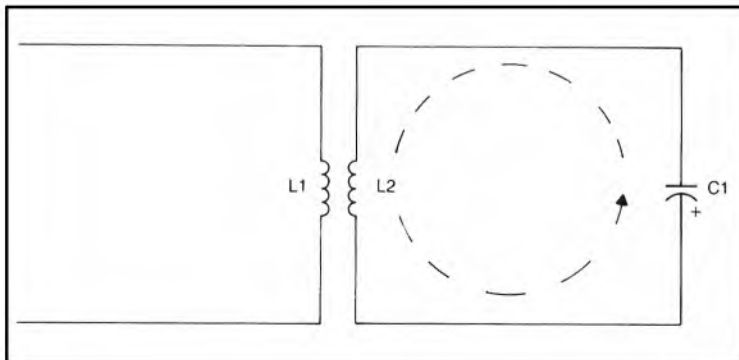


Fig. 3-10. Current flow during second half cycle.

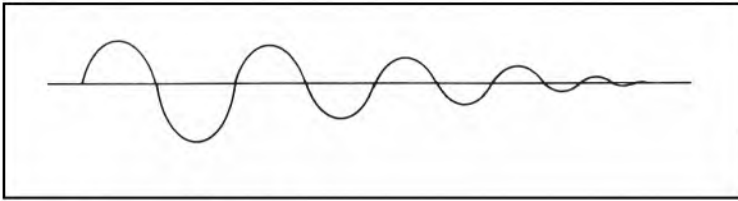


Fig. 3-11. Damped sine wave.

values of the coil and capacitor. The formula for finding the frequency of oscillations (the resonant frequency) is as follows:

$$F = \frac{1}{2 \pi \sqrt{LC}}$$

F represents the frequency in hertz, L represents the inductance of the coil in henries, and C is the capacitance in farads, π (pi) is a mathematical constant that is equal to approximately 3.14. (This rounded off value of pi is more than close enough for electronic purposes. Component value tolerances render further precision rather pointless. In fact, some technicians simply use 3 for π when working out calculations.)

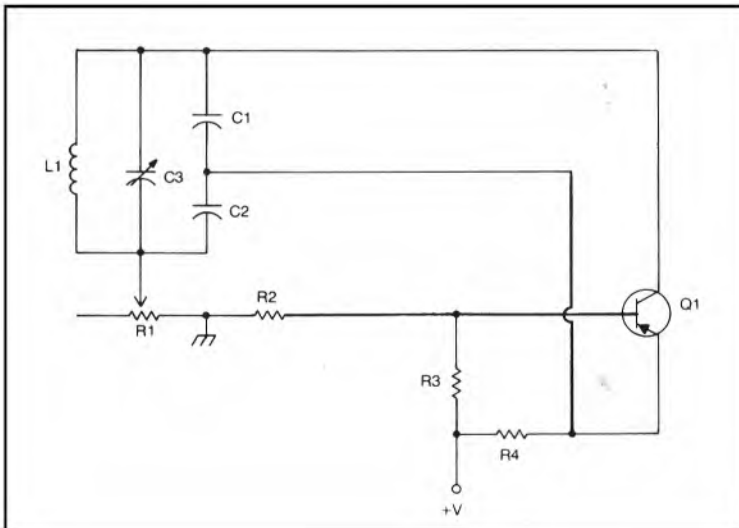


Fig. 3-12. Colpitts oscillator.

Two common sine wave oscillator circuits are the Colpitts oscillator and the Hartley oscillator. These two circuits are really quite simple. They consist primarily of a simple amplifier (to keep the oscillations from dying out) and a parallel resonant LC circuit.

Colpitts Oscillator. A Colpitts oscillator taps off the oscillation signal by using a split capacitance. The schematic of a typical Colpitts oscillator is shown in Fig. 3-12. In determining the resonant frequency, the three capacitors are treated as a single capacitor. The formula for determining the capacitance of two capacitors in series (C1 and C2) is as follows.

$$\frac{1}{C_s} = \frac{1}{C1} + \frac{1}{C2}$$

This can then be added to the third capacitor (C3) which is in parallel.

$$C_T = C_s + C3$$

C3 is a variable capacitor to allow convenient manual adjustment of the resonant (oscillation) frequency.

Hartley Oscillator. The Hartley oscillator is very similar to the Colpitts oscillator. The primary difference in the Hartley oscillator is that the signal is tapped off from a split inductance, rather than from a split capacitance as in the Colpitts circuit. A schematic diagram for a typical Hartley oscillator circuit is shown in Fig. 3-13.

While LC oscillator circuits are relatively simple to design and work with, they do have a number of significant short-comings. First off, coils and capacitors are often rather bulky physically, especially when relatively large values are required. Also their values are rarely precise. Their true value may deviate from the nominal value by as much as 25 percent (or even more with some components). This, of course, makes it difficult to obtain a precise, specific frequency. Moreover, standard components often fluctuate somewhat in value with variations in temperature. This can cause the oscillator's output frequency to drift somewhat as the circuit warms up.

Crystal Oscillator. When greater precision is required, crystal oscillators are often used. In electronics terminology, a crystal (often abbreviated as XTAL) is made up of a thin slab of crystalline quartz sandwiched between two metallic plates. A crystal can be used in place of a resonant LC circuit.

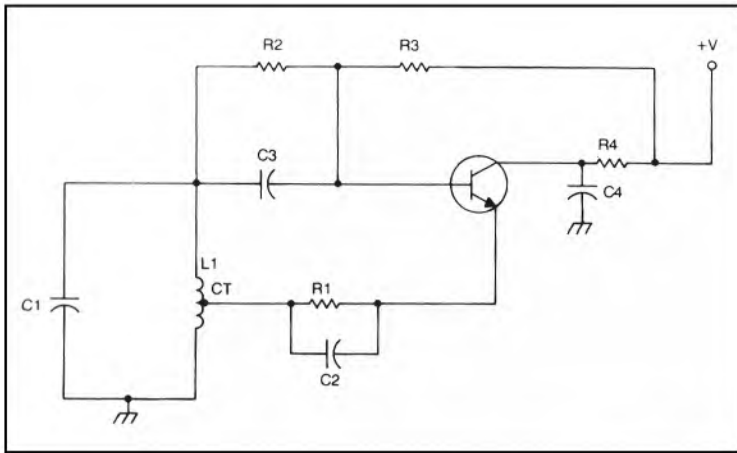


Fig. 3-13. Hartley Oscillator.

The secret of a crystal's operation lies in the *piezoelectric effect*. A crystal has two sets of axes. The x-axis passes through the corners of the crystal, while the y-axis is perpendicular to (but in the same plane as) the x-axis. A mechanical stress along the y-axis will cause an electrical voltage to be generated along the x-axis. Conversely, an electrical voltage applied to the x-axis will create a mechanical stress along the y-axis. This is the piezoelectric effect.

When a crystal is placed in an electronic circuit, it behaves like a coil, a resistor, and a pair of capacitors connected as shown in the equivalent circuit of Fig. 3-14.

A typical crystal oscillator circuit is shown in Fig. 3-15. Crystal oscillators can hold a frequency very precisely, especially if the crystal is kept at a constant temperature. In broadcasting applica-

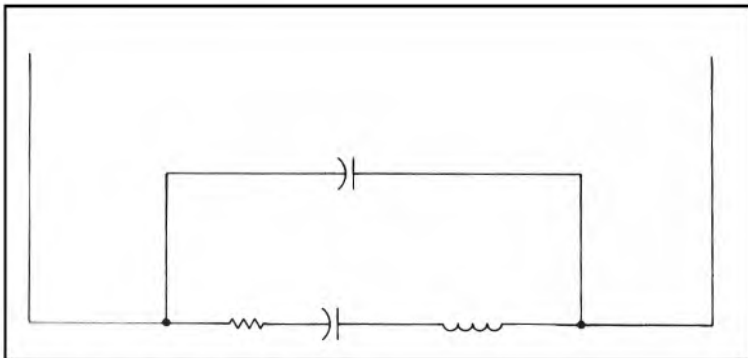


Fig 3-14. Equivalent circuit of a crystal.

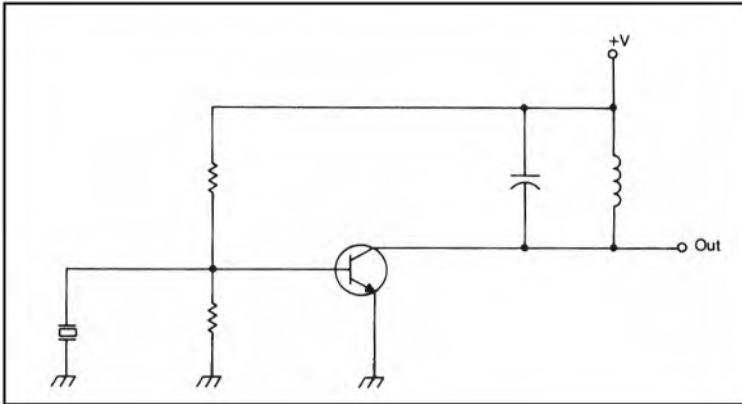


Fig. 3-15. Typical crystal oscillator.

tions, crystals are often operated from within special temperature controlled ovens. In electronic music, this degree of precision is somewhat superfluous. The human ear is simply not that discriminating.

The frequency of a crystal oscillator is determined by the size and shape of the quartz slab. Obviously it is not very easy to make a crystal oscillator variable over a wide range of frequencies.

Integrated Circuits. These days it is often more convenient to employ integrated circuits, rather than discrete components. A number of special purpose, dedicated oscillator chips are available, but general purpose devices may also be used. For generating sine waves, op amps (operational amplifiers) do a reasonably good job with just a handful of resistors and capacitors. No bulky (and often hard to find) coils are required.

Figure 3-16 shows a fairly simple, but adequate sine wave oscillator circuit built around an op amp IC. Almost any commonly available op amp device may be used in this application, especially the popular 741 or 748. A high precision device is not necessary.

Three parts lists are given for this circuit (Tables 3-1 through 3-4), each for a specific frequency. The resistor and capacitor values are interrelated. The output frequency is determined by the following formula:

$$F = \frac{1}{2 \pi RC}$$

Where

$$R = R1 = R2 = 2R3 = 4R4 = R5 \text{ and } C = C2 = C3 = C1$$

R3 is a variable resistor (potentiometer) to allow some variation in the output frequency. This is actually a fine tuning control. This circuit does not have a wide frequency range for any specific combination of components.

Rectangle Wave Oscillators

Probably the most popular class of waveforms in modern electronic music is the rectangle wave (refer back to Fig. 3-4). Rectangle waves are very rich in harmonics, and are usually quite pleasant to listen to even by themselves. They are considered a complex waveform.

A complex waveform (anything other than a pure sine wave) consists of a number of simultaneous frequencies in a specific

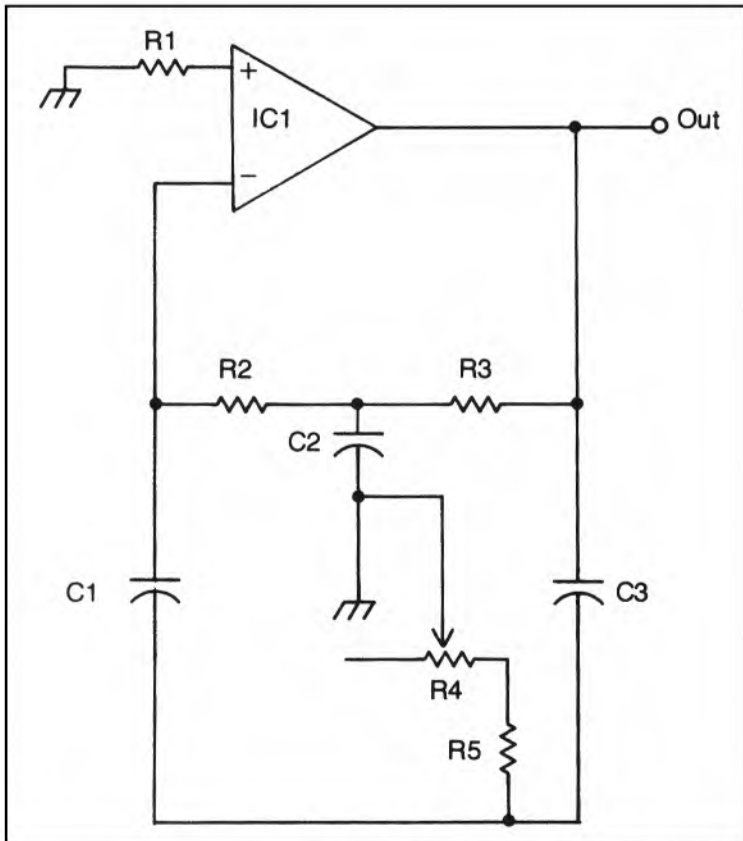


Fig 3-16 Op amp sine wave oscillator.

Table 3-1. Parts List for the Op Amp Sine Wave Oscillator in Fig. 3-16

R1	33 k Ω resistor
R2, R3	15 k Ω resistor
R4	10 k Ω potentiometer
R5	3.9 k Ω resistor
C1, C3	0.1 μ F capacitor
C2	0.2 μ F capacitor
IC1	741 or similar op amp IC

combination. The nominal frequency, or perceived pitch is called the *fundamental frequency*. Additional frequencies higher than the fundamental frequency are called *overtones*. Some complex waveforms also contain a few frequencies that are lower than the fundamental frequency. Not surprisingly, these are called *under-*

Table 3-2. Parts List for the Op Amp Sine Wave Oscillator in Fig. 3-16. The Output Frequency is 1000 Hz.

R1	3.3 k Ω resistor
R2, R3	1.5 k Ω resistor
R4	1 k Ω potentiometer
R5	390 Ω resistor
C1, C3	0.1 μ F capacitor
C2	0.2 μ F capacitor
IC1	741 or similar op amp IC

tones. In most waveforms, the fundamental frequency is the lowest and strongest element.

In the standard waveshapes (shown in Fig. 3-1 through 3-4), all of the overtones are exact multiples of the fundamental frequency. This type of overtone is called a *harmonic*. Each harmonic is identified by its relationship to the fundamental frequency. For exam-

Table 3-3. Parts List for the Op Amp Sine Wave Oscillator in Fig. 3-16. The Output Frequency is 2000 Hz.

R1	1.5 k Ω resistor
R2, R3	820 Ω resistor
R4	500 Ω potentiometer
R5	180 Ω resistor
C1, C3	0.1 μ F capacitor
C2	0.2 μ F capacitor
IC1	741 or similar op amp IC

Table 3-4. Parts List for the Op Amp Sine Wave Oscillator in Fig. 3-16. The Output Frequency is 5000 Hz.

R1	6.2 kil	resistor
R2, R3	3.3 K Ω	resistor
R4	1.5 k Ω	potentiometer
R5	820 Ω	resistor
C1, C3	0.05	capacitor
C2	0.1 /xF	capacitor
IC1	741	or similar op amp IC

pie, if the fundamental frequency is 100 Hz, the second harmonic is 200 Hz, the third harmonic is 300 Hz, the 4th harmonic is 400 Hz, the fifth harmonic is 500 Hz, and so forth. Additional examples are summarized in Table 3-5.

Each waveshape contains a specific set of harmonics in a definite amplitude relationship to each other and the fundamental frequency.

In an ideal rectangle wave, the voltage switches instantly between a high and a low level, with no intermediate levels. Of course practical circuits never quite meet the ideal (although some can come quite close). In a practical rectangle wave oscillator circuit, a finite amount of time is required for the output to switch between levels. This is called the *slew rate*. Figure 3-17 is an exaggerated illustration of this effect.

Table 3-5. Harmonics for Various Fundamentals.

Fundamental	250 Hz	370 Hz	1020 Hz
Second	500 Hz	740 Hz	2040 Hz
Third	750 Hz	1110 Hz	3060 Hz
Fourth	1000 Hz	1480 Hz	4080 Hz
Fifth	1250 Hz	1850 Hz	5100 Hz
Sixth	1500 Hz	2220 Hz	6120 Hz
Seventh	1750 Hz	2590 Hz	7140 Hz
Eighth	2000 Hz	2960 Hz	8160 Hz
Ninth	2250 Hz	3330 Hz	9180 Hz
Tenth	2500 Hz	3700 Hz	10200 Hz
Eleventh	2750 Hz	4070 Hz	11220 Hz
Twelfth	3000 Hz	4440 Hz	12240 Hz
Thirteenth	3250 Hz	4810 Hz	13260 Hz
Fourteenth	3500 Hz	5180 Hz	14280 Hz
Fifteenth	3750 Hz	5550 Hz	15300 Hz
Sixteenth	4000 Hz	5920 Hz	16320 Hz
Seventeenth	4250 Hz	6290 Hz	17340 Hz
Eighteenth	4500 Hz	6660 Hz	18360 Hz
Nineteenth	4750 Hz	7030 Hz	19380 Hz
Twentieth	5000 Hz	7400 Hz	beyond audible range

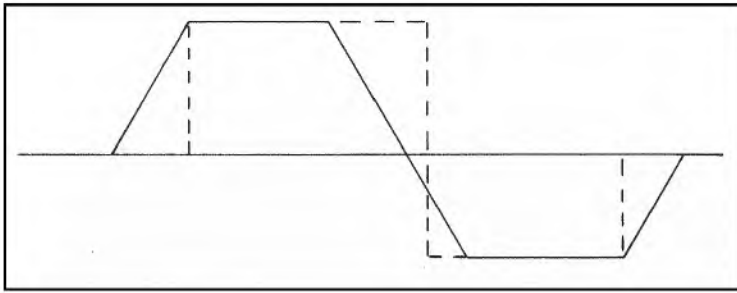


Fig. 3-17. Rectangle wave with an exaggerated slew.

Another shortcoming of practical rectangle wave oscillator circuits is illustrated (again in exaggerated fashion) in Fig. 3-18. This is called *over-shoot*. Still another common form of distortion in rectangle waves is called *ringing*. This is illustrated in Fig. 3-19.

Each of these distortions of the ideal waveshape have a subtle effect on the harmonic content of the signal. Fortunately, modern circuits are good enough that the effects will rarely be audible. Still, you should be aware of the possibilities of distortion.

Various rectangle waves are identified by their *duty cycle*. A rectangle wave's duty cycle is the ratio between its high voltage time to total cycle time. For instance, if the output voltage is at a high level for one third of every complete cycle, the duty cycle is 1:3. If the output voltage level is high for one quarter of the complete cycle, the duty cycle is 1:4. When the duty cycle is 1:2 (the output voltage level is high for exactly half of each complete cycle), we have the special (and extremely useful) form of the rectangle wave called the square wave. If a pure square wave is fed

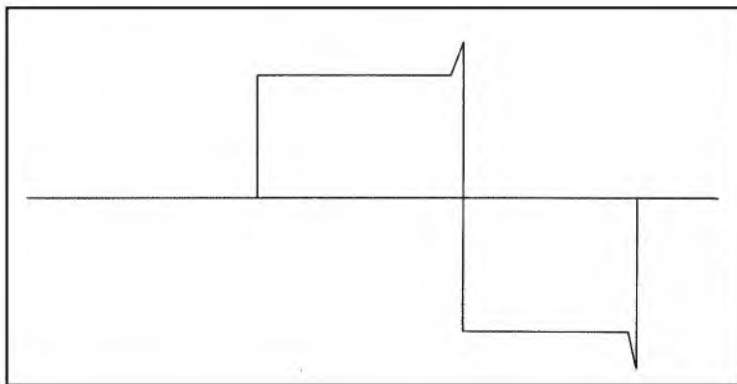


Fig. 3-18. Rectangle wave with an exaggerated overshoot.

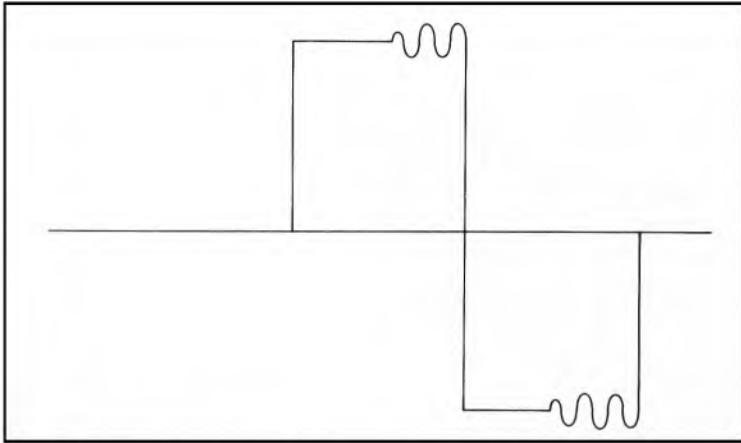


Fig. 3-19. Rectangle wave with an exaggerated ringing.

through a loudspeaker, a mellow reedy sound, somewhat suggestive of an oboe, will be heard.

The duty cycle of a rectangle wave determines the harmonic content of the signal. All harmonics that are multiples of the second number in the duty cycle figure are absent. All other harmonics are present. For example, if the duty cycle is 1:2 (square wave) all even (every second) harmonics are absent from the signal. If the duty

Table 3-6. Harmonic Content of Rectangle Waves of Various Duty Cycles.

1:2	1:3	1:4	1:5
Fundamental	Fundamental	Fundamental	Fundamental
Third	Second	Second	Second
Fifth	Fourth	Third	Third
Seventh	Fifth	Fifth	Fourth
Ninth	Seventh	Sixth	Sixth
Eleventh	Eighth	Seventh	Seventh
Thirteenth	Tenth	Ninth	Eighth
Fifteenth	Eleventh	Tenth	Ninth
Seventeenth	Thirteenth	Eleventh	Eleventh
Nineteenth	Fourteenth	Thirteenth	Twelfth
	Sixteenth	Fourteenth	Thirteenth
	Seventeenth	Fifteenth	Fourteenth
	Nineteenth	Seventeenth	Sixteenth
	Twentieth	Eighteenth	Seventeenth
		Nineteenth	Eighteenth
			Nineteenth

cycle is 1:3, every third harmonic is missing. If the duty cycle is 1:4, every fourth harmonic is absent. This concept is presented more thoroughly in Table 3-6.

Clearly rectangle waves of varying duty cycles can produce a wide variety of tonal qualities, even without further modulation and synthesis. Thousands of circuits for rectangle wave oscillators have been developed. We will look at a few in the next several pages.

Two Transistor Oscillator. Figure 3-20 shows the schematic for a simple two transistor rectangle wave oscillator. The two transistors may be virtually any general purpose NPN type transistor, such as 2N3904 or 2N2222. Notice how each transistor is in a mirror image circuit. When one transistor is saturated (conducting at its maximum level), the other is cutoff. This state will be held for a specific time determined by the associated RC circuit, then the transistors will very rapidly reverse states. A rectangle wave signal may be tapped off from the collector of either transistor. Remember the two collectors will be 180° out of phase with each other.

For a square wave (1:2 duty cycle) mirror image components should be of equal value. That is, $R1 = R4$, $R2 = R3$, and $C1 = C2$.

A wide variety of component values may be used in this circuit, so a definite parts list is not given here.

$R1$ and $R4$ should be in the 5 k Ω to 100 k Ω range, and $R2$ and $R3$ should be between 2 k Ω and 5 k Ω . The capacitors should work with any value from about 0.05 μ F up to 0.5 μ F. Nothing is terribly critical in this circuit.

One major disadvantage with this circuit for electronic music

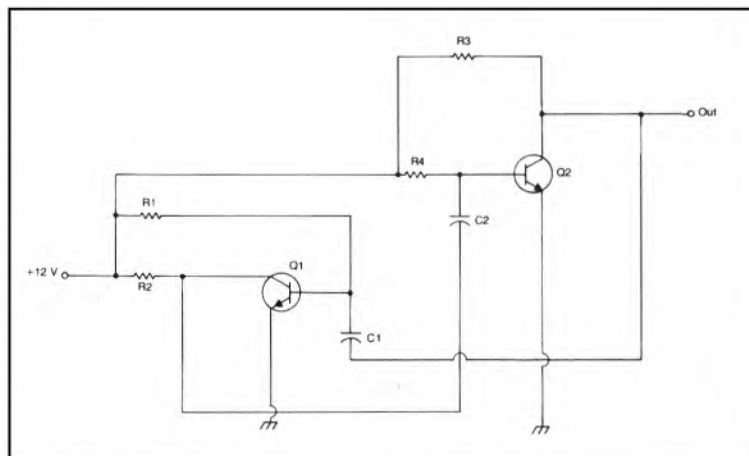


Fig. 3-20. Two transistor rectangle wave oscillator.

Table 3-7. Parts List for the Op Amp Square Wave Oscillator in Fig. 3-21.

R1, R2	10 k Ω resistor
R3	4.7 k Ω resistor
R4	500 k Ω potentiometer
C1	0.1 μ F capacitor
IC1	almost any op amp (741 or similar)

purposes is that the output frequency (and/or duty cycle) can not be easily changed while the oscillator is in operation. You could use a dual potentiometer for R1/R4, but this is only a fair solution at best.

Op Amp Oscillator. A more versatile square wave oscillator is shown in Fig. 3-21. This circuit is built around an op amp IC. Any common op amp device may be used (741, 748, etc.). A high precision unit is not required. The internal noise of the IC chip will be drowned out by the strong generated signal.

A parts list for this circuit is given in Table 3-7. The output frequency may be easily changed by readjusting the potentiometer.

This circuit will generate square waves only. Rectangle waves with other duty cycles can not be obtained without modifying the circuit. Figure 3-22 shows a simple modification for generating duty cycles other than 1:2. The diode will only allow the signal to pass through the second potentiometer when the output is of the correct polarity. At all other times, the feedback signal will flow through the

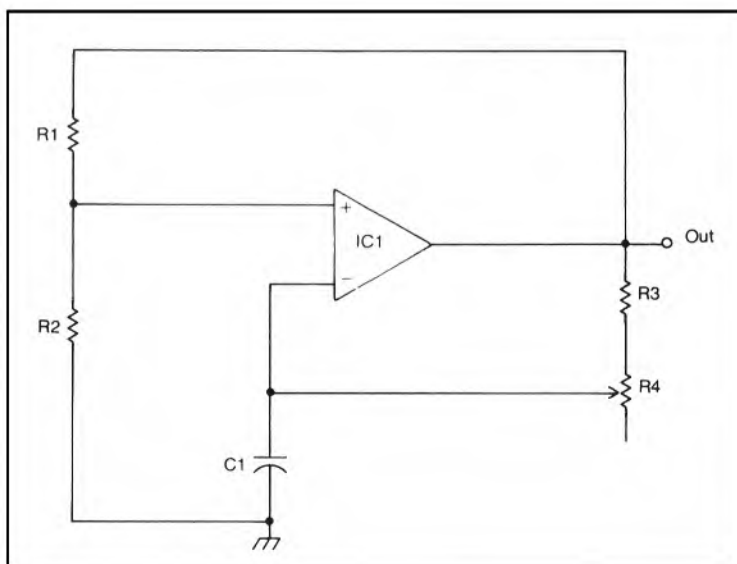


Fig. 3-21. Op amp square wave oscillator.

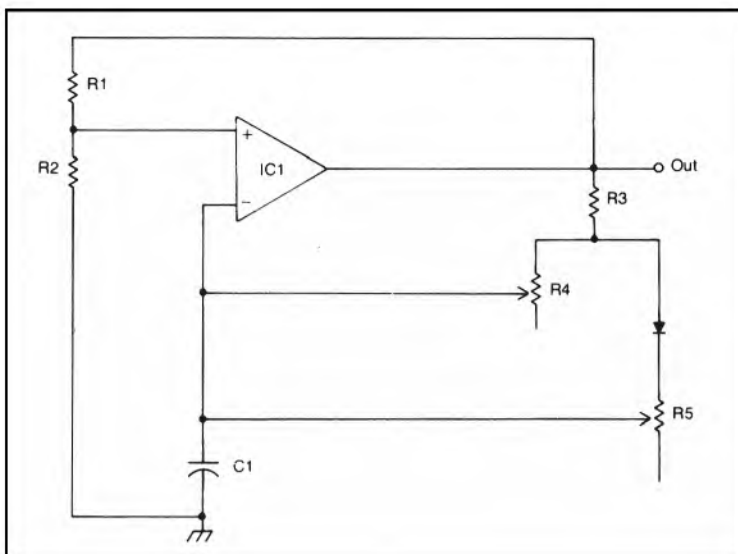


Fig. 3-22. Adapting the circuit of Fig. 3-21 for variable duty cycles.

original potentiometer (R4). One important restriction on this circuit is that the R5/diode combination (when forward biased) should have a lower resistance than R4. Unless this is the case, circuit operation will be unreliable.

Unfortunately, the output frequency of this circuit is dependent on both R4 and R5. That is, when the duty cycle is changed, so is the frequency. This is because the resistors determine the absolute time of each output level (high or low), not a percentage of the total cycle time. In addition, to change the frequency without changing the duty cycle (and hence, the tonal quality of the signal) both potentiometers must be carefully readjusted. Their action is interdependent.

Timer IC Oscillator. Timer ICs, such as the popular 555 device, are also useful in making rectangle wave oscillators. A simple circuit built around the 555 chip is shown in Fig. 3-23. The frequency is defined by the two resistors and C1. C2 should be about $0.01 \mu F$, regardless of the output frequency. The duty cycle of the output signal is determined by the ratio of R2 to the sum of R1 + R2. A true square wave is not possible.

A parts list is not included for this circuit because IC1 (555 IC) and C2 (0.01 μF capacitor) are the only fixed parts. The other three components (R1, R2, and C1) are selected for the desired output. The formula is as follows:

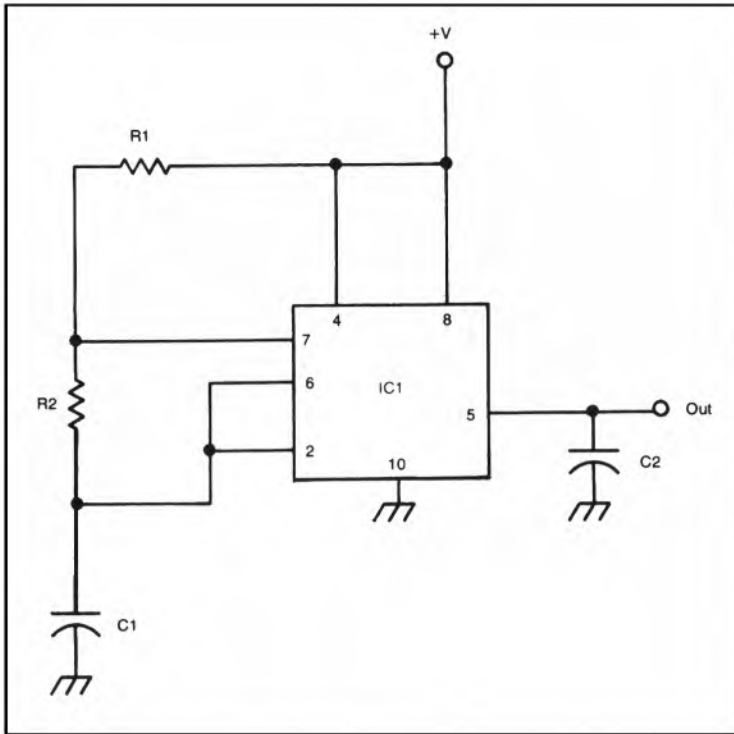


Fig. 3-23. 555 timer rectangle wave oscillator.

$$F = \frac{1.44}{(R1+2R2)C1}$$

The duty cycle can be calculated by determining the following ratio:

$$R2:(R1+2R2)$$

Figure 3-24 shows a more deluxe rectangle wave oscillator. This circuit is built around two 555 timer ICs. A 556 dual timer may be substituted, if you prefer. The parts list for this rectangle wave oscillator is given in Table 3-8.

Both frequency and duty cycle are adjustable in this circuit. Potentiometer R1 determines the frequency, while potentiometer R4 controls the duty cycle. The nice thing about this circuit is that the frequency and the duty cycle may be set independently of each

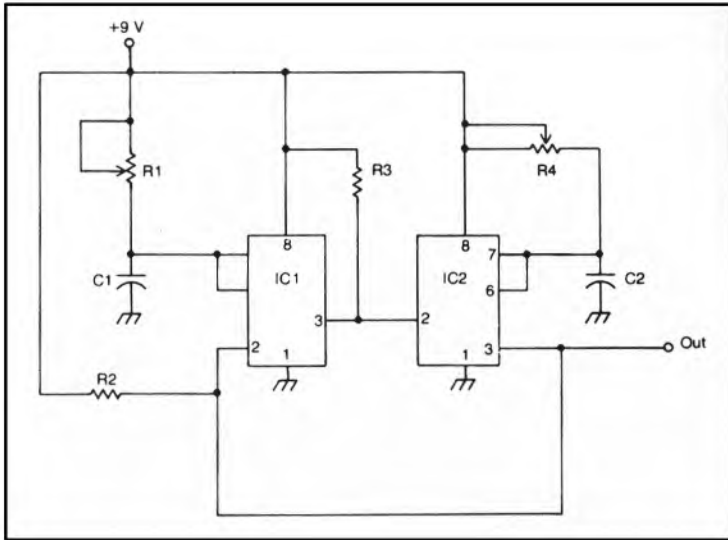


Fig. 3-24. Deluxe 555 timer rectangle wave oscillator.

other. In the simpler circuits shown earlier, these two factors are interrelated, and one can't be changed without affecting the other. The circuit of Fig. 3-24 is, therefore, far more suitable for electronic music work. The duty cycle (and therefore the harmonic content and the sound) of the signal can be easily varied while a constant pitch is held. No critical tune-up is required.

Figure 3-25 shows an external pulse width controller that can be added on to almost any square wave oscillator to add independently variable duty cycle capabilities. Use this circuit only with rectangle waves as the input signal. Other waveshapes could cause unpredictable results. It shouldn't hurt anything, but it is likely to sound unpleasant and be difficult to control. The parts list for the duty cycle controller circuit is given in Table 3-9.

Digital Logic Oscillators. High quality square waves may also be generated through digital logic circuits. Figure 3-26 shows a

Table 3-8. Parts List for the Deluxe 555 Rectangle Wave Oscillator in Fig. 3-24.

R1, R4	100 k Ω potentiometer
R2, R3	3.3 k Ω resistor
C1	0.1 μ F capacitor
C2	0.01 μ F capacitor
IC1, IC2	555 timer IC

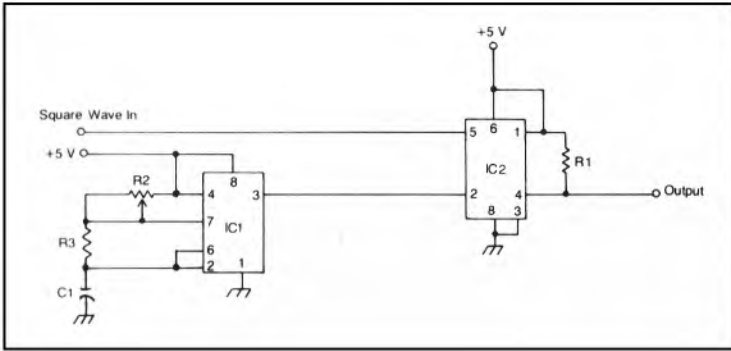


Fig. 3-25. Duty cycle controller.

typical oscillator built around three NAND (Not AND) gates. Any logic family (TTL, CMOS, or whatever) could be used. Since all the NAND gates have their inputs tied together, simple inverters could be substituted.

While functional, this type of circuit isn't entirely suitable for electronic music applications. This circuit was designed for operation in the megahertz (millions of cycles per second) range, which is far above the audible region.

Triangle Wave Oscillators

The triangle wave (sometimes called the delta wave) (see Fig. 3-2) is, in a manner of speaking, a cross between a sine wave and a square wave. Like the square wave, the triangle wave consists of the fundamental and all of the odd harmonics (third, fifth, seventh, ninth, and so forth). These harmonics, however, are extremely weak compared to the fundamental, so a triangle wave can often be substituted for a sine wave unless the application is critical. A low pass filter (see Chapter 5) can improve the resemblance to a sine wave. A high pass filter, on the other hand, can weaken the fundamental, increasing the resemblance to a square wave.

Integrator. Many commercial synthesizers obtain triangle waves by passing square waves through a special low pass filter

Table 3-9. Parts List for the Duty Cycle Controller in Fig. 3-25

R1, R3	4.7 k Ω resistor
R2	100 k Ω potentiometer
C1	5000 pF capacitor
IC1	TL507 A/D converter IC
IC2	555 timer IC

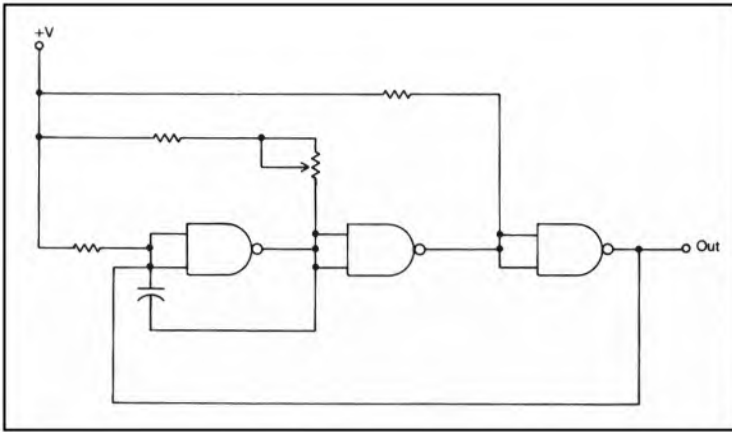


Fig. 3-26. Digital square wave oscillator.

circuit, called an integrator. Such a circuit is shown in Fig. 3-27. Table 3-10 is the parts list.

Op Amp Oscillator. Figure 3-28 illustrates a more advanced, dedicated triangle wave oscillator using op amps. The potentiometer (R5) offers manual control of the output frequency. The parts list is given in Table 3-11.

555 IC Oscillator. The 555 timer IC can also be used to generate a reasonably pure triangle wave. A circuit for this is illustrated in Fig. 3-29. The output frequency is adjusted via the potentiometer (R1). The parts list for this circuit is given in Table 3-12. This is one of the simplest possible triangle wave oscillator circuits. Only five components are needed in addition to the 555 timer IC itself.

Digital IC Oscillator. Digital circuits are usually thought of only in terms of square waves, but Fig. 3-30 shows a circuit for

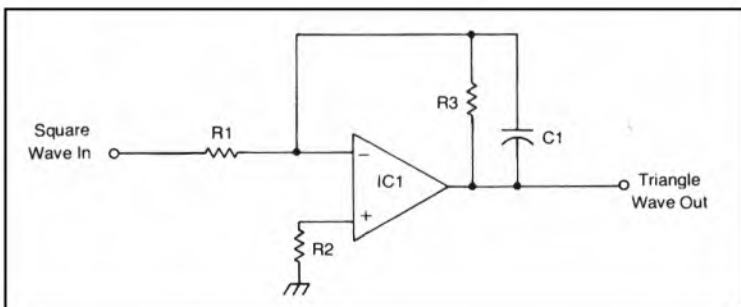


Fig. 3-27. Integrator.

R1, R2	10 kΩ resistor
R3	100 kΩ resistor
C1	0.1 μF capacitor
IC1	741, or similar op amp IC

Table 3-10. Parts List for the Integrator in Fig. 3-27.

generating a fair approximation of a triangle wave using three digital inverters. The parts list is given in Table 3-13.

Since a CMOS circuit is called for in the parts list, let's take a slight detour to mention some of the necessary precautions that

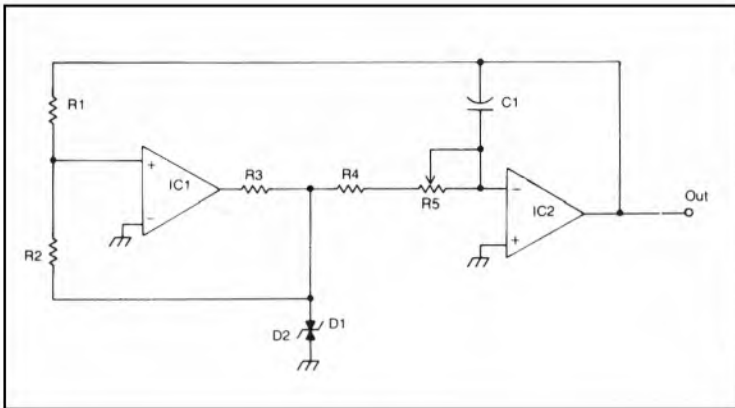


Fig. 3-28. Triangle wave oscillator.

should be followed when working with CMOS ICs. These devices are prone to damage by bursts of static electricity. Newer units are harder than the older devices, but care should still be taken. All of the pins should be shorted together until the chip is to be actually inserted in the circuit. CMOS ICs are usually sold and shipped with their pins stuck in a piece of conductive foam and/or wrapped in

R1	8.2 kΩ resistor
R2	470 kΩ resistor
R3	10 kΩ resistor
R4	39 kΩ resistor
R5	100 kΩ potentiometer
C1	0.1 μF capacitor
IC1, IC2	101, or similar, op amp IC

Table 3-11. Parts List for the Triangle Wave Oscillator in Fig. 3-28.

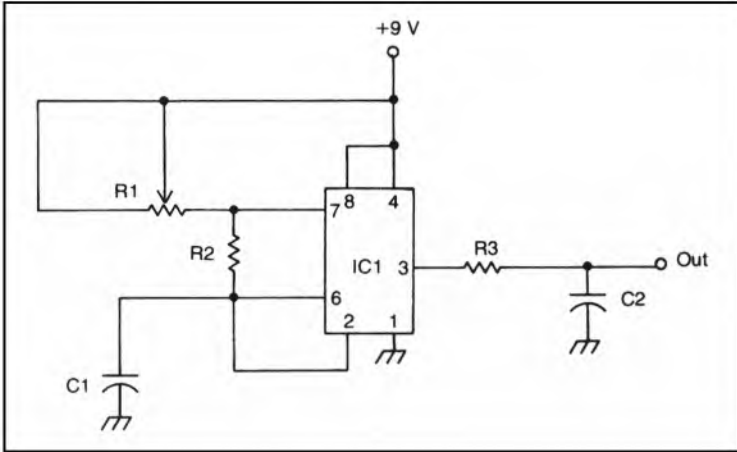


Fig. 3-29. 555 triangle wave oscillator.

aluminum foil. Leave them that way until you are ready to use the device.

Sockets are always a good idea when working with ICs. Expensive ICs can be easily damaged by too much heat while soldering. Even when working with cheap ICs like 741 op amps, a socket can be worthwhile. If you have to replace the chip for any reason, you won't have to desolder all those little pins and increase the risk of solder bridges.

Use of sockets is especially desirable when working with CMOS ICs, since they are particularly delicate. If you do decide to solder a CMOS IC directly, use only a grounded soldering iron to avoid the risk of static discharge. It would also be a good idea to ground yourself.

If a triangle wave is fed directly to a loudspeaker, the resulting sound should have a smooth, flute-like character.

Sawtooth Wave Oscillators

Sawtooth waves are very rich in strong harmonics. They include all of the harmonics, both even and odd. The harmonics are

R1	100 kΩ potentiometer
R2	1 kΩ resistor
R3	12 kΩ resistor
C1	0.1 μF capacitor
C2	0.022 μF capacitor
IC1	555 timer IC

Table 3-12. Parts List for the 555 Triangle Wave Oscillator in Fig. 3-29.

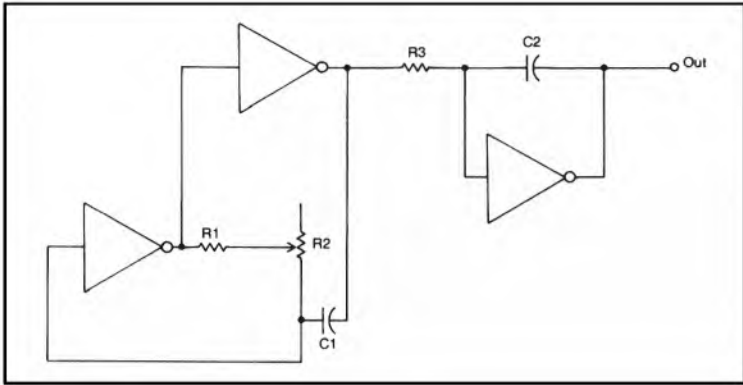


Fig. 3-30. Digital triangle wave oscillator.

relatively strong, with respect to the fundamental, although the fundamental is still strong enough to give the signal a definite sense of pitch. Fed through a loudspeaker, a sawtooth produces a very raspy sort of sound.

There are two basic types of sawtooth waves—ascending (refer back to Fig. 3-3A), and descending (Fig. 3-3B). The difference in sound between the two signals is fairly subtle, although having both available can come in handy when using a sawtooth wave signal as a control voltage. The circuits in this section generate ascending sawtooth waves. The descending form can easily be obtained by passing an ascending signal through a voltage inverter (see Chapter 9).

Op Amp Oscillator. Figure 3-31 shows the schematic for a sawtooth wave oscillator built around three standard op amp ICs.

Notice that the sawtooth wave requires a more complex circuit than the rectangle wave or triangle wave. For this reason, sawtooth wave oscillators are often omitted in inexpensive synthesis systems.

Table 3-13. Parts List for the Digital Triangle Wave Oscillator in Fig. 3-30.

R1	4.7 k Ω resistor
R2	100 k Ω potentiometer
R3	22 k Ω resistor
C1	0.022 μ F capacitor
C2	0.047 μ F capacitor
IC1	$\frac{1}{2}$ (3 sections) CD4049 hex inverter CMOS IC

Circuit should be powered by 9 or 12 volts.

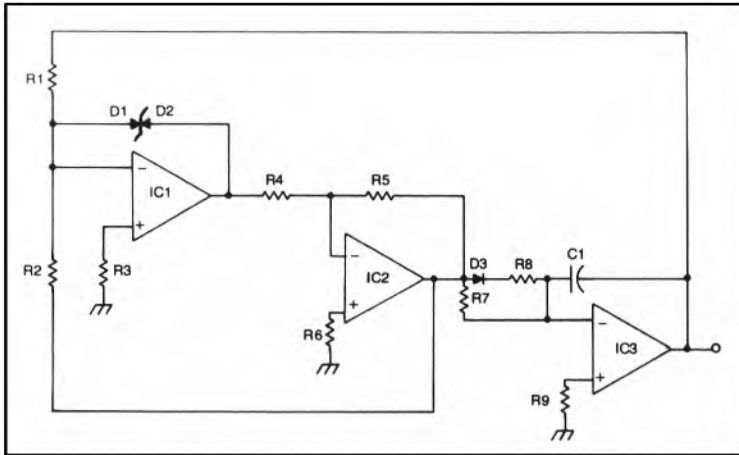


Fig. 3-31. Op amp sawtooth wave oscillator.

Simpler circuits may be used to generate sawtooth waves, but the waveform will not be as pure. In many cases, this lack of purity will not be a problem.

Digital IC Oscillator. Figure 3-32 shows a sawtooth wave oscillator built around TTL type digital circuits. The parts list is given in Table 3-14. IC1 generates a high speed square wave, which IC4 and capacitor C2 convert into an audio frequency ascending sawtooth wave signal. IC2 and IC3 provide frequency division.

UJTs and a FET. Another sawtooth wave oscillator circuit is shown in Fig. 3-33. This circuit uses two UJTs (uni-junction transistors) and FET (field-effect transistor). The potentiometer allows the output frequency to be changed.

While this circuit is useful in electronic music synthesis, it does not produce a very pure sawtooth wave signal. Figure 3-34 shows roughly what a typical output signal from this circuit might

Table 3-14. Parts List for the Digital Sawtooth Wave Oscillator in Fig. 3-32.

R1, R4, R5,	2.2 k Ω resistor
R6, R7, R8, R12	2.5 k Ω potentiometer
R2	1 k Ω resistor
R3, R9, R10, R11	0.001 μ F capacitor
C1	0.22 μ F capacitor
C2	7405 hex inverter TTL IC
IC1	7492 BCD counter TTL IC
IC2, IC3	74LS161 up counter TTL IC
IC4	

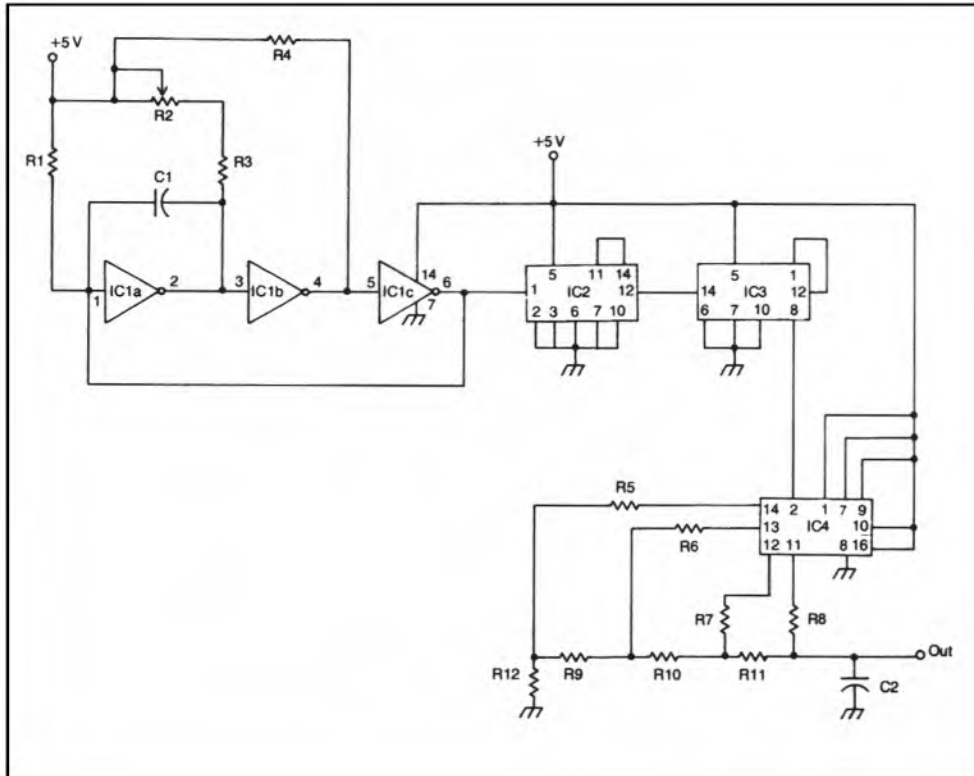


Fig. 3-32. Digital sawtooth wave oscillator.

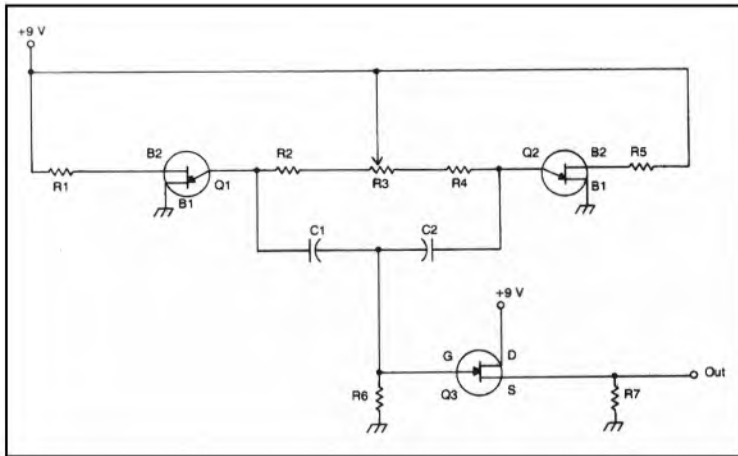


Fig. 3-33. Transistor sawtooth wave oscillator.

look like. For the best linearity, the values of C1 and C2 should be very closely matched. It may be a good idea to use precision capacitors here if you are concerned about generating a clean sawtooth wave signal.

The parts list for this circuit is given in Table 3-15.

Odd Waveforms

Most oscillators generate one of the basic waveshapes (sine, rectangle, triangle, or sawtooth). Other, more exotic waveshapes are usually created by combing signals (*additive synthesis*) or removing (filtering out) portions of a signal (*subtractive synthesis*).

In some cases, however, it may be more practical to build a circuit that can generate an unusual waveform directly.

Often this is done by deliberately distorting a basic waveshape. Refer back to the circuit in Fig. 3-33. You'll recall that C1 and C2 should be closely matched for good linearity. Obviously, this means

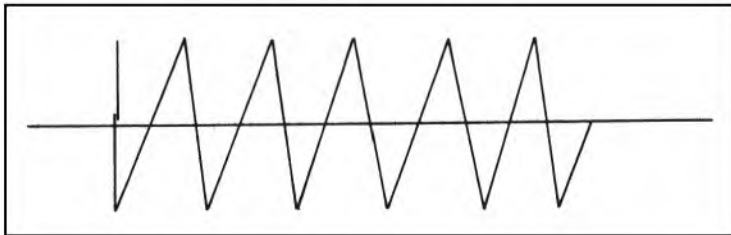


Fig. 3-34. Output signal of the circuit shown in Fig. 3-33.

Table 3-15. Parts List for the Transistor Sawtooth Wave Oscillator in Fig- 3-33.

R1, R5, R7	4.7 kΩ resistor
R2, R4	100 kΩ resistor
R3	1 MΩ potentiometer
R6	1 MΩ resistor
C1, C2	0.01 μF capacitor
Q1, Q2	UJT (TIS-43, or similar)
Q3	FET (MPF103, Radio Shack RS2028, or similar)

we can distort the output waveform by changing the value of one of the capacitors. Figure 3-35 shows a modification of the circuit that allows six different output waveforms. Figure 3-36 shows a couple of waveforms that might be generated by this oscillator. They are distortions of the basic sawtooth wave, but they may sound quite different. Each waveshape is switch selectable. A 6 position rotary switch should be used. The capacitors may be almost anything in the range of about 0.001 μ F up to about 0.1 μ F.

Staircase Wave. Another popular nonstandard waveform is the staircase wave. This is illustrated in Fig. 3-37. A simple circuit

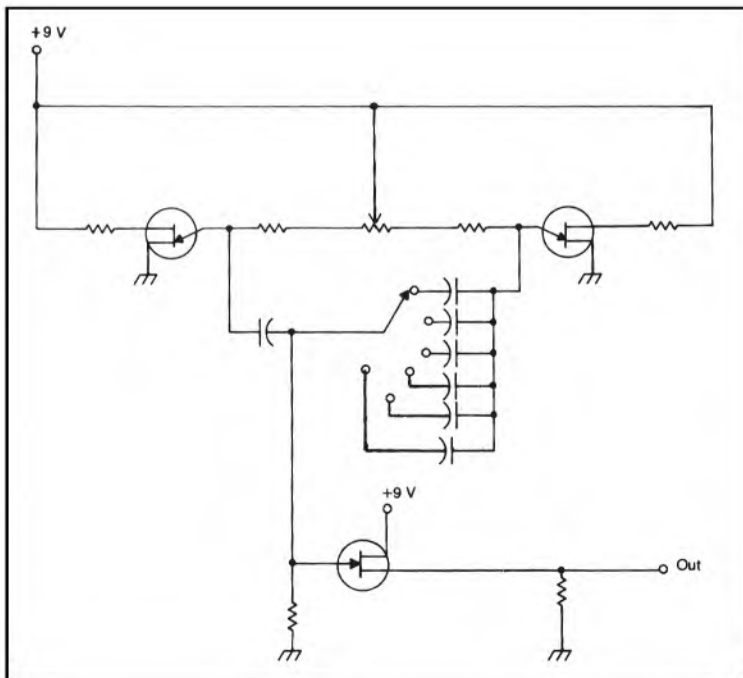


Fig. 3-35. Modification of the circuit shown in Fig. 3-33.

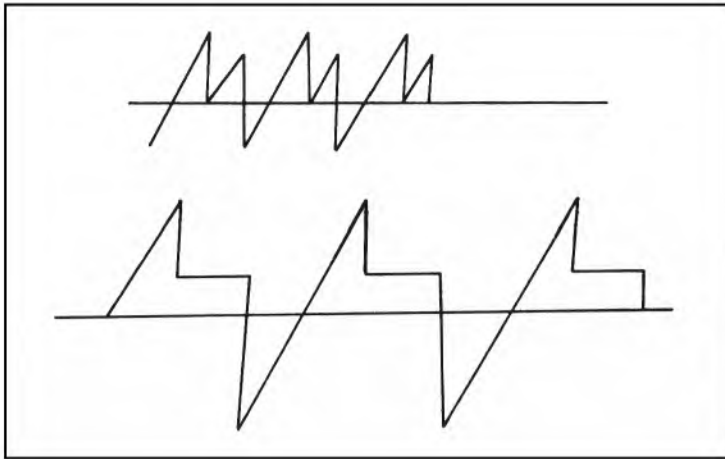


Fig. 3-36. Possible output signals of the modified circuit shown in Fig. 3-35.

for converting a square wave to a staircase wave is shown in Fig. 3-38. When S1 is closed, the circuit is reset and a new staircase cycle begins. Obviously a manual switch is not very practical here. Many different types of electronic switching may be used. The block diagram in Fig. 3-39 shows how a second square wave oscillator and a digital gate may be used. The second oscillator resets the staircase on each positive half-cycle. The second oscillator should have a much lower frequency than the first oscillator. The input oscillator must have a high frequency to keep the staircase output signal in the audible range. The input frequency may be above the audible range. For best results, the two oscillator frequencies should be exact multiples of each other. This insures that each cycle of the staircase wave will have the same number of steps.

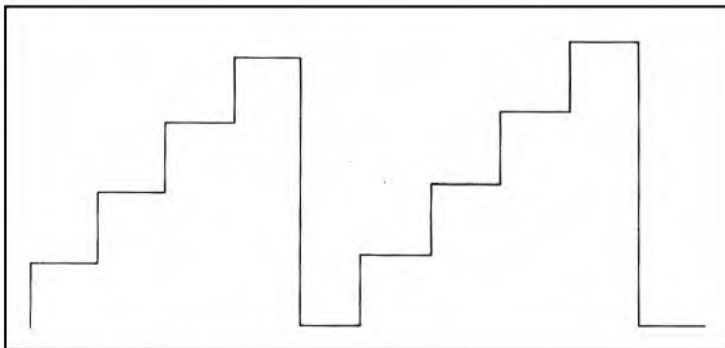


Fig. 3-37. Staircase wave.

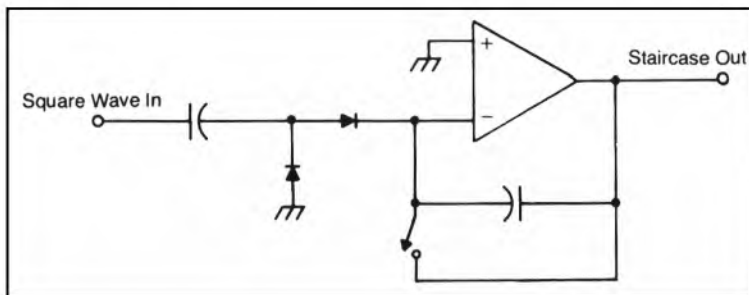


Fig. 3-38. Op amp staircase wave generator.

Another approach is illustrated in Fig. 3-40. After a specific number of input cycles, the counter triggers the gate and resets the staircase generator. This system removes the possibility of an irregular number of steps in the output signal.

The circuit that was shown back in Fig. 3-32 can be converted into a sixteen step staircase generator simply by removing capacitor C2. You might want to make this capacitor selectable with a switch to give you a choice of output waveforms—sawtooth or staircase.

Yet another staircase wave oscillator is illustrated in Fig. 3-41. This circuit's output signal is somewhat different from those produced by the two previous circuits. In this case, the staircase climbs up, then steps back down, as illustrated in Fig. 3-42.

Incidentally, adding a capacitor between the output and ground can give you a fair digital approximation of a triangle wave.

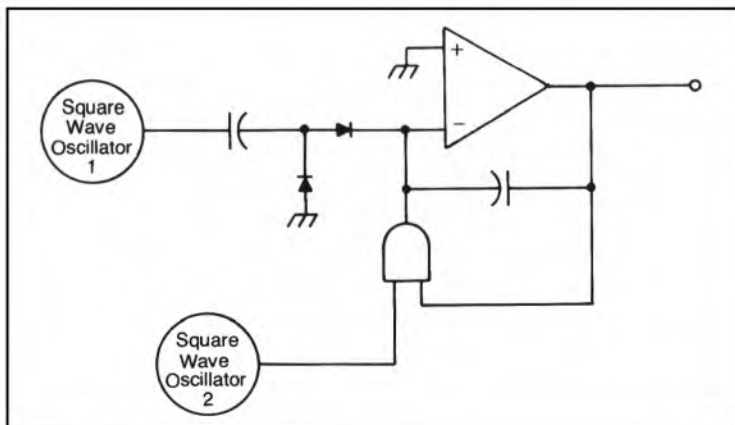


Fig. 3-39. Practical staircase wave generator.

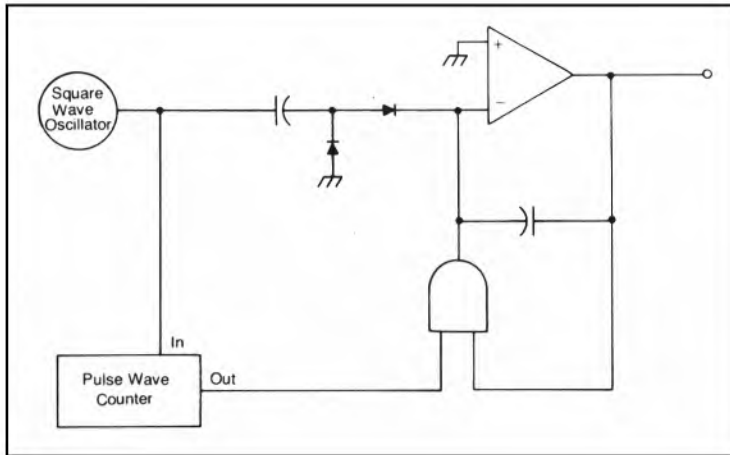


Fig. 3-40. Another practical staircase wave generator.

The parts list for the staircase wave oscillator circuit of Fig. 3-41 is given in Table 3-16.

Programmable Waveform Generator. Two 555 timer ICs (or a single 556 dual timer IC) can be wired up to generate a wide variety of oddball waveforms. The circuit is shown in Fig. 3-43, and the parts list is given in Table 3-17. By twiddling the two potentiometers (R1 and R3) and selecting various capacitors with switches S1 and S2, you can generate a wide range of unique sounds and effects. Some of these will be very musical and pleasant. Others will be out and out obnoxious. If nothing else, you should have a few hours of fun playing with this circuit.

Another circuit that can generate a wide variety of unusual signals is shown in Fig. 3-44. This gadget uses 10 CMOS switches (IC3, 4, and 5) and 10 potentiometers (R3 through R12) to allow you to program thousands of waveforms. The circuit will step through each of the potentiometers in sequence, at a rate determined by R1 (controlling the frequency of a 555 oscillator). To save some cash on

Table 3-16. Parts List for the Digital Staircase Wave Generator in Fig. 3-41.

R1	100 k Ω potentiometer
R2	10 k Ω resistor
R3, R5	22 k Ω resistor
R4, R7	33 k Ω resistor
R6, R8	27 k Ω resistor
IC1	555 timer IC
IC2, IC3	CD4042 quad latch CMOS IC

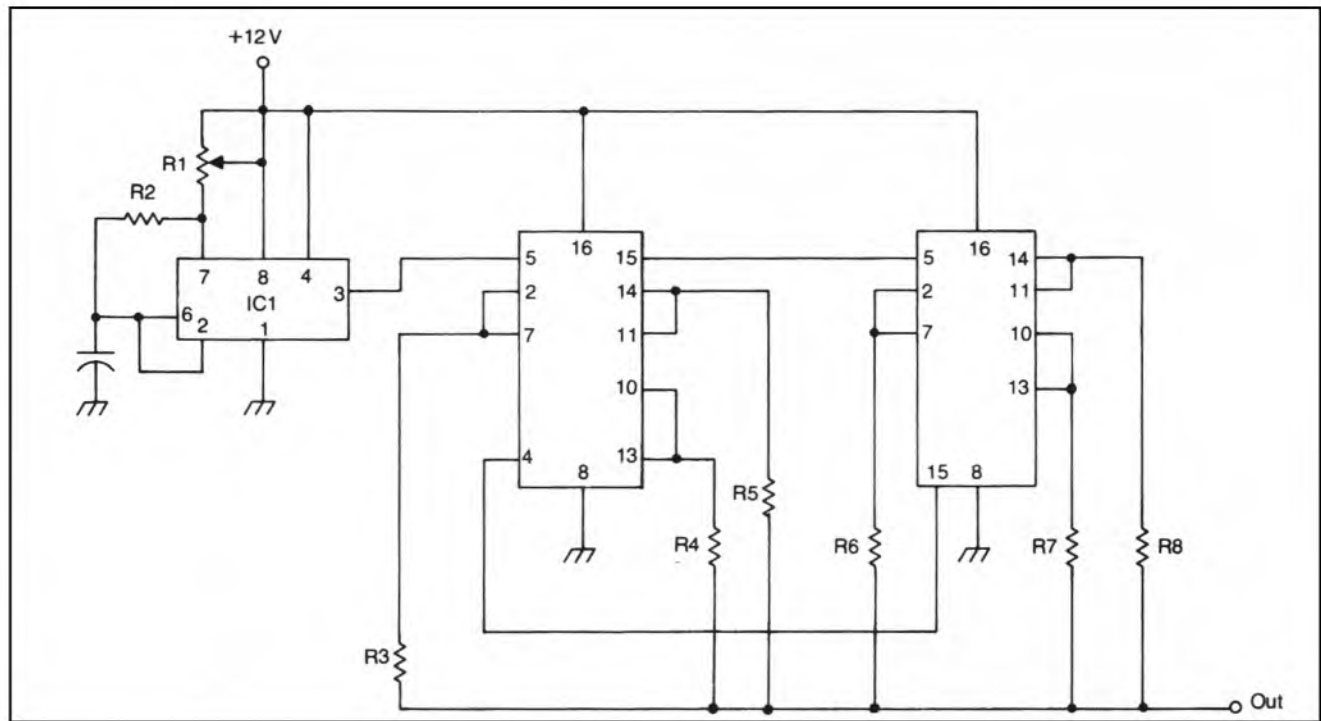


Fig. 3-41. Digital staircase wave generator.

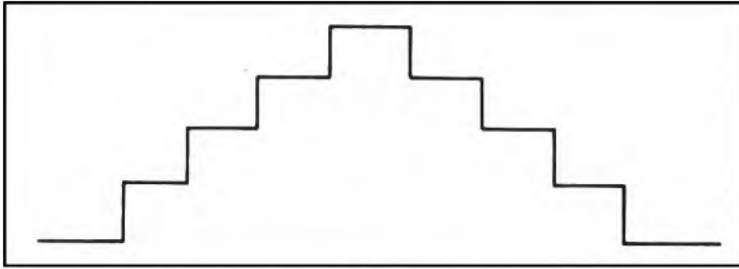


Fig. 3-42. Output signal of the circuit shown in Fig. 3-41.

this circuit, you might want to use miniature trim pots for R3 through R12, but this will make the circuit harder to use. Front panel mounted potentiometer's are the best bet.

By slowing the 555 timer's frequency down below the audible range (substitute a 1 μF to 5 μF electrolytic capacitor for C1), you can use this circuit as a ten step sequencer. Sequencers and their uses will be discussed in Chapter 8.

The parts list for the programmable waveform generator is given in Table 3-18.

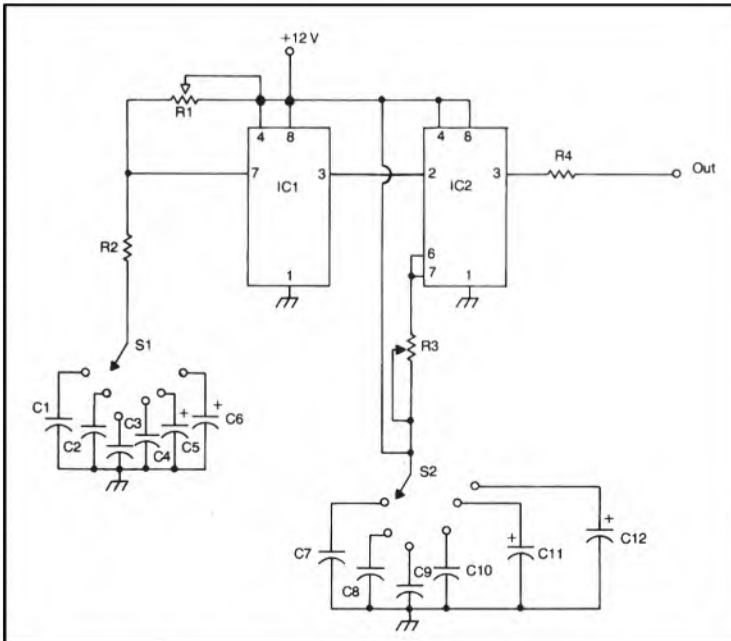


Fig. 3-43. 555 odd waveform generator.

Table 3-18. Parts List for the Programmable Waveform Generator in Fig. 3-44.

R1	500 k Ω potentiometer
R2	1 k Ω resistor
R3-R12	10 k Ω potentiometer
C1	0.1 μ F capacitor
IC1	555 timer IC
IC2	CD4017 decade counter CMOS IC
IC3, IC4, IC5	CD4066 quad switch CMOS IC

Multi-Waveform Generators

Often it is handy to have a single circuit that can generate two or more of the basic signal waveforms. Such a device is called a **multi-waveform generator**.

Such a circuit can reduce the size and cost of your total synthesizer. By combining functions you achieve greater efficiency.

In some cases the various waveshapes are simultaneously available, which allows use of differing signals with exactly the same frequency and phase (cycles beginning and ending at the same instant). This can be extremely useful in some complex patches.

Figure 3-45 shows the circuit for a relatively simple multi-waveform generator. This circuit can produce triangle, ascending sawtooth, or square waves.

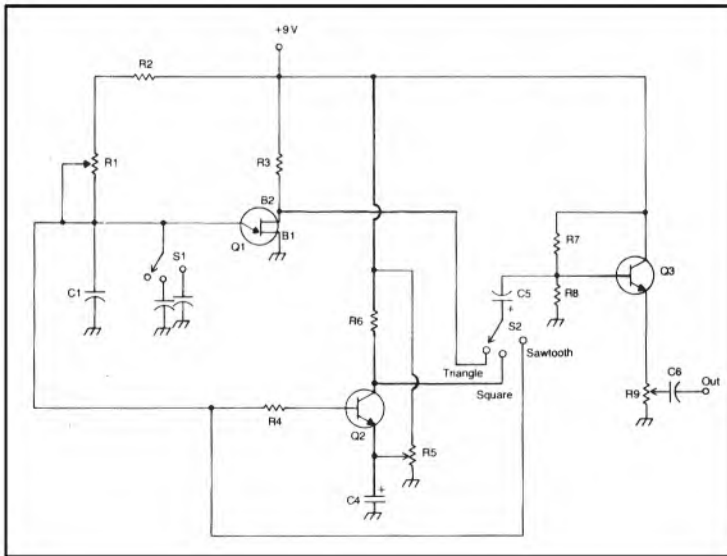


Fig. 3-45. Multi-waveform generator.

Table 3-19. Parts List for the Multi-Waveform Generator in Fig. 3-45.

R1	25 k Ω potentiometer
R2	2.2 k Ω resistor
R3	1.5 k Ω resistor
R4, R6	10 k Ω resistor
R5	50 k Ω potentiometer
R7, R8	100 k Ω resistor
R9	5 k Ω potentiometer
C1	0.01 μ F capacitor
C2	0.1 μ F capacitor
C3	1 μ F capacitor
C4, C5	47 μ F electrolytic capacitor
C6	0.47 μ F capacitor
Q1	UJT (2N2646, or Radio Shack RS2029, or similar)
Q2, Q3	PNP transistor (2N4124, GE-20, RCA SK3020, or similar)
S1, S2	3 pole, single throw switch

One disadvantage with this circuit is that the outputs are not simultaneously available. Only one waveform may be used at a time. The output signal's waveshape is selected with the three position switch (S2).

Switch S1 is a range switch to allow this circuit to cover a wide range of output frequencies. Depending on the accuracy of your component values, this device should have a range of about 200 Hz to 20 kHz (20,000 Hz). The potentiometer (R2) is used to fine tune the output frequency within a range.

The parts list for the multi-waveform generator is given in Table 3-19.

THE VOLTAGE-CONTROLLED OSCILLATOR

So far, all of the oscillator circuits we have looked at either operate at a fixed output frequency, or are manually variable—that is, the output frequency may be altered within a specific range by turning the shaft of a potentiometer, or a variable capacitor. While such circuits certainly have their uses, composing electronic music with only this type of device is extremely tedious at best.

A number of composers have experimented with electronic music synthesis for many years. These people were dedicated technicians and artists, often involved in the avant-garde. They were willing to spend weeks, or even months putting together a taped piece that might run for a minute or two.

Voltage Control

Electronic music did not become popular until the sixties. Bob

Moog introduced his first modular synthesizer that was built around the then revolutionary concept of voltage control. Electrical voltages can set the signal parameters more accurately and far faster than any human operator can twiddle knobs. Much of the work involved in sound synthesis can be automated by voltage control.

Until the advent of computer controlled digital synthesizers in the past few years, virtually all practical synthesizers used voltage control to a greater or lesser extent. Voltage control brought electronic music out from the world of weird beeps and boops and into the world of real music.

Voltage control techniques have also paved the way for live performance and real-time synthesis without the previously necessary intervention of a tape recorder.

In this book, we will frequently return to the concept of voltage control. Regretably, digital synthesis techniques are beyond the scope of this work. This subject warrants a volume of its own.

Dc Control Voltage. The voltage-controlled oscillator (vco) is an important part of any synthesizer. The output frequency is determined by a voltage fed into the circuit's input. There are many possible sources for this control voltage. For the time being we will briefly mention the most important of these.

Most synthesizers have some form of keyboard to allow the musician to easily relate to the instrument and play it. This keyboard generally resembles that of a piano or organ. Whenever a key is depressed, a specific dc voltage is generated. This voltage can be used to drive any voltage-controlled module, and is most frequently fed into voltage-controlled oscillators. The output voltage of each key is carefully adjusted so that the voltage-controlled oscillator (or vco) generates a definite frequency corresponding to the position on the keyboard. Usually the standard Western twelve-tone scale will be used, but this is not specifically neces-

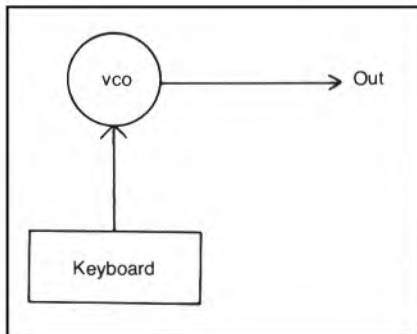


Fig. 3-46. Controlling a vco with a voltage keyboard.

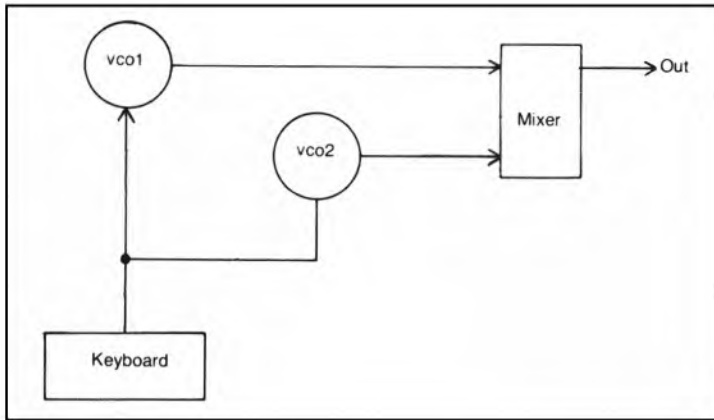


Fig. 3-47. Controlling two vcOs in unison.

sary. This simple, basic hook-up is shown in Fig. 3-46. This drawing is an example of what is known as a *patch diagram*.

Patch diagrams are a handy way to notate the way a synthesizer's modules are interconnected to produce a specific effect. They will be used extensively throughout this book.

A single control voltage may be split to feed two or more separate modules. This is illustrated in Fig. 3-47. The two oscillators will sound in unison. They may have different waveshapes, and their signals may be processed in different ways further down the line, but they will be operating at the same frequency (assuming, of course, that the same type of circuitry is used in both oscillators). Because the odds are that the two (or more) unison oscillators will be out of phase with each other, this type of hookup is often used for chorus effects. The effect of mixing out of phase signals is illustrated in Fig. 3-48. Portions of the signals will reinforce each other, while other portions will cancel each other out. This causes the sound to fluctuate in a very natural sounding way, and gives the sensation of several instruments being played together. For instance, a solo violin sounds quite different from a dozen violins being played in unison.

The chorus effect can be strengthened if one of the oscillators is detuned slightly so that they are not perfectly in unison. This can be easily accomplished by adding a second control voltage to the input of one of the vcOs. This is a fixed dc voltage, which is referred to as a bias. The hookup is shown in Fig. 3-49. This is one of the uses for the variable voltage source presented in Chapter 2.

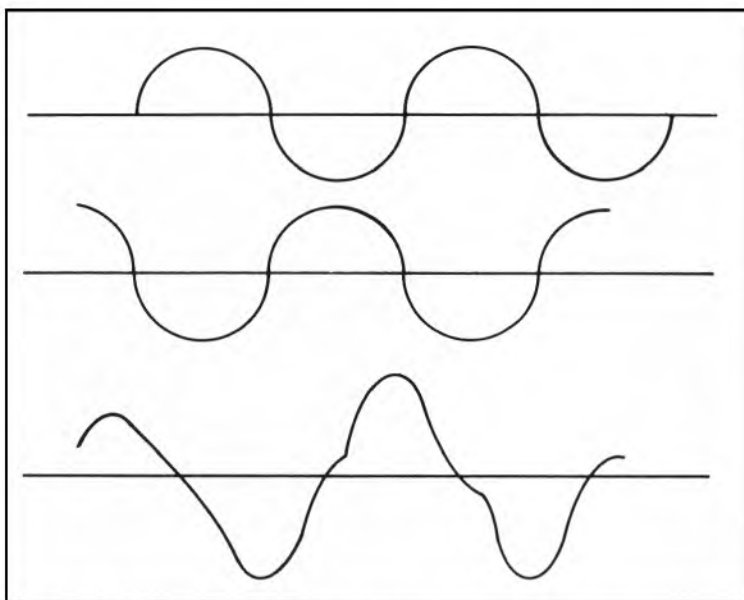


Fig. 3-48. Combining out of phase signals.

By increasing the bias voltage, the two vcos may be played in fixed harmony. For example, vco2 may be adjusted so that it is a third above vco1. This relationship will hold up and down the scale. For example, when vco1 sounds a C, vco2 will sound an E. When vco1 plays a F, vco2 will be playing an A.

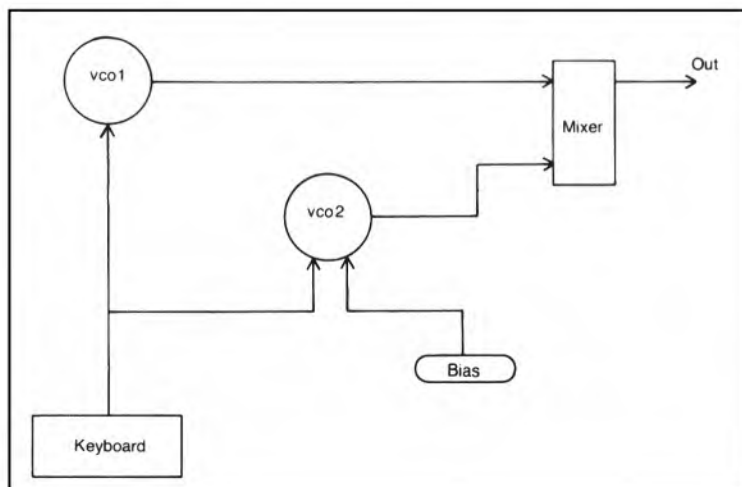


Fig. 3-49. Adding a bias voltage to the patch shown in Fig. 3-47.

More about keyboards and basic controllers (voltage control sources) will be presented in Chapter 7.

Ac Control Voltage. A vco may also be controlled by an ac signal, such as from another oscillator.

If the controlling oscillator has a very low frequency (below about 2 Hz or 3 Hz), it is possible to actually hear the controlling waveform. For instance, if the controlling oscillator generates a square wave, the output of the vco will alternate between a low and a high pitch. The distance between the two output frequencies is dependent on the amplitude (strength) of the controlling signal. To take a second example, let's assume the controlling signal is an ascending sawtooth wave. The output frequency will start out at a low pitch then build slowly and smoothly up to a maximum high frequency, then quickly drop back down to the original low frequency and start over again. Refer back to Fig. 3-3 and Fig. 3-4 to see how these effects sonically resemble the controlling waveshape.

If the controlling waveform is increased to a frequency of about 5 Hz to 10 Hz, a *vibrato* effect will be obtained. *Vibrato* is a rapid fluctuation of pitch and it adds warmth and life to a sound. A pure tone often sounds dull, flat, and lifeless. Adding a little bit of vibrato can make electronic music seem less mechanical.

However, vibrato effects can become very tiresome, especially if a deep vibrato (wide pitch fluctuations) is used. For most pleasing results, the controlling oscillator should put out a very low amplitude signal (or the signal may be attenuated) so that the vco's output frequency will only vary very slightly.

When the controlling waveform is increased in frequency to the audible range (above about 20 Hz) some very striking effects start to appear. This is called frequency modulation (or FM), and it will be discussed in Chapter 4.

Circuits

A number of voltage-controlled oscillator circuits are explained in the following pages.

Two Transistor vco. Figure 3-50 shows the schematic for a simple two transistor vco circuit. The parts list is given in Table 3-20. Notice that this circuit requires an +18 volt power supply, so the main power supply circuit that was presented in Chapter 2 is not appropriate here.

This circuit offers two simultaneous outputs. Output 2 is a standard ascending sawtooth wave. Output 1 generates a less com-

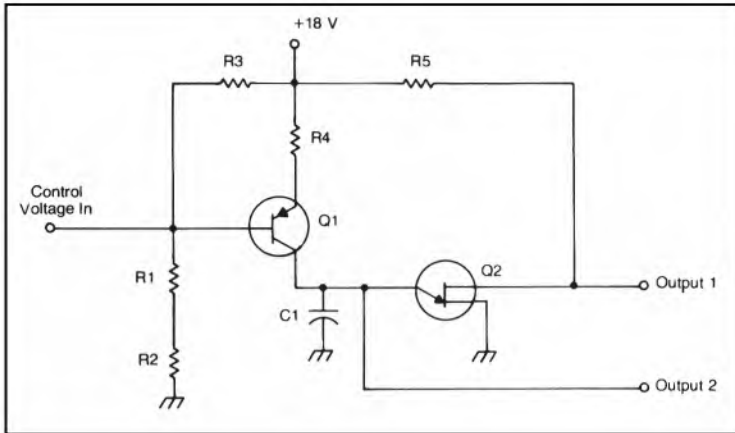


Fig. 3-50. Two transistor vco.

mon waveshape, that may be called a negative spike wave. This waveshape is illustrated in Fig. 3-51.

Both of the waveforms generated by this vco are very rich in harmonics.

Five Transistor vco. A somewhat more complex vco circuit is shown in Fig. 3-52. The parts list is given in Table 3-21. This circuit generates a relatively pure sine wave, and it is ideal for use in additive synthesis systems (see Chapter 4).

Vcos that produce sine waves are fairly uncommon, so this circuit can come in very handy when the need arises.

Op Amp vco. A third voltage-controlled oscillator circuit is illustrated in Fig. 3-53. This one is built around an op amp IC and a SCR. It is probably the most suitable of these three circuits for use in an electronic music synthesis system. Since the circuit is fairly simple, relatively inexpensive to build, and quite small, it is practical to include a number of these vcos in your synthesizer. While you

Table 3-20. Parts List for the Two Transistor Voltage-Controlled Oscillator in Fig. 3-50.

R1	6.8	k Ω	resistor
R2	680	Ω	resistor
R3, R4	2.2	k Ω	resistor
R5	330	Ω	resistor
C1	0.022	μ F	capacitor
Q1			PNP transistor (2N2393, Motorola HEP-52, or Radio Shack RS 2021)
Q2			UJT transistor (2N441, 2N2077, GE-4, or Radio Shack RS 2006)

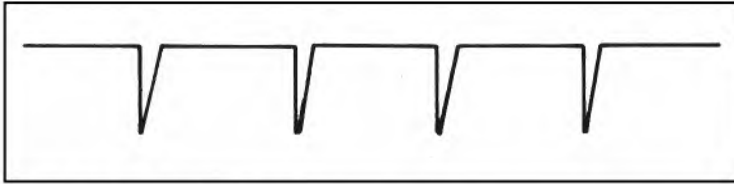


Fig. 3-51. Negative spike wave.

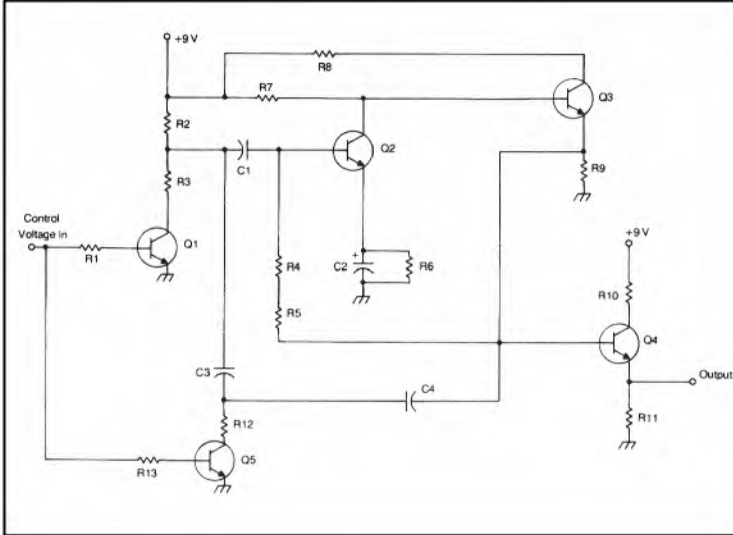


Fig. 3-52. Sine wave vco.

Table 3-21. Parts List for the Sine Wave Voltage-Controlled Oscillator in Fig. 3-52.

R1, R13	470 Ω resistor
R2, R12	560 Ω resistor
R3, R4, R9	8.2 k Ω resistor
R5	1 k Ω resistor
R6	420 Ω resistor
R7	2.2 k Ω resistor
R8, R11	4.7 k Ω resistor
R10	100 Ω resistor
C1	0.047 μ F capacitor
C2	1 μ F capacitor
C3	0.01 μ F capacitor
C4	0.022 μ F capacitor
Q1-Q5	NPN transistor (2N4264, GE-17, Motorola HEP-50, or Radio Shack RS-2016)

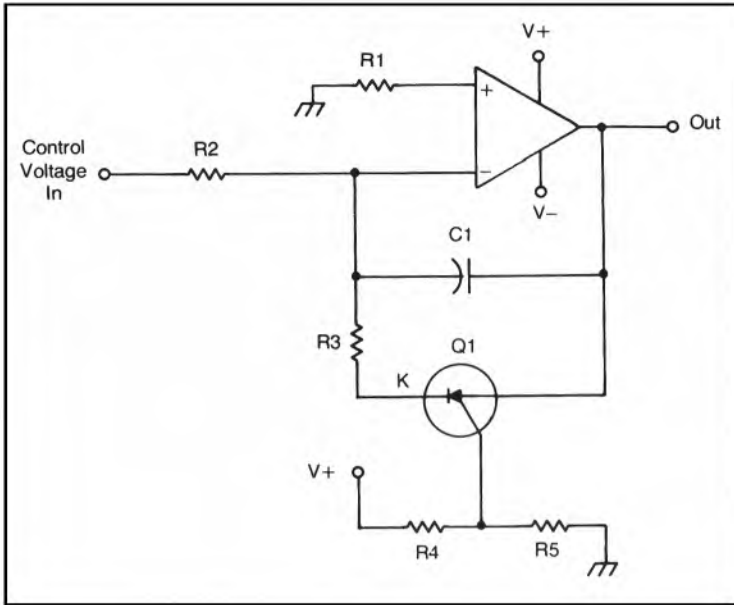


Fig. 3-53. Op amp vco.

can do some limited synthesis with a single vco, advanced electronic musicians often find they never have enough vcOs.

A reasonable starting point would be to build three to five vcOs. The circuit of Fig. 3-53 can be built small enough to fit several complete vcOs in a single module. Only the input and output jacks need to appear on the front panel in the basic version.

You may, however, want to include several switch selectable capacitors (for C1) to allow the circuit to cover a wide range of output frequencies. Keep these capacitors between about 0.01 μF and 1 μF in value.

Another possible modification would be to replace resistor R3 with a 2.2 k Ω , fixed resistor and a 10 k Ω , potentiometer as a fine turning control. This will allow you to play several of these oscillators in harmony without having to resort to external bias voltages.

Table 3-22. Parts List for the Op Amp Voltage-Controlled Oscillator in Fig. 3-53.

R1-R3, R5	10 k Ω resistor
R4	3.3 k Ω resistor
C1	0.15 μF capacitor
Q1	SCR (2N6027, or equivalent)
IC1	op amp IC (741, or equivalent)

For more precision fine tuning you could try a 8.2 k Ω . fixed resistor and a 2.5 k Ω potentiometer. A multiple turn potentiometer would allow extremely fine control.

Both of these modifications are shown in Fig. 3-54.

9400 vco IC Sine voltage-controlled oscillators have numerous applications in many areas of electronics, it is not surprising that several dedicated vco ICs have been developed in recent years. The hobbyist can take advantage of these devices to build or expand an electronic music synthesizer.

A particularly handy chip is the 9400 Voltage-to-Frequency/Frequency-to-Voltage converter. This device can be used in either

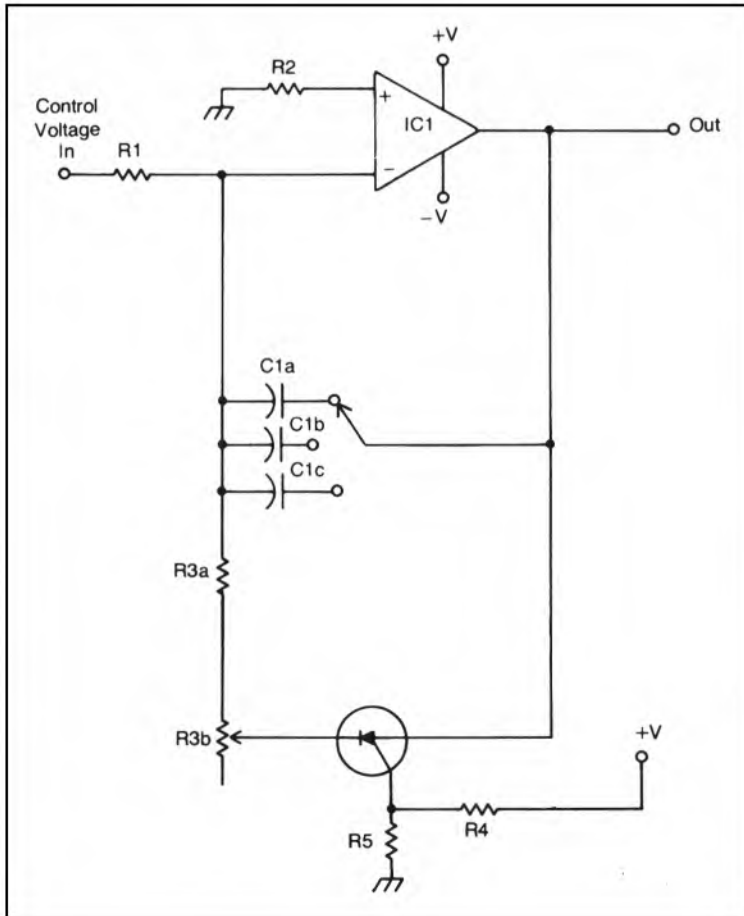


Fig. 3-54. Modifying the op amp vco of Fig. 3-53.

Table 3-23. Parts List for the Voltage-to-Frequency Converter in Fig. 3-55.

R1	1 MΩ resistor
R2, R4	33 kΩ resistor
R3, R5	100 kΩ resistor
C1	0.001 μF capacitor
C2	47 pF capacitor
IC1	9400 V/F - F/V converter IC

of two modes. In the frequency-to-voltage mode, an input frequency is converted into a proportional dc frequency at the output. This mode will be examined in more detail later in this chapter.

In the voltage-to-frequency mode, this circuit behaves like a vco. The output frequency is proportional to the input voltage. A 9400 based voltage-controlled oscillator circuit is illustrated in Fig. 3-55. The parts list is given in Table 3-23. The output is a square wave.

566 vco IC Probably the most commonly available vco IC is the 566, which is made by several manufacturers. It is readily available from a number of convenient sources, including the widespread Radio Shack chain of stores. Some manufacturers and dealers refer to the 566 as a function generator, rather than a vco, but there is no difference, aside from the name.

The 566 generates two simultaneously available output waveforms. They are square waves and triangle waves. Both sig-

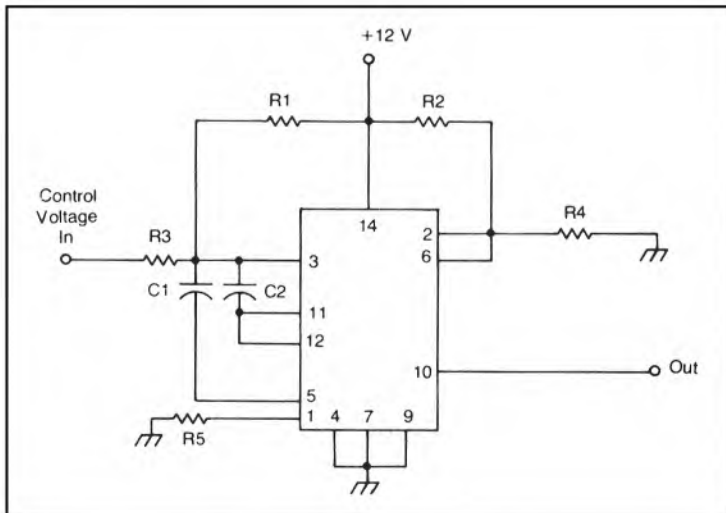


Fig. 3-55. 9400 voltage-to-frequency converter.

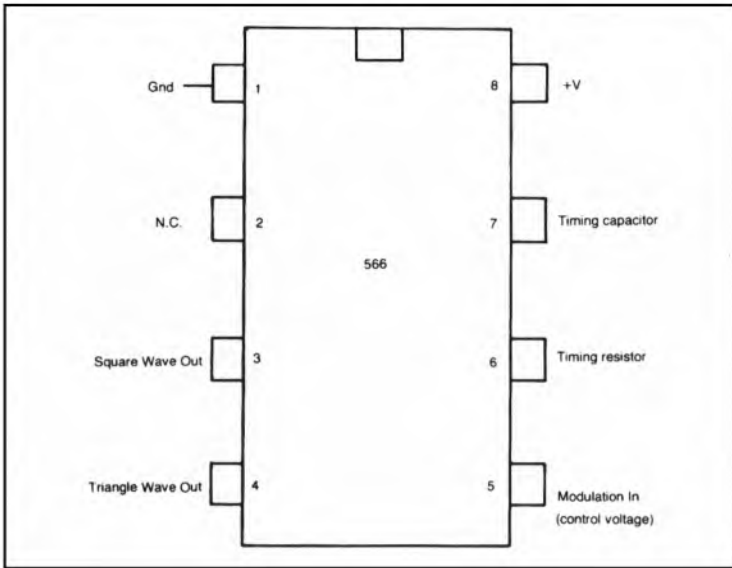


Fig. 3-56. Pin-out diagram of the 566 vco IC

nals are fully buffered, which makes interfacing to other circuits a simple matter.

The pin-out diagram for the 566 vco IC is shown in Fig. 3-56. This diagram shows the somewhat more popular 8-pin DIP package. This chip is also available in an 8-pin round package. The pin

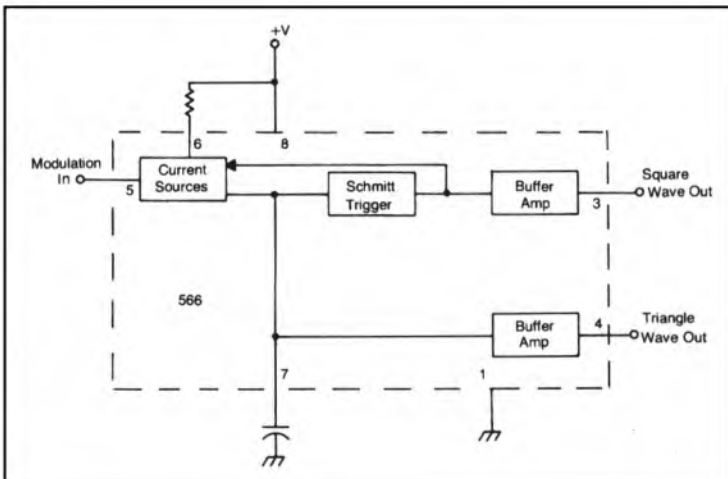


Fig. 3-57. Functional block diagram of the 566 vco IC

numbers are completely compatible for both packages, so they may be easily interchanged.

A functional block diagram of the basic 566 vco circuit is shown in Fig. 3-57.

This device is capable of generating frequencies up to 1 MHz (1,000,000 Hz). This is obviously well beyond the range required for electronic music.

Three main factors determine the output frequency of the 566 vco. First, there is the control voltage, which is applied to the modulation input (pin 5). Second, the timing resistor which is connected between pin 6 and the positive power supply. Third, the timing capacitor which is connected between pin 7 and ground. The oscillator can cover a 10 to 1 frequency range with a single fixed capacitor. For a wider frequency range, it may be necessary to include a range switch to select between several capacitors.

The formula for determining the output frequency of the 566 is as follows:

$$F = \frac{2(V - V_c)}{RCV}$$

where F is the output frequency in hertz, V is the positive supply voltage, V_c is the control voltage applied to pin 5 and R and C represent the timing resistor and capacitor respectively. For stable operation, the value of R should be kept between 2 k Ω and 20 k Ω .

This chip is very flexible about power requirements. Single or double ended power supplies (+ only or \pm) from 10 to 24 volts are acceptable. It can be driven by the 12 V output of the power supply described in Chapter 2.

Despite all of this, the 566 vco IC is not quite ideal for use in an electronic music synthesis system. The control voltage operates on the output frequency in a linear, rather than an exponential manner. The human ear hears exponentially.

To explain the difference, let's assume a 1 volt control voltage generates a 500 Hz output signal. In a linear system, increasing the control voltage to 2 volts will add 500 Hz to the output frequency for a 1000 Hz (1 kHz) output. Raising the control voltage another volt (a total of 3 volts) adds another 500 Hz bringing the output frequency to 1500 Hz (1.5 kHz). Each volt (or fraction thereof) represents a fixed frequency that is simply added to the total.

In musical applications, however, we aren't interested in in-

creasing frequencies by fixed amounts—we are interested in ratios. For example, an octave represents a doubling of frequency. Again, let's assume a 1 volt control voltage signal generates a 500 Hz output. Increasing the control voltage to 2 volts will multiply the original output by 2 to give us 1 kHz, as with the linear system. But raising the control voltage to 3 volts in an exponential system should produce an output that is 2 octaves higher than the original 1 volt frequency—that is, 2000 Hz (2 kHz).

It is very difficult to use linear voltage control musically, although some commercial synthesizers do manage.

The problem with the 566 vco IC is that, while it is a dedicated voltage-controlled oscillator, it was not specifically designed for electronic music applications.

A few manufacturers do market dedicated ICs that are specifically designed for electronic music applications.

CEM 3340 vco IC Curtis Electro Music Specialties manufactures a vco IC called the CEM 3340. This device can cover a 50,000:1 frequency range under exponential control. The tracking is extremely accurate, so the vco can easily be kept in tune. It can also be phase synced with another oscillator for special effects.

The CEM 3340 vco can generate square, pulse, sawtooth, and triangle waves. The signals may be used simultaneously.

While electronic music systems are usually built around exponential control, as discussed above, this chip can use both exponential and linear voltage control at the same time (separate inputs are provided).

A basic vco circuit built around the CEM 3340 is shown in Fig. 3-58. Notice that a dual polarity power supply is called for. The circuit is intended to operate on ± 15 volts, but the specifications have more than enough leeway to make the ± 12 volt outputs of the power supply from Chapter 2 perfectly adequate.

SSM2030 vco IC Another company, Solid State Music, also manufactures several dedicated vco ICs for electronic music applications. Figure 3-59 shows the pin-out diagram of their SSM 2030 vco.

This device can generate simultaneously available pulse, sawtooth, and triangle wave outputs over a 1,000,000:1 range. The duty cycle of the pulse (or rectangle) wave may also be voltage controlled. The SSM 2030 vco may be synchronized with an external oscillator for special effects.

A dual power supply from ± 9 volts to ± 18 volts is needed to

operate this device, with ± 15 volts being the standard called for in the specification sheet.

SPECIAL OSCILLATOR CIRCUITS

Noise generators and sound effects circuits produce complex waveforms that do not fall into a basic category. They are dedicated circuits that produce specific effects.

Noise Generators

Another commonly used signal source in the electronic music synthesizer is a *noise generator*. You might think noise in a sound

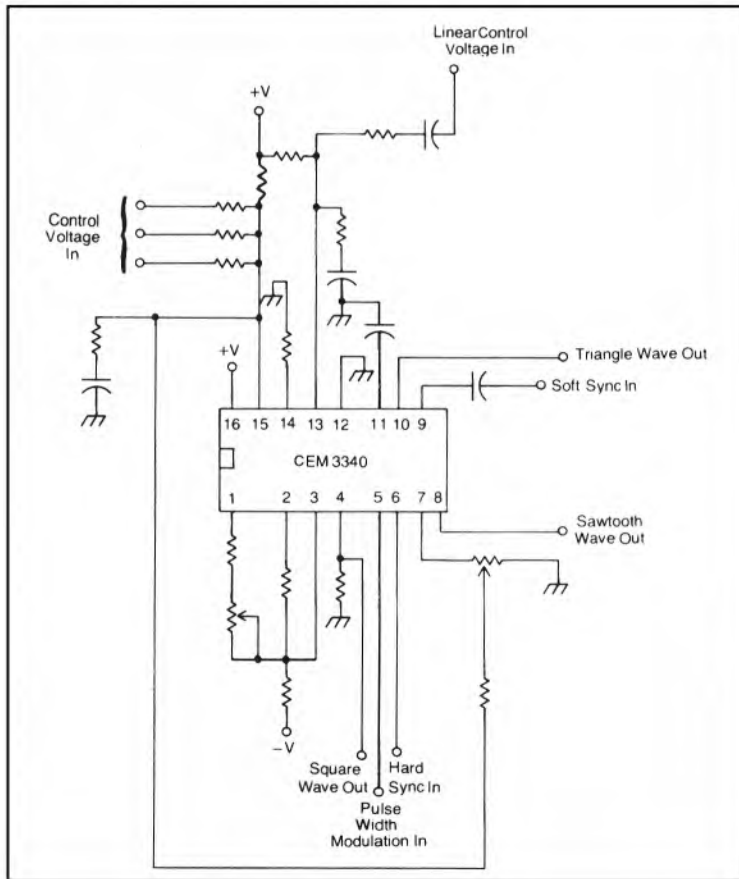


Fig. 3-58. Basic CEM 3340 vco circuit.

system would be something to be avoided at all costs, and certainly not deliberately generated. That is true, but we are using the word "noise" here in a somewhat different way than you're probably used to.

Ordinarily, noise is considered to be any unwanted sound or signal. In this case, however, we are using noise to mean a signal that is comprised of a large number of random frequencies. As you will soon see, such a haphazard signal is really quite useful in sound synthesis.

Noise Color. Noise, in this context, comes in two primary types, or colors. They are white noise and pink noise.

White noise is analogous to white light. It is a composite of all frequencies in nominally equal amounts. Any specific frequency is no more or less likely to be present at any randomly selected instant than any other frequency. If you tune a FM receiver between channels, the hissing sound you will hear is essentially white noise. In technical terms, white noise contains equal energy per frequency.

Pink noise, on the other hand, contains equal energy per octave. That is, at any randomly selected instant there is equal probability of a frequency within a specific octave. Since octaves are a doubling of frequency, higher octaves contain more discrete frequencies. Starting at 100 Hz, the first octave consists of all frequencies between 100 and 200 Hz (a 100 Hz range). The second octave covers 200 Hz to 400 Hz (a 200 Hz range). The third octave contains all frequencies from 400 Hz to 800 Hz (a 400 Hz range). In white noise, there is emphasis on the higher octaves. For this reason the FM between station hiss sounds like it has a high pitch.

In pink noise, the low octaves are boosted (and/or the high octaves are attenuated) to even up the probability of each octave at any randomly determined instant. Pink noise may be obtained by passing white noise through a low-pass filter (see Chapter 5).

Both white noise and pink noise can be very helpful in sound synthesis. These signals can be used to create percussive effects, indefinitely pitched sounds, and noise that might be heard from traditional instruments (i.e., the player's breath while playing a wind instrument, or a bow scraping across the strings). Noise is also useful in synthesizing sound effects like gunshots, explosions, drums, thunder, crashing surf, or gentle rainfall. Moreover, if tightly filtered, a noise generator can be used as a random voltage source.

Circuits. Noise generators are generally very simple circuits,

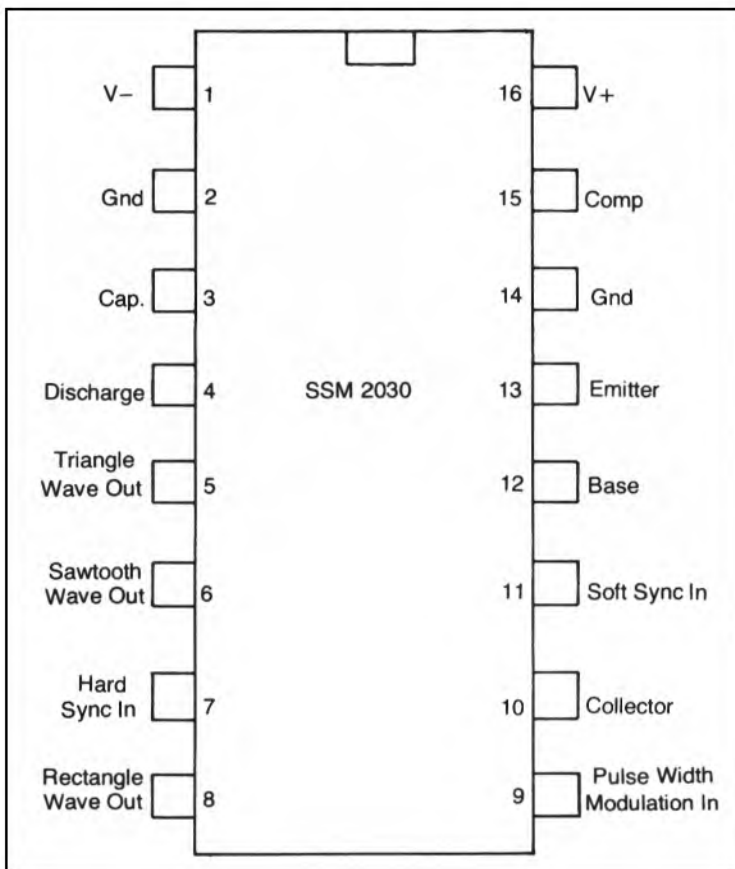


Fig. 3-59. Pin-out diagram for the SSM 2030 vco IC.

so there really isn't much reason not to include one in any sound synthesis system. It is extremely rare to ever need more than one, however.

Most traditional noise generators are built around a natural defect of semiconductors. Most diodes and transistors generate a certain amount of noise whenever current flows through them. Occasionally, you might build a noise generator that won't produce a steady stream of noise. If this happens, simply try another diode or transistor of the same type. Occasionally you'll come across one that isn't particularly noisy.

A typical noise generator circuit built around a diode is shown in Fig. 3-60. The parts list is given in Table 3-24. A fixed resistor may be substituted in place of the potentiometer (R2). This resis-

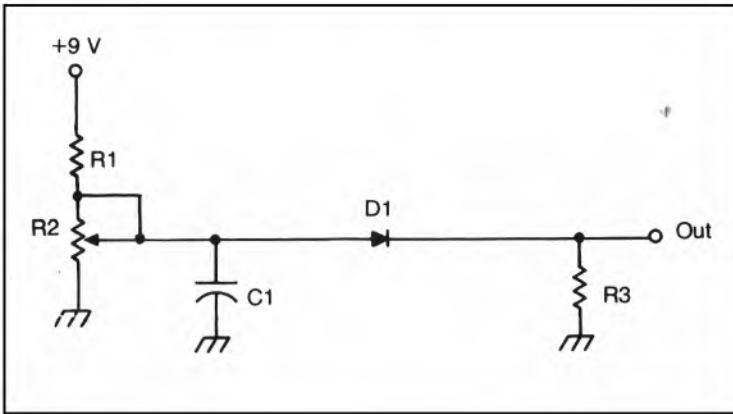


Fig. 3-60. Diode noise generator.

tance affects the average amplitude of the noise, and, to some extent, the color.

This circuit generates white noise (more or less). For the deeper sounding pink noise, you can use any of the low-pass filter circuits from Chapter 5 as shown in the patch diagram of Fig. 3-61.

A fairly new IC is available for easy noise generation. It is numbered S2688 or MM5837 (depending on the manufacturer) and is available from a number of suppliers, including Radio Shack. The pin-out diagram for this device is illustrated in Fig. 3-62.

The MM5837 is a digital device that actually generates pseudo-random frequencies in a repeating pattern. This pattern is internally chaotic and takes so long to repeat that the ear can not distinguish the result from truly random noise.

This chip is extremely easy to use. It is in an 8-pin DIP package, but, as Fig. 3-62 indicates, only four of the pins are used. They are the output, the ground, and two negative supply voltages. V_{DD} should be about -14 volts, while V_{SS} should be between -14 and -28 volts. Note that this device can not be powered by our basic power supply (refer back to Chapter 2).

Table 3-24. Parts List for the Diode Noise Generator in Fig. 3-60.

R1	470 Ω resistor
R2	100 k Ω potentiometer
R3	47 Ω resistor
C1	500 pF capacitor
D1	almost any diode (Radio Shack 276-1101, or similar)

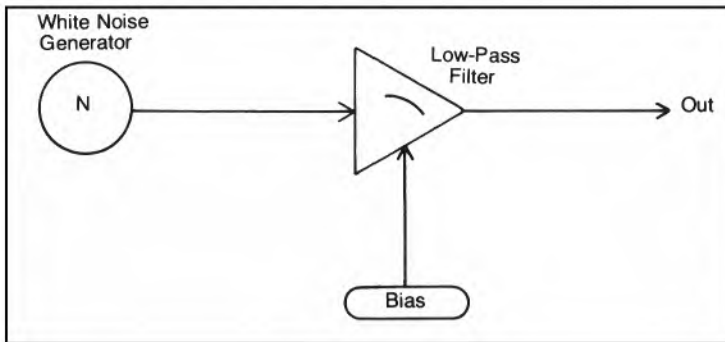


Fig. 3-61. Creating pink noise from white noise.

Sound Effects Circuits

Besides music, per se, many sound effects can be synthesized. They may be created by patching together discrete modules as in ordinary synthesis. But some sound effects are more easily obtained by building special, dedicated circuits.

While these circuits are designed to create specific sound effects, there is no law that says you must use them only as intended. Experiment with different resistor and capacitor values (an experimenter's breadboard, as mentioned in Chapter 1, comes in particularly handy here). Sometimes altering just a couple of component values can create striking new effects.

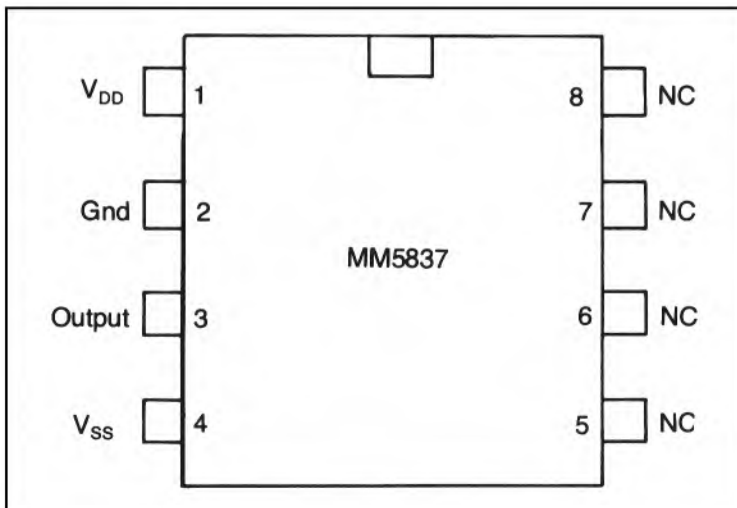


Fig. 3-62. Pin-out diagram for the MM 5837 digital noise generator IC

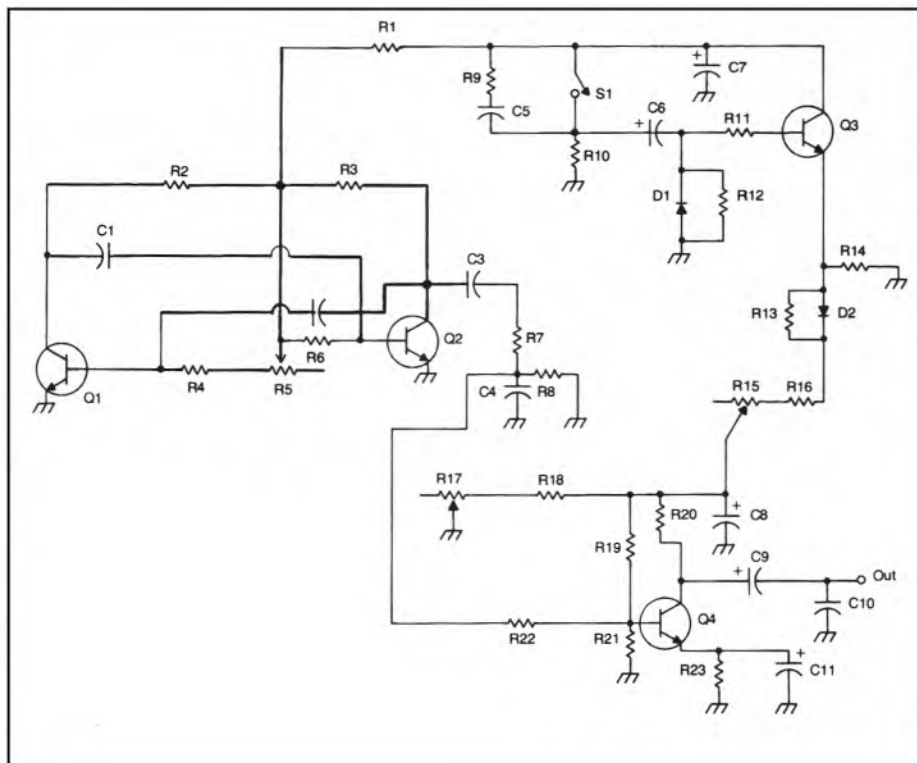


Fig. 3-63. Hank effect circuit.

The output signals from these circuits can also be freely modified by other synthesis modules, just like a regular oscillator's signal.

Honk. Figure 3-63 shows the schematic for a circuit that can generate a honk like a car horn, along with a variety of other novel sounds. Three potentiometers allow you to select a number of different sounds without changing any of the component values. The parts list for this circuit is given in Table 3-25.

The switch (S1) should be a NO (normally open) pushbutton type. Everytime the switch is temporarily closed, a honk will be generated.

The pitch of the effect can be changed by adjusting potentiometer R5. The other two potentiometers (R15 and R17) control the shape or the *envelope* of the output signal. The *attack* (set by R15) is the time required for the signal to build up from its minimum level (0) to its maximum level. Conversely, the *decay* (set by R17) is the time it takes for the signal to die back down to zero. Altering

Table 3-25. Parts List for the Honk Effect Circuit in Fig. 3-63.

R1	820 Ω resistor
R2, R3	15 k Ω resistor
R4	100 k Ω resistor
RS	100 k Ω potentiometer
R6, R19, R22	120 k Ω resistor
R7	39 k Ω resistor
R8	56 k Ω resistor
R9	560 Ω resistor
R10, R20	10 k Ω resistor
R11	47 k Ω resistor
R12	470 k Ω resistor
R13	1 M Ω resistor
R14, R21	12 k Ω resistor
R15, R17	2.5 k Ω potentiometer
R16	1 k Ω resistor
R18	100 Ω resistor
R23	2.2 k Ω resistor
C1, C2, C3	0.033 μ F capacitor
C4	0.001 μ F capacitor
C5	0.1 μ F capacitor
C6, C9	2.2 μ F 35 V electrolytic capacitor
C7	500 μ F 35 V electrolytic capacitor
C8, C11	30 μ F 35 V electrolytic capacitor
C10	5000 pF capacitor
S1	SPST NO push button switch
Q1, Q2	NPN transistor (2N2318, 2N2926, Motorola HEP-726, RCA SK3018, or Radio Shack RS 2031)
Q3, Q4	NPN transistor (GE-20, Motorola HEP-50, or Radio Shack RS 2016)

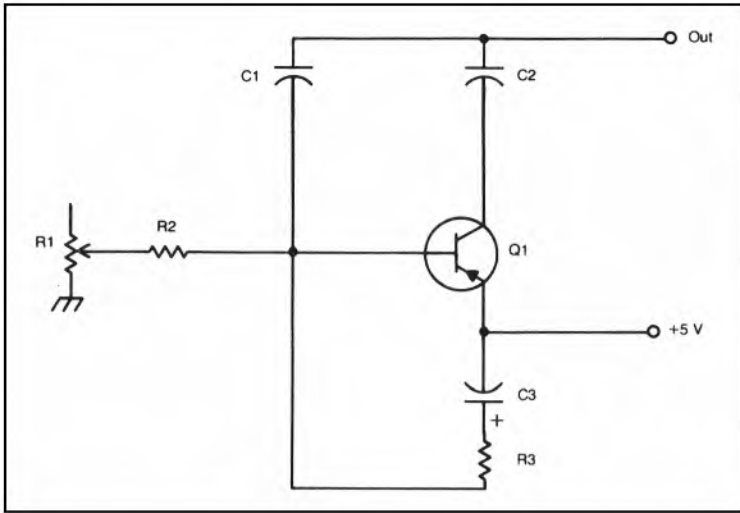


Fig. 3-64. Bird chirp circuit.

either or both of these times can have a drastic effect on a sound. For more information on envelopes, and their effects on a sound, see Chapter 6.

Bird Chirp. For some reason, the chirping of a bird has always been a popular electronic sound effect. Perhaps this is because it happens to be one of the easiest sound effects to generate electronically. Around Christmas time, you're likely to see ornaments that produce a warbling, bird-like sound. These ornaments contain circuits similar to the ones shown in this section.

A typical bird chirp circuit is shown in Fig. 3-64. The parts list is given in Table 3-26. Nothing is particularly critical in this circuit, so you should feel free to experiment with various component values. Particularly, try different values for C3. This capacitor should be an electrolytic type within the 25 μF to 500 μF range.

Table 3-26. Parts List for the Bird Chirp Circuit in Fig. 3-64.

R1	100 k Ω potentiometer
R2	15 k Ω resistor
R3	1.5 k Ω resistor
C1	0.022 μF capacitor
C2	0.047 μF capacitor
C3	see text
Q1	general purpose PNP transistor

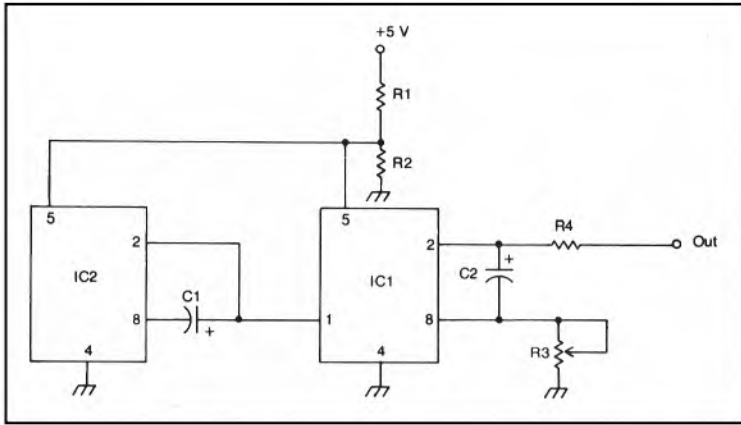


Fig. 3-65. IC bird chirp circuit.

By changing component values, you can create many different sound effects with this circuit. Many will sound nothing at all like a bird, but you may still find them useful.

A second bird chirp circuit is illustrated in Fig. 3-65. This one is built around two oscillator ICs. One oscillator generates the tone, while the second turns the first on and off to create the chirping effect. The parts list for this one is given in Table 3-27.

Once again you should feel free to experiment with various component values. C2 and R3 control the tone frequency, while C1 sets the chirp rate. The smaller the value of C1, the faster the chirp. If C1 is made small enough, frequency modulation (FM) effects will begin to show up in the output signal. (See Chapter 4 for more information on FM, and its effects.)

Putt-Putt. The circuit shown in Fig. 3-66 can generate a number of sputtering effects, ranging from a slow putt-putt-putt effect to the roar of a racing car.

The main components for determining the output's characteristics are R1, and the two capacitors. R2 prevents damage to the

Table 3-27. Parts List for the IC Bird Chirp Circuit in Fig. 3-65.

R1	3.3 k Ω resistor
R2	1.5 k Ω resistor
R3	25 k Ω potentiometer
R4	47 Ω resistor
C1	33 μ F 15 V electrolytic capacitor (see text)
C2	0.1 μ F capacitor
IC1, IC2	3909 LED flasher/oscillator IC

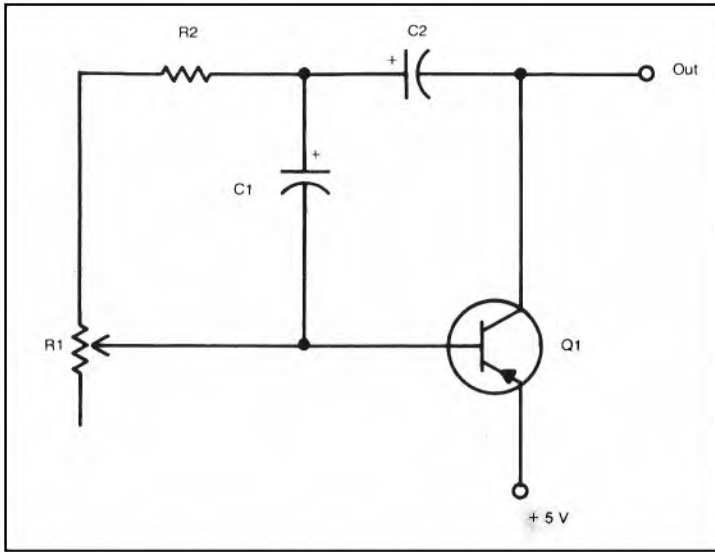


Fig. 3-66. Putt-putt circuit.

transistor from a too low resistance. C1 and C2 can take almost any value from $0.01 \mu\text{F}$ up to about $5 \mu\text{F}$. The two capacitors do not necessarily have to be of equal value. Many intriguing effects can be achieved by using mismatched capacitors.

The parts list for the putt-putt circuit is given in Table 3-28.

Triangle Simulator. The next circuit we'll examine is a triangle simulator. Now, this has nothing to do with triangle waves. This device produces a sound similar to a percussionist's triangle in an orchestra.

An orchestral triangle is a length of metal tube that is bent into the shape of a triangle. When struck, this gadget produces a loud, ringing tone with no definite pitch. Because its pitch is indeterminate, a triangle will sound like it is in tune with any other instrument (or group of instruments) playing in any key. It is a useful instrument for adding a little punctuation to a musical line.

Table 3-28. Parts List for the Putt-Putt Circuit in Fig. 3-66.

R1	1 M Ω potentiometer
R2	15 k Ω resistor
C1, C2	see text
Q1	almost any general purpose PNP transistor

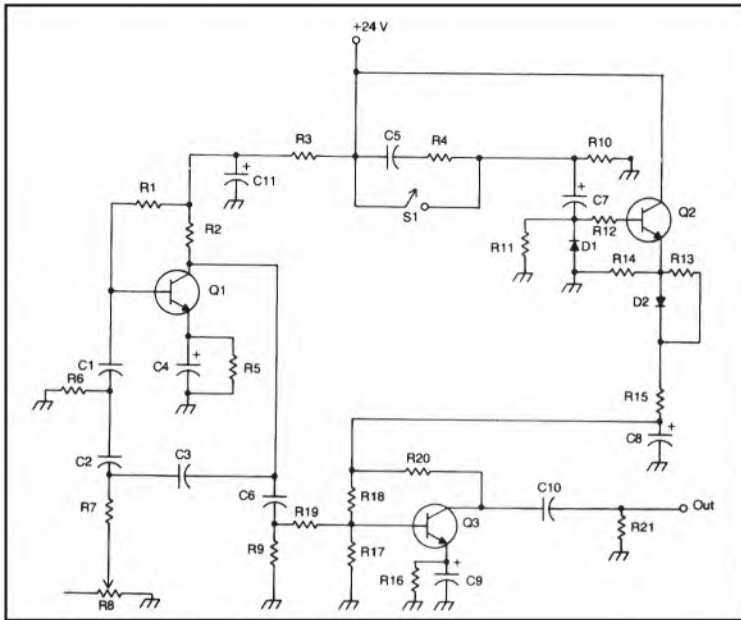


Fig. 3-67. Triangle simulator.

Table 3-29. Parts List for the Triangle Simulator in Fig. 3-67.

R1, R18	120 k Ω resistor
R2, R10, R14, R20	10 K Ω resistor
R3	1.5 k Ω resistor
R4	470 Ω resistor
R5, R16	2.2 k Ω resistor
R6, R17	12 k Ω resistor
R7	2.7 k Ω resistor
R8	25 k Ω potentiometer
R9	3.3 k Ω resistor
R11	470 k Ω resistor
R12	47 k Ω resistor
R13	1 M Ω resistor
R15	820 Ω resistor
R19	220 k Ω resistor
R21	22 k Ω resistor
C1, C2, C3	1000 pF capacitor
C4	10 μ F 35 V electrolytic capacitor
C5	0.1 μ F capacitor
C6	390 pF capacitor
C7	1.5 μ F 35 V electrolytic capacitor
C8, C9	30 μ F 35 V electrolytic capacitor
C10	470 pF capacitor
C11	330 μ F 35 V electrolytic capacitor
S1	NO SPST pushbutton switch
D1, D2	IN4148 or IN914
Q1, Q2, Q3	NPN transistor (GE-20, Motorola HEP-50, or Radio Shack RS 2016)

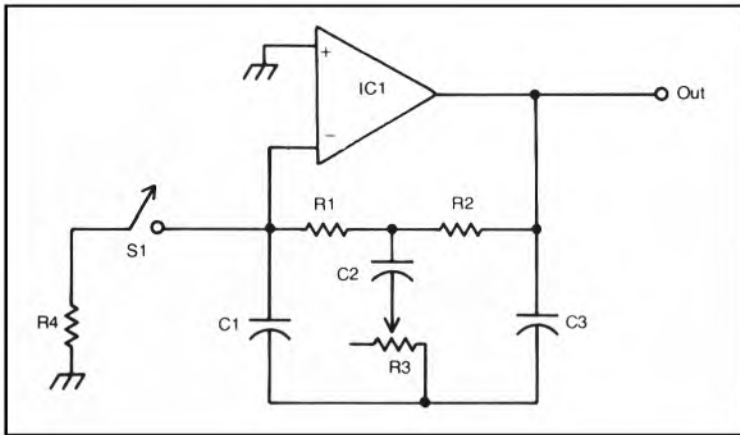


Fig. 3-68. Gong circuit.

The schematic diagram for the triangle simulator is given in Fig. 3-67. Table 3-29 is the parts list.

Potentiometer R8 controls the apparent size of the triangle. S1 must be a normally open type. Each time this switch is closed, the electronic triangle is struck, and the sound is produced.

Gong. Figure 3-68 shows a circuit for producing a gong, or large bell effect. The parts list is given in Table 3-30.

R3 should be a miniature trimpot. Once you've built the circuit, adjust this control until it breaks into oscillation and a constant tone is heard. Then back off on the trimpot setting until the oscillation just stops. Now every time switch S1 (a normally open type) is closed, the gong will be struck, or the bell will be rung.

Try experimenting with other values for resistors R1 and R2. By substituting resistors from 27 k Ω to about 680 k Ω , a wide variety of sounds can be generated, ranging from a tiny tinkling bell up through the gong to struck bamboo rods and drums. For realistic effects R1 and R2 should have identical values. However, feel free to experiment with mis-matched values for unusual effects.

Table 3-30. Parts List for the Gong Circuit in Fig. 3-68.

R1, R2	220 k Ω resistor (see text)
R3	1 M Ω potentiometer
R4	10 k Ω resistor
C1, C3	0.001 μ F capacitor
C2	0.005 μ F capacitor
S1	NO SPST push-button switch
IC1	almost any op amp IC (741, or similar)

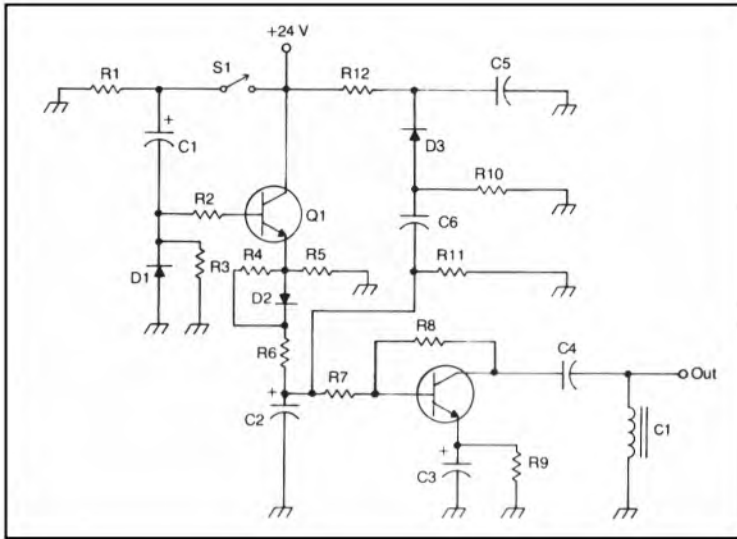


Fig. 3-69. Cymbal crash synthesizer.

Cymbal Crash. Figure 3-69 shows a circuit that can simulate the crash of a cymbal. The parts list is given in Table 3-31.

Switch S1 is once again a normally open type. Closing this switch briefly is the electronic equivalent of striking the cymbal.

Diode D3 should be selected for internal noise. This part of the circuit is a simple noise generator similar to the one shown back in Fig. 3-60.

The coil is an iron core type with a dc resistance of about 90 ohms ± 10 percent.

Table 3-31. Parts List for the Cymbal Crash Synthesizer in Fig. 3-69.

R1, R5, R8, R11	10 k Ω resistor
R2	47 k Ω resistor
R3	470 k Ω resistor
R4	1.5 M Ω resistor
R6	680 Ω resistor
R7, R12	120 k Ω resistor
R9	2.2 k Ω resistor
R10	100 K Ω resistor
C1, C3	1.5 μ F 35 v electrolytic capacitor
C2	50 μ F 35 v electrolytic capacitor (see text)
C4, C6	0.047 μ F capacitor
C5	0.1 μ F capacitor
L1	coil—see text
S1	NO SPST push-button switch
Q1, Q2	NPN transistor (GE-20, Motorola HEP-50, or Radio Shack RS2016)

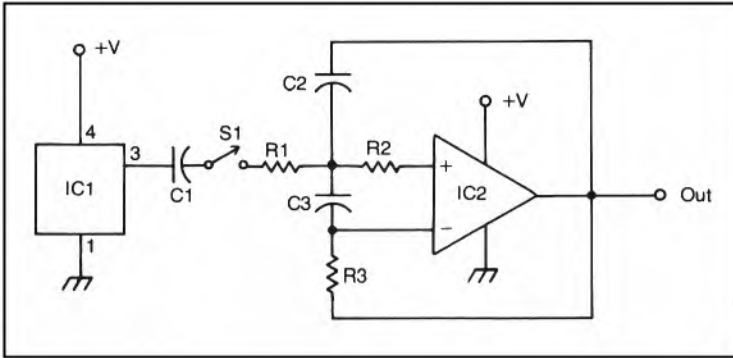


Fig. 3-70. Snare drum simulator.

Experiment with different values for C2. This electrolytic capacitor sets the decay time, or how long the cymbal will continue to ring after it is struck. For realistic sounding cymbals, this capacitor should have a value in the 50 μF to 100 μF range.

Snare Drum. Figure 3-70 shows a circuit for synthesizing the sound of a snare drum. The parts list is given in Table 3-32.

Increasing the values of capacitors C2 and C3 will produce the sound of a larger drum. Reducing the values of these components will give the effect of a smaller drum.

SN76477/88 Sound Effects Synthesizer. Many sound effects can be created with a relatively new IC device from Texas Instruments (and distributed by many dealers). This is the SN76477 Sound Effects Synthesizer chip. Its pin-out diagram is shown in Fig. 3-71.

A more advanced version of this device is the SN76488 which has an on-chip amplifier. The pin-out diagram for this IC is given in Fig. 3-72. Notice that the two chips are not pin to pin compatible, although they operate similarly.

Table 3-32. Parts List for the Snare Drum Simulator in Fig. 3-70.

R1	3.9 k Ω resistor
R2	1 k Ω resistor
R3	6.8 k Ω resistor
C1	0.047 μF capacitor
C2, C3	0.022 μF capacitor (see text)
S1	NO SPST push-button switch
IC1	Digital noise generator IC (MM5837)
IC2	op amp IC (741, or similar)

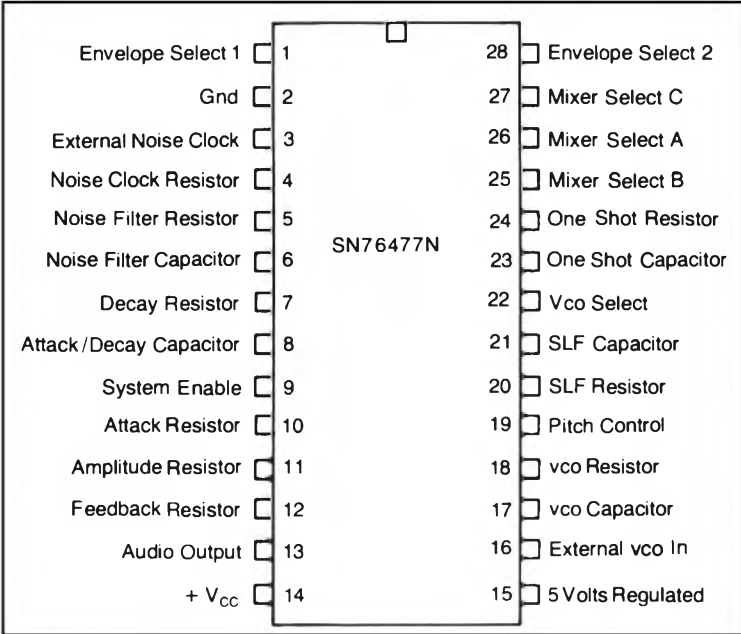


Fig. 3-71. SN76477 Sound Effects Generator.

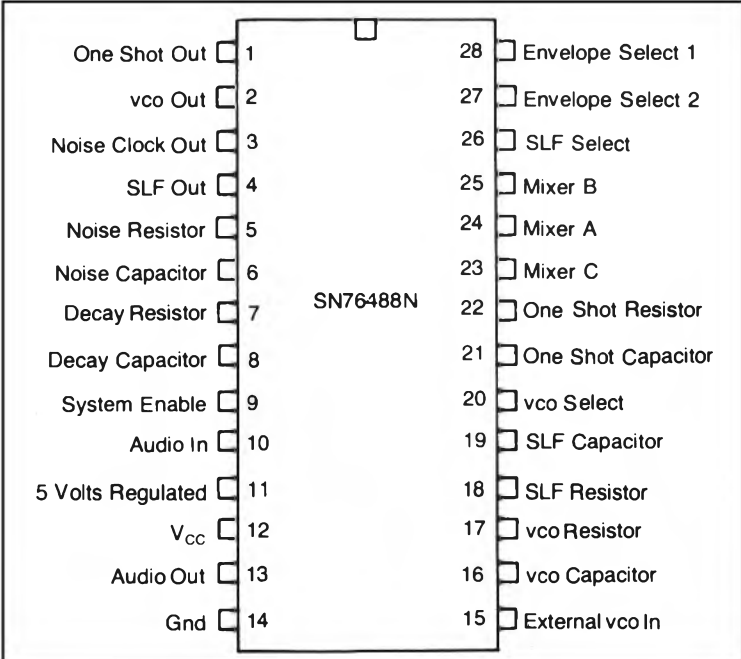


Fig. 3-72. SN76488 Sound Effects Generator.

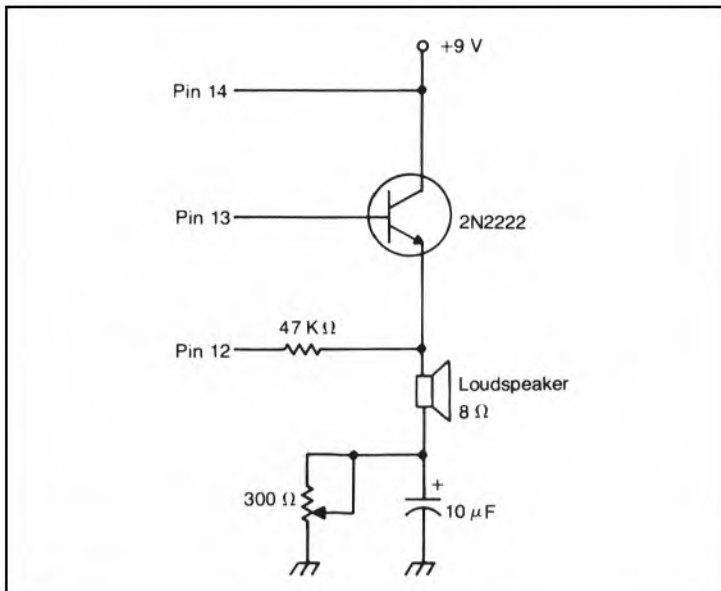


Fig. 3-73. Simple amplifier for the SN76477.

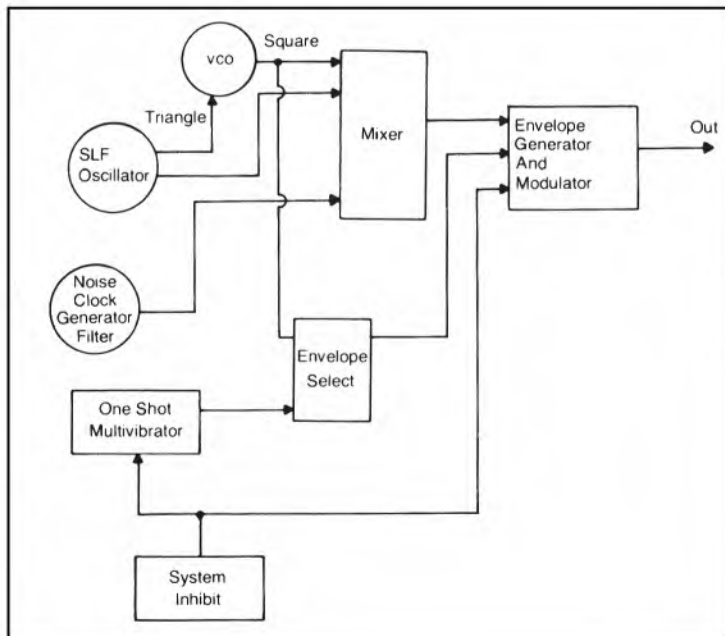


Fig. 3-74. SN76477/88 Sound Effects Generator.

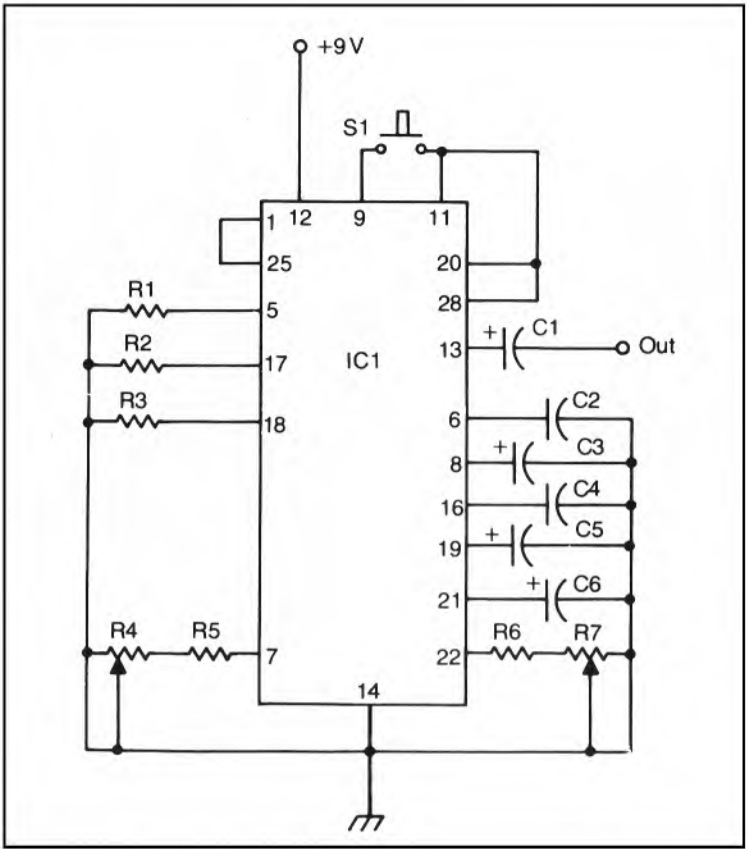


Fig. 3-75. Sound effect circuit.

Table 3-33. Parts List for Fig. 3-75.

R1	680 k Ω resistor
R2	470 k Ω resistor
R3	1 M Ω resistor
R4	100 k Ω potentiometer
R5	330 k Ω resistor
R6	56 k Ω resistor
R7	10 k Ω potentiometer
C1	100 μ F 30 V electrolytic capacitor
C2	470 pF capacitor
C3	2.2 μ F 30 V electrolytic capacitor
C4	0.0047 μ F capacitor
C5	0.33 μ F capacitor
C6	33 μ F 30 V electrolytic capacitor
S1	SPST Normally Open push-button switch
IC1	SN76488N sound effects generator IC

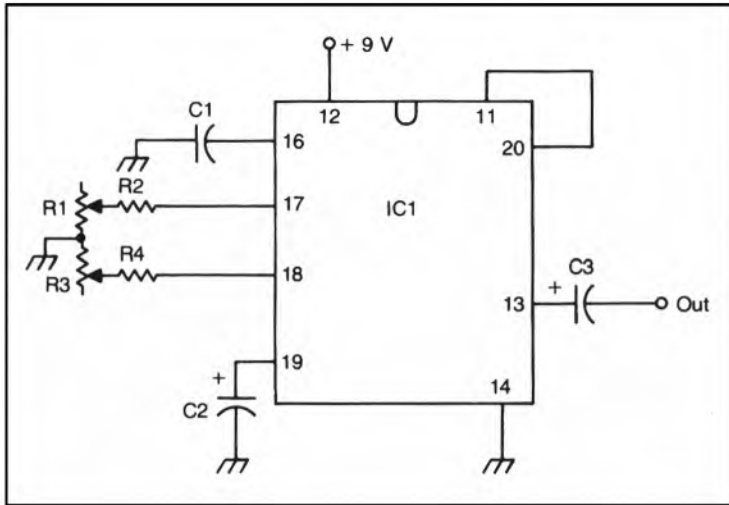


Fig. 3-76. Sound effect circuit.

A simple amplifier for use with the SN76477 is illustrated in Fig. 3-73. The output of the SN76488 may be fed directly to a loudspeaker, or to the input of a larger amplifier.

Of course, any sound effect from either of these devices may be fed through other synthesizer modules for manipulation of various aspects of the signal.

A block diagram of both chips is shown in Fig. 3-74. You can see that this unit is virtually a complete (albeit simple) sound synthesizer on a single chip. You could try to play musical lines with the built-in vco, but this unit is not really suitable for music per se. The SN76477/88 was designed for the creation of sound effects and that is where it shines.

Some typical circuits are shown in Fig. 3-74 through Fig. 3-79. The parts lists are given in Table 3-33 through Table 3-37.

You could also build a universal sound effects generator by connecting switchable capacitors and potentiometers to the appropriate pins.

Table 3-34. Parts List for Fig. 3-76.

R1, R3	2.5 k Ω potentiometer
R2, R4	470 Ω resistor
C1	0.1 μ F capacitor
C2	10 μ F electrolytic capacitor
C3	100 μ F electrolytic capacitor
IC1	SN76488 sound effect generator IC

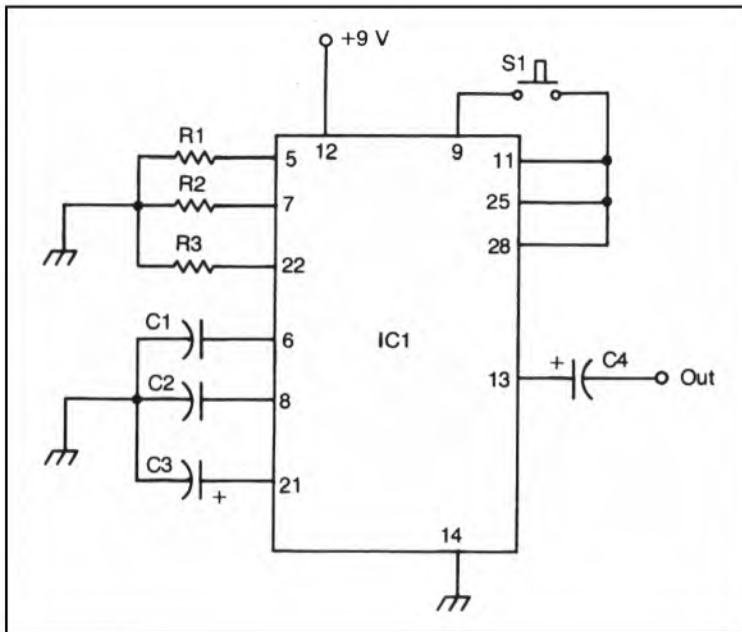


Fig. 3-77. Sound effect circuit.

Table 3-35. Parts List for Fig. 3-77.

R1	6.2 k Ω resistor
R2, R3	470 k Ω resistor
C1	500 pF capacitor
C2	0.5 μ F capacitor
C3	50 μ F electrolytic capacitor
C4	100 μ F electrolytic capacitor
S1	Normally Open SPST push button switch
IC1	SN76488 sound effect generator IC

Table 3-36. Parts List for Fig. 3-78.

R1, R3	5 k Ω potentiometer
R2, R4	1 k Ω resistor
C1	10 μ F electrolytic capacitor
C2	35 μ F electrolytic capacitor
C3	100 μ F electrolytic capacitor
IC1	SN76488 sound effect generator IC

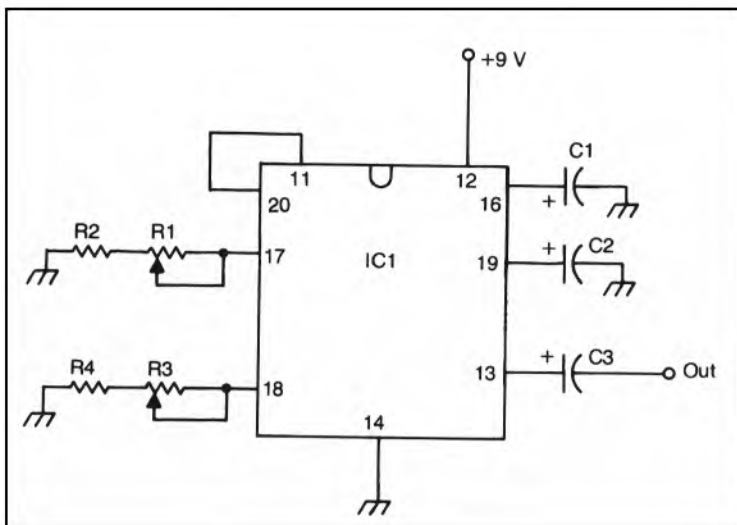


Fig. 3-78. Sound effect circuit.

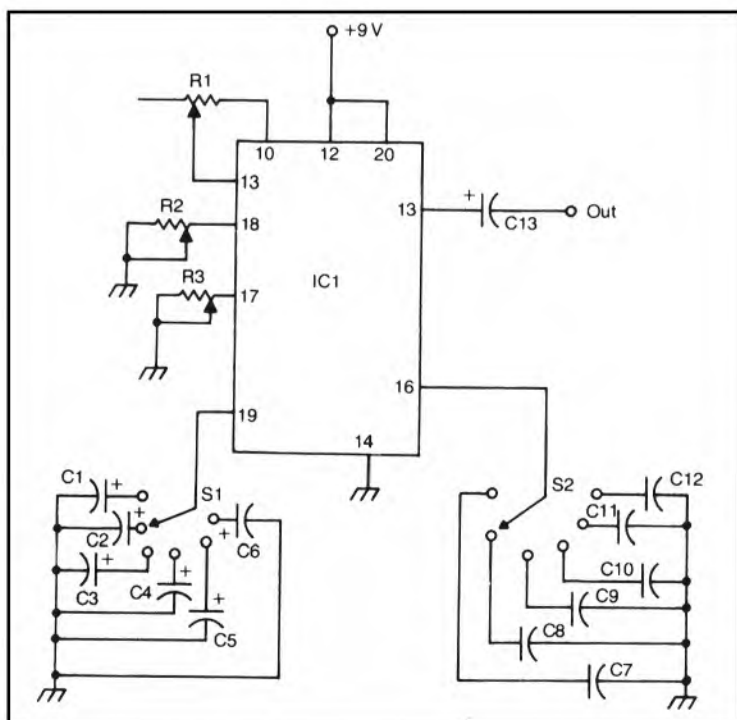


Fig. 3-79. Sound effect circuit.

Table 3-37. Parts List for Fig. 3-79.

R1	10 k Ω potentiometer
R2	500 k Ω potentiometer
R3	250 k Ω potentiometer
C1	2.2 μ F electrolytic capacitor
C2	5 μ F electrolytic capacitor
C3	10 μ F electrolytic capacitor
C4	22 μ F electrolytic capacitor
C5	47 μ F electrolytic capacitor
C6, C13	100 μ F electrolytic capacitor
C7	0.001 μ F capacitor
C8	0.0047 μ F capacitor
C9	0.01 μ F capacitor
C10	0.022 μ F capacitor
C11	0.047 μ F capacitor
C12	0.1 μ F capacitor
S1, S2	Single-pole/6 throw rotary switch
IC1	SN76488N complex sound generator IC

EXTERNAL INPUT SIGNALS

While most electronic music begins with some form of oscillator, almost any ac signal in the audio range can be manipulated electronically for synthesis effects.

The signal from the pickup of an electric guitar is often used. Any sound that can be picked up by a microphone can conceivably be processed through synthesis modules.

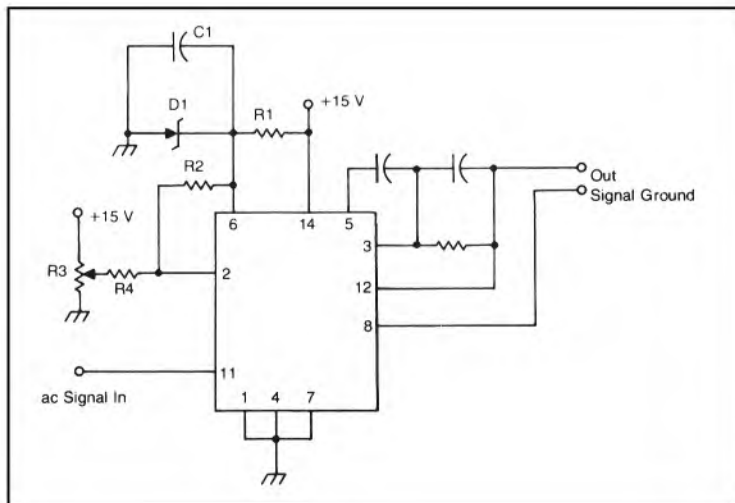


Fig. 3-80. Frequency-to-Voltage converter.

You should use care when applying external signals to an electronic music synthesizer. If the signal level is too high, distortion will result, and some components could be damaged. High level signals should be passed through an attenuator before being processed by the synthesizer.

On the other hand, a too low level signal could pick up an objectionable amount of noise (in the conventional sense). It's even possible that an input signal could be too weak to drive a module at all, in which case the output would be zero. If you run into this kind of problem, you should add a preamplifier.

The most common use of an external signal is as the starting point for the synthesized sound. External signals can also be put to work as control voltages. Here is where we can use the other mode of the 9400 Voltage-to-Frequency/Frequency-to-Voltage Converter described earlier.

The circuit shown in Fig. 3-80 accepts an input signal (almost any waveform will be acceptable) and produces a voltage at the output that is directly proportional to the frequency of the input signal. For example, a guitarist can use the phrasing of the notes of a guitar to control an oscillator and can produce a completely electronic sound.

The parts list for the Frequency-to-Voltage converter circuit is given in Table 3-38.

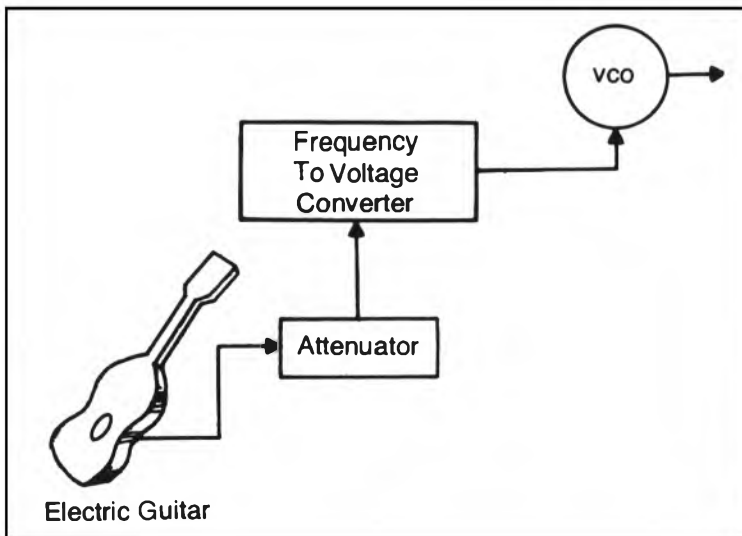


Fig. 3-81. Driving a vco with an electric guitar.

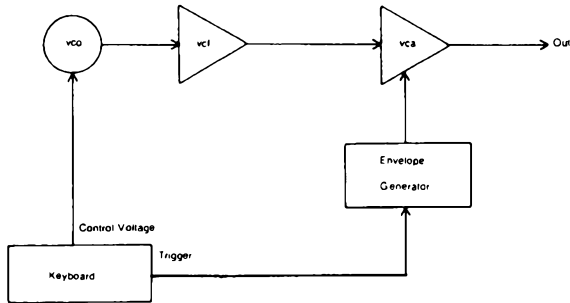
Table 3-38. Parts List for the Frequency-to-Voltage Converter in Fig. 3-80.

R1,R2	10 k Ω resistor
R3	100 k Ω potentiometer
R4	470 k Ω resistor
D1	5.1 V zener diode
IC1	9400 V-F/F-V converter IC

As you can tell by the length of this chapter, there are many ways to initiate a signal in an electronic music synthesizer. In a sense, we have barely scratched the surface of the possibilities, although there are certainly enough circuits in this chapter to get you started.

In the following chapters, we will examine ways that these signals may be electronically manipulated to create new sounds.

Chapter 4



Blending Signal Sources

While many effects can be achieved by manipulating the signal from a single oscillator (or sound effects circuit, or external input), electronic musicians generally seek more complex textures. Two or more signals may be combined in any of a variety of ways to produce thousands of new effects.

In some cases, the signals will be heard as separate voices as with traditional instruments. For example, in a duet for piano and violin, we can tell that two instruments are playing. In electronic music, however, the boundaries between voices are sometimes blurred. Often two or more simultaneous signals are perceived as a single complex voice.

We can also use one signal to modify a second signal, producing a third signal unlike either of the original sources.

In this chapter, we will explore some of the myriad ways electronic music signals can be combined and blended.

ADDITIVE SYNTHESIS

When two or more simple signals are simultaneously sounded, they may appear to be a single complex signal from a single source. This effect is called *additive synthesis*. Most early electronic music relied entirely on the techniques of additive synthesis. Each voice was painstakingly built up by combining sine waves on tape.

Theoretically, any complex signal could be synthesized by adding sine waves of the correct frequencies and relative am-

plitudes. For example, mixing together the proper level sine waves at 300 Hz, 600 Hz, 900 Hz, 1200 Hz, 1500 Hz, 1800 Hz, 2100 Hz, and so forth, will produce a 300 Hz sawtooth wave.

In practice, however, building up complex signals from pure sine waves is rarely done in modern electronic music. This is because of the tedium and complexities involved in the process and the practical problems of keeping dozens of sine wave oscillators in perfect tune with each other. It is far easier to generate common signals like sawtooth waves and rectangle waves directly (see Chapter 3).

Additive synthesis is most useful in creating signals that can't be easily generated by a single oscillator. In addition, the electronic musician is by no means limited to sine waves when working with additive synthesis. Any combination of waveforms may be used.

Let's assume we want to create a signal with a harmonic content like that shown in Table 4-1. We could start out with a sawtooth wave (which contains all of the harmonics) and filter out the ones we don't want (second, tenth, fourteenth, etc.). This technique is called subtractive synthesis and will be discussed in Chapter 5.

In some cases, subtractive synthesis will be easier than additive synthesis, but in this example, we would need five very tight

Table 4-1. Additive Synthesis. The First Example's Desired Harmonic Content.

Fundamental	100 Hz	330 Hz
Third	300 Hz	990 Hz
Fourth	400 Hz	1320 Hz
Fifth	500 Hz	1650 Hz
Sixth	600 Hz	1980 Hz
Seventh	700 Hz	2310 Hz
Eighth	800 Hz	2640 Hz
Ninth	900 Hz	2970 Hz
Eleventh	1100 Hz	3630 Hz
Twelfth	1200 Hz	3960 Hz
Thirteenth	1300 Hz	4290 Hz
Fifteenth	1500 Hz	4950 Hz
Sixteenth	1600 Hz	5280 Hz
Seventeenth	1700 Hz	5610 Hz
Nineteenth	1900 Hz	6270 Hz
Twentieth	2000 Hz	6600 Hz
Twenty-first	2100 Hz	6930 Hz
Twenty-third	2300 Hz	7590 Hz
Twenty-fourth	2400 Hz	7920 Hz
Twenty-fifth	2500 Hz	8250 Hz
All higher harmonics are too weak to be audible.		

band reject filters (see Chapter 5). It would be far more convenient to create this signal by additive synthesis. It can be done with three oscillators—a square wave oscillator tuned to the desired fundamental, a sawtooth oscillator tuned to the output signal's fourth harmonic, and a sine wave at the resulting sixth harmonic. This solution is summarized in Table 4-2. The waveforms are shown in Fig. 4-1 and Fig. 4-2.

Additive synthesis also comes in handy when we want to create signals with nonharmonic overtones. For instance, consider the signal defined in Table 4-3. This oddball waveform could be made up by combining three square waves with fundamental frequencies of 150 Hz, 325 Hz, and 500 Hz. The component waveforms are analyzed in Table 4-4.

If you need all of the harmonics, but don't have a sawtooth wave oscillator handy, you could combine two square wave oscillators. One should have a fundamental frequency that is twice (one octave above) that of the other. This is demonstrated in Table 4-5.

Table 4-2. Additive Synthesis. The First Example's Solution.

Square Wave			
Fundamental	100 Hz	330 Hz	
Third	300 Hz	990 Hz	
Fifth	500 Hz	1650 Hz	
Seventh	700 Hz	2310 Hz	
Ninth	900 Hz	2970 Hz	
Eleventh	1100 Hz	3630 Hz	
Thirteenth	1300 Hz	4290 Hz	
Fifteenth	1500 Hz	4950 Hz	
Seventeenth	1700 Hz	5610 Hz	
Nineteenth	1900 Hz	6270 Hz	
Twenty-first	2100 Hz	6930 Hz	
Twenty-third	2300 Hz	7590 Hz	
Twenty-fifth	2500 Hz	8250 Hz	
Sawtooth Wave			Complex Wave Harmonic
Fundamental	400 Hz	1320 Hz	4th
Second	800 Hz	2640 Hz	8th
Third	1200 Hz	3960 Hz	12th
Fourth	1600 Hz	5280 Hz	16th
Fifth	2000 Hz	6600 Hz	20th
Sixth	2400 Hz	7920 Hz	24th
Use a low-pass filter to remove all higher harmonics.			
Sine Wave			
Fundamental	600 Hz	1980 Hz	6th

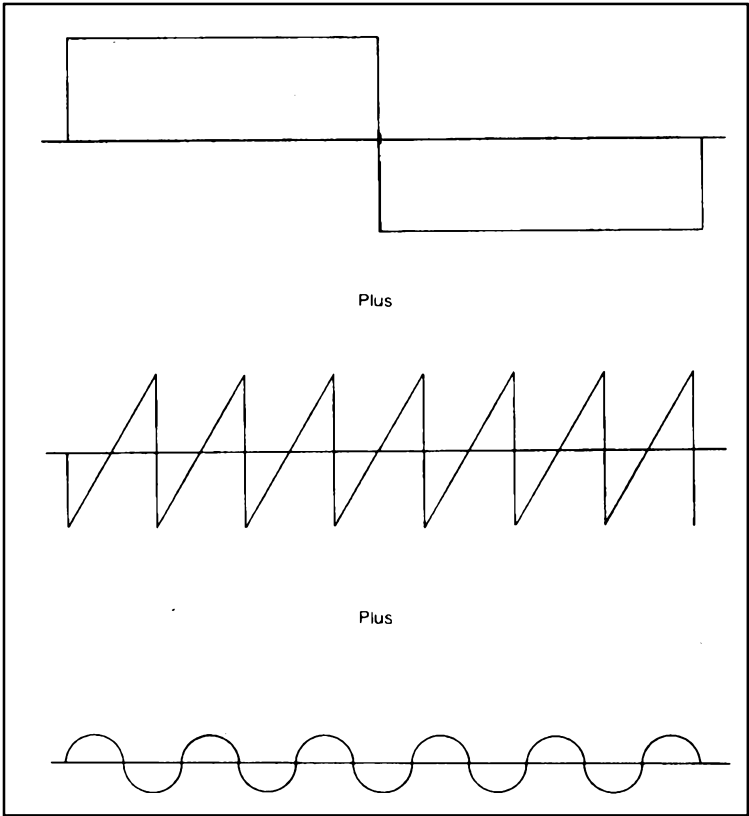


Fig. 4-1. Additive synthesis input waveforms.

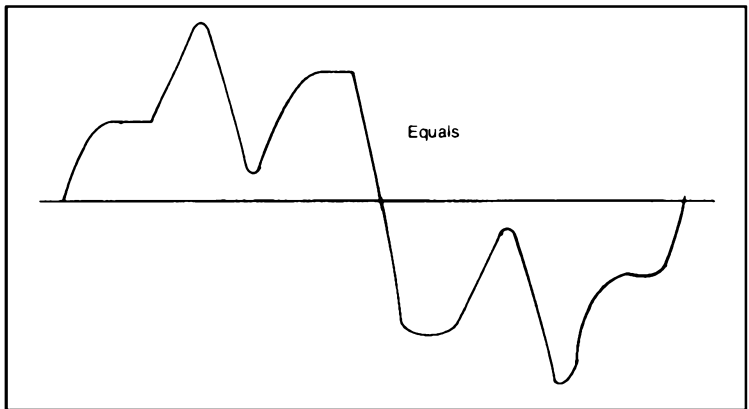


Fig. 4-2. Additive synthesis output waveforms.

150 Hz Fundamental
325 Hz _____
450 Hz Third
500 Hz _____
750 Hz Fifth
975 Hz _____
1050 Hz Seventh
1350 Hz Ninth
1500 Hz Tenth
1625 Hz _____
1650 Hz Eleventh
1950 Hz Thirteenth
2250 Hz Fifteenth
2275 Hz _____
2500 Hz _____
2925 Hz _____
3500 Hz _____
3575 Hz _____
4225 Hz _____
4500 Hz Thirtieth
4875 Hz _____
5500 Hz _____
6500 Hz _____
7500 Hz Fiftieth

Table 4-3. Additive Synthesis. The Second Example's Desired Overtone Series.

To achieve good results with additive synthesis, you need good oscillators with a minimum of frequency drift. All of the oscillators must track each other (stay in tune) with as much precision as possible. Capabilities for synchronizing the oscillators are also helpful. Some of the voltage-controlled oscillators in Chapter 3 have synchronization inputs. However, you may get some very peculiar effects if you synchronize two or more oscillators whose

Table 4-4. Additive Synthesis. The Second Example's Solution.

Harmonics	Square Wave 1	Square Wave 2	Square Wave 3
Fundamental	150 Hz	325 Hz	500 Hz
Third	450 Hz	975 Hz	1500 Hz
Fifth	750 Hz	1625 Hz	2500 Hz
Seventh	1050 Hz	2275 Hz	3500 Hz
Ninth	1350 Hz	2925 Hz	4500 Hz
Eleventh	1650 Hz	3575 Hz	5500 Hz
Thirteenth	1950 Hz	4225 Hz	6500 Hz
Fifteenth	2250 Hz	4875 Hz	7500 Hz

All higher harmonics are too weak to be of significance.

Table 4-5. Additive Synthesis. The Third Example Gets All of the Harmonics From Two Square Waves.

Square Wave 1	Square Wave 2	Result
200 Hz		Fundamental
	400 Hz	Second
600 Hz		Third
	800 Hz	Fourth
1000 Hz		Fifth
	1200 Hz	Sixth
1400 Hz		Seventh
	1600 Hz	Eighth
1800 Hz		Ninth
	2000 Hz	Tenth
2200 Hz		Eleventh
	2400 Hz	Twelfth
2600 Hz		Thirteenth
	2800 Hz	Fourteenth
3000 Hz		Fifteenth
	3200 Hz	Sixteenth

outputs are not harmonically related. This may or may not be desirable. The effect is illustrated in Fig. 4-3.

Another potential problem to keep in mind when working with additive synthesis is that in some cases a very complex signal (especially if it has many strong nonharmonic overtones) may lose its definite sense of pitch. This may be useful as accompaniment, but it should be obvious, that you can't play a melody without a firm pitch.

Additive synthesis can be very exciting as you discover strange new waveshapes. It can get tedious unless you're tolerant of sloppy results, however.

Most modern commercial electronic music synthesizers emphasize subtractive synthesis (see Chapter 5) over additive synthesis, but the smart and creative electronic musician will take advantage of both techniques.

BLENDING METHODS

When working with additive synthesis, and many other electronic music synthesis techniques, it is often necessary to combine several output signals into a single input line. Conversely, it is also often necessary to feed a single output signal into several separate inputs. In either case, what is required is a method for combining multiple lines into a single line.

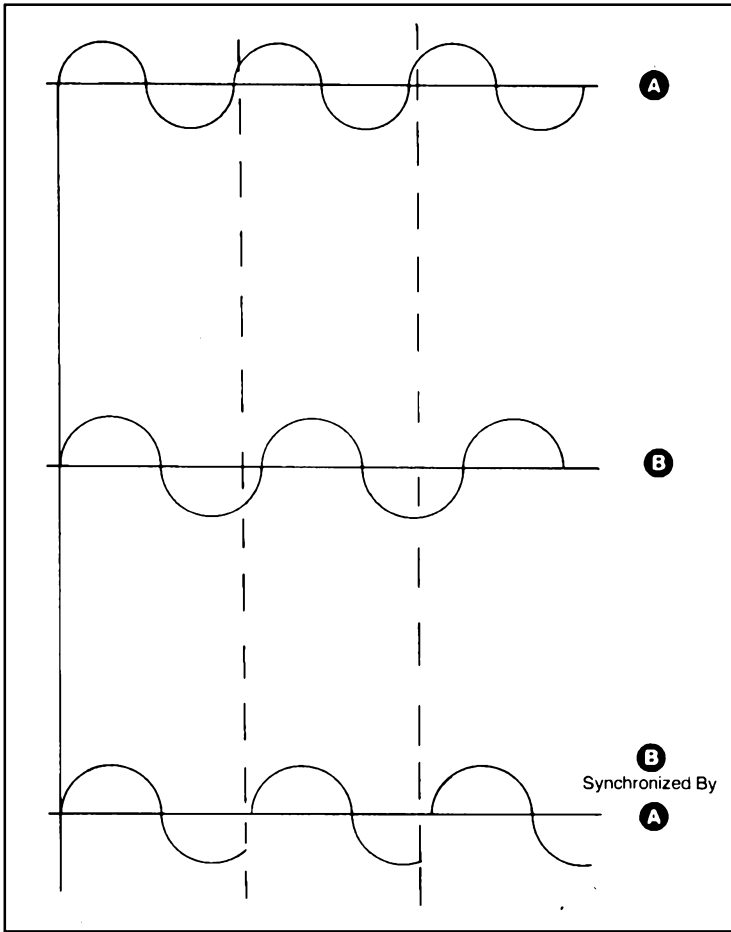


Fig. 4-3. Synchronizing nonharmonic frequencies.

Patch Panels

The easiest, most direct way of doing this is to use a patch panel. Many modular synthesizers include patch panels. Normalized synthesizers (see Chapter 10) use hard-wired switches.

A *patch panel* is simply a series of jacks wired together in parallel, as shown in Fig. 4-4. Any and all lines plugged into the jacks will be shorted together, and will behave electrically like a single line. There is no difference between inputs and outputs.

This is not the best possible solution in all cases. Often interconnected inputs, or interconnected outputs may interact with each other, causing the related circuits to behave in undesirable ways.

Also, splitting a single output signal into many input lines may lead to excessive noise, as the signal level is a constant and will be divided between each of the lines. Both of these problems can be dealt with by using active mixers, as discussed in the next section of this chapter.

A degree of isolation can be achieved in a patch panel by hooking up capacitors between the jacks, as shown in Fig. 4-5. This will block any dc component of the signals, allowing only the ac waveforms to pass. Of course, this means such a patch panel could not be used for dc control voltages.

Many modular synthesizers use different size plugs for control voltages and ac signals. This is a good idea, because it is easier to keep track of what you're doing. However, as mentioned in an earlier chapter, there is no reason why an ac signal can't be used as a control voltage too. It is a good idea to include a patch panel that connects both types of jack, for easy conversion.

Patch panels aren't perfect, and they're certainly not an elegant approach to electronic music synthesis, but they can come in extremely handy and are quite inexpensive to include in a synthesis system. All you need is a front panel, the appropriate jacks, a few capacitors (if needed), and a couple of inches of wire. It is undoubtedly worthwhile to include a couple of independent patch panels into any modular electronic music synthesizer.

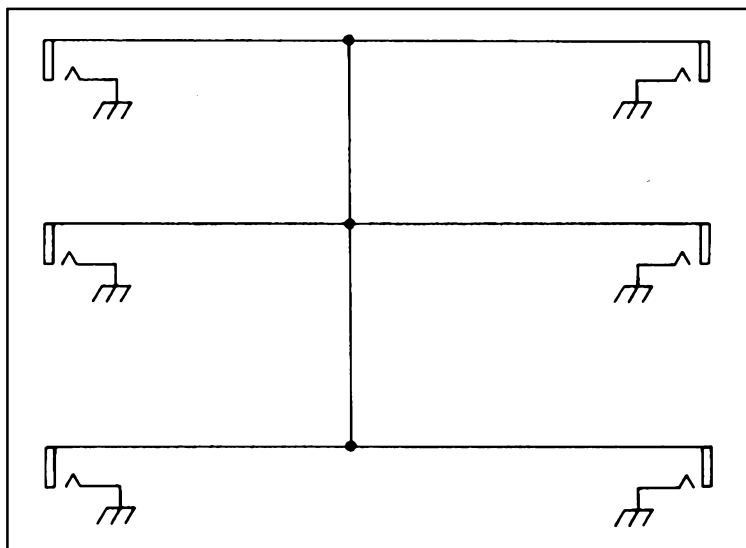


Fig. 4-4. Patch panel.

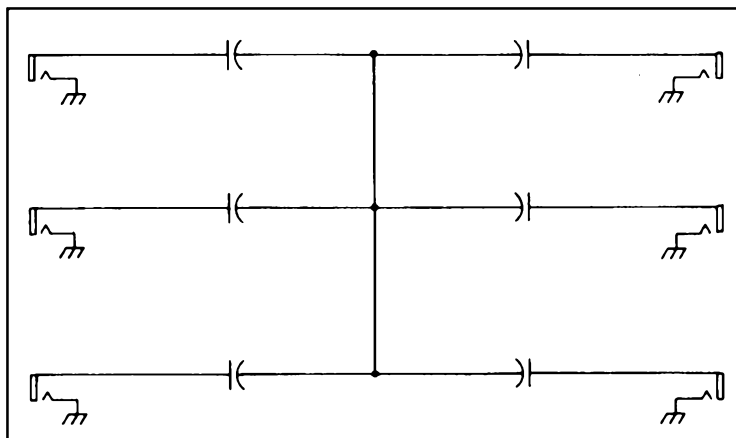


Fig. 4-5. Patch panel with isolating capacitors.

Mixers

It was mentioned earlier that circuits connected together with a patch panel may suffer from interaction or signal degradation. Moreover, a patch panel offers no facilities for controlling the relative amplitudes of the various signals. For these tasks you need a circuit called a *mixer*.

A super-simple passive mixer circuit is shown in Fig. 4-6. This circuit consists simply of three variable resistors or potentiometers. R1 can attenuate the level of signal 1 while R2 attenuates the level of signal 2. R3 serves as a master attenuation control, and affects the total output (both signal 1 and signal 2) equally.

This circuit can be used in a few noncritical applications, but it leaves a lot to be desired. For one thing, it can only attenuate (or reduce the level of) the signal. No provision is made for boosting or amplifying weak signals. A potentially more serious problem with

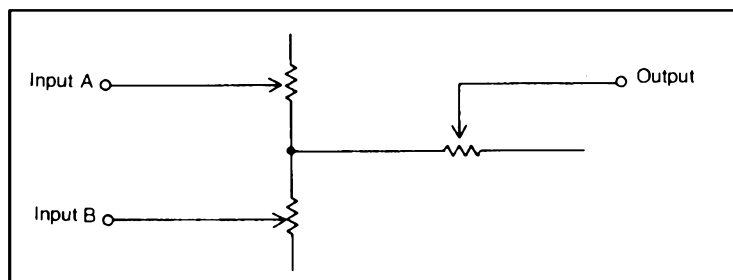


Fig. 4-6. Simple passive mixer.

this simplistic approach, is that the circuits connected to the inputs of the mixer are free to interact with each other.

Most, if not all, sound synthesis systems contain at least one (and preferably several) active mixers. An active mixer circuit is built around semiconductor devices with amplification abilities. In a properly designed mixer circuit, signal interaction should not be a problem. Signals should not be able to get from one input to another.

In an active mixer, signals may be attenuated, or boosted, as necessary. Moreover, input and output impedances can be matched for best signal transfer and lowest distortion. With a simple passive resistance circuit, like the one shown in Fig. 4-6, the impedance will vary with each adjustment of any of the potentiometers.

A simple active mixer circuit is shown in Fig. 4-7. The parts list is given in Table 4-6.

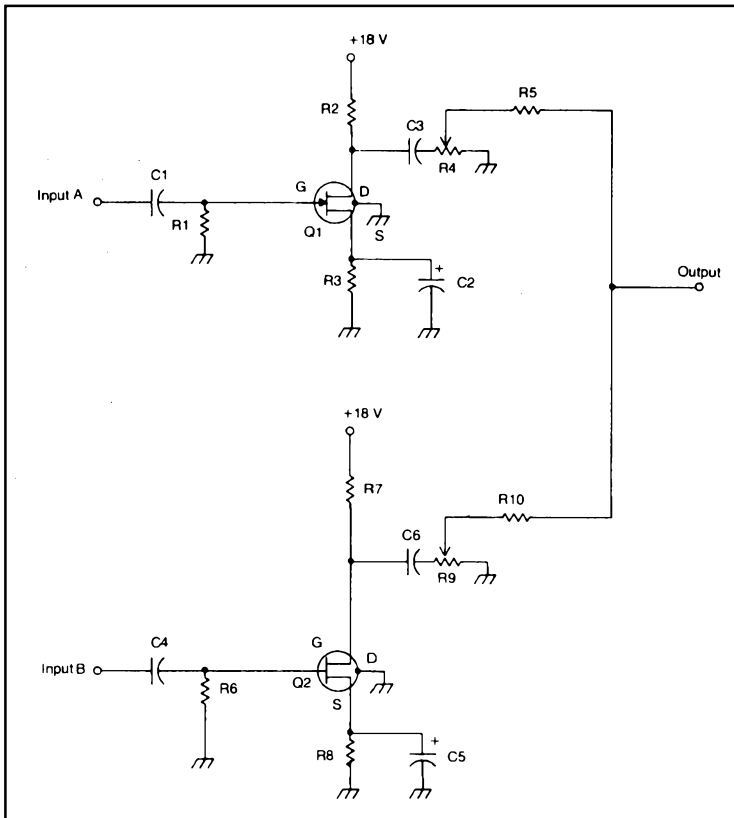


Fig. 4-7. Transistor mixer.

Table 4-6. Parts List for Fig. 4-7.

R1, R6	2.2 M Ω resistor
R2, R7	560 Ω resistor
R3, R8	6.8 k Ω resistor
R4, R9	500 k Ω potentiometer
R5, R10	100 k Ω resistor
C1, C4	0.05 μ F capacitor
C2, C5	25 μ F electrolytic capacitor
C3, C6	0.1 μ F capacitor
Q1, Q2	FET (Radio Shack RS2028 or similar)

Op amp ICs are excellent building blocks for high quality mixers. A simple circuit is illustrated in Fig. 4-8. The parts list is given in Table 4-7.

While a garden variety op amp chip like the 741 may be used in this type of application, it is preferable to use a high quality, low-noise type IC, especially if low level signals are being fed through the circuit. Cheap op amps like the 741 tend to generate some random noise which could prove objectionable.

An improved, deluxe op amp mixer circuit is shown in Fig. 4-9. By using an independent op amp envelope follower (unity gain amplifier) for each input, nearly perfect isolation between inputs can be achieved. Once again, it is worth the extra expense to use high quality low noise op amp ICs. The parts list for this circuit is given in Table 4-8.

An intriguing, and often useful variation on the basic op amp mixer is illustrated in Fig. 4-10. An ordinary op amp mixer is called an adder, because each of the inputs are added to the output. The circuit of Fig. 4-10, on the other hand, is called a subtractor. The signal fed through input 2 is subtracted from the signal being passed

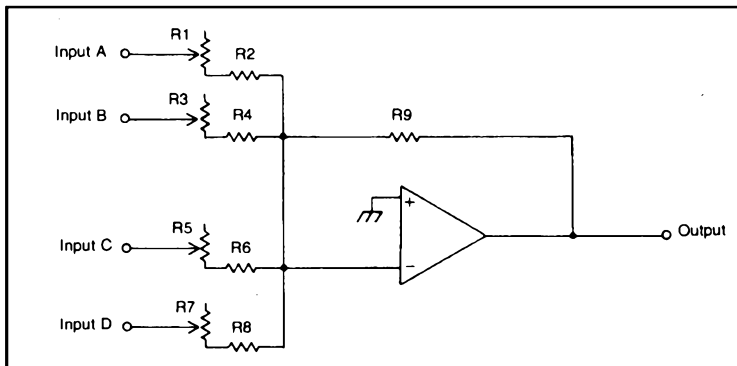


Fig. 4-8. Op amp mixer.

Table 4-7. Parts List for Fig. 4-8.

R1, R3, R5, R7	100 k Ω potentiometer
R2, R4, R6, R8	1 k Ω resistor
R9	10 k Ω resistor
IC1	low noise op amp IC

through input 1, and the difference between the two signals appears at the output of the circuit. Figure 4-11 illustrates the effect. A parts list for the subtracter circuit of Fig. 4-10 is given in Table 4-9.

Just as it is often necessary in electronic sound synthesis to combine several output signals into a single input (by using a mixer), it is also often necessary to split up a one signal to feed several different inputs. A simple patch panel (as discussed earlier) can usually do the trick, but in some instances, signal interaction and/or impedance mismatches can cause problems. The various inputs may also have to be fed with different level signals.

The answer to this problem is to use a sort of reverse mixer, or active *signal splitter*. Such a circuit is shown in Fig. 4-12. The parts list is given in Table 4-10.

There really isn't too much to say about mixers. They are really nothing more than straightforward multiple input amplifier circuits, and are not, in themselves, exciting. But they are absolutely essential for any kind of serious electronic music synthesis, beyond the level of hokey bleeps and bloops that quickly wear out their welcome. Be sure to include two or three good mixers in your synthesis system. It's more than worth the cost.

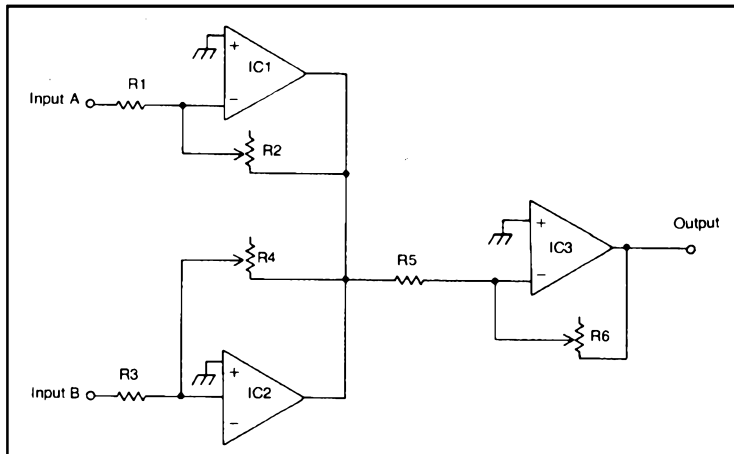


Fig. 4-9. Deluxe op amp mixer.

Table 4-8. Parts List for Fig 4-9.

R1, R3, R5	10 k Ω resistor
R2, R4, R6	100 k Ω potentiometer
IC1, IC2, IC3	low noise op amp IC

Impedance Matching

The problem of impedance matching was mentioned briefly in the section on mixers, but the subject is important enough to warrant a somewhat more detailed discussion here.

Impedance is essentially ac resistance. Every signal source output has a characteristic impedance. Similarly, every input will also have a specific impedance. If the output and input impedances match exactly, most of the signal power will get through from the

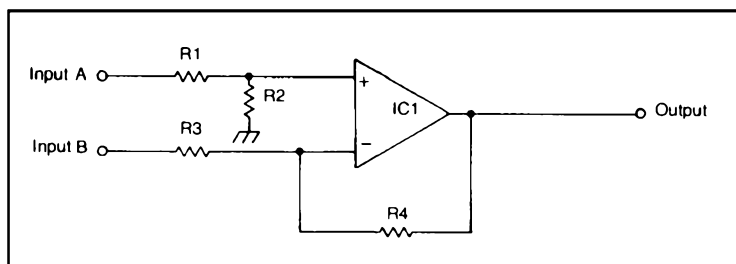


Fig. 4-10. Op amp subtractor.

output of circuit 1 to the input of circuit 2, with very little of the original energy being wasted as heat.

However, if the impedances are not matched (i.e., one circuit has a larger impedance than the other), more of the power supplied by the source will be wasted, with less being available to the second circuit. This can introduce considerable noise and/or distortion to a signal when the impedance mismatch is severe (the impedances differ greatly).

Fortunately, impedance matching is not critical on a precision level. For instance, you could probably feed a 500 ohm impedance output into a 600 ohm impedance input without even being able to

Table 4-9. Parts List for Fig. 4-10.

R1, R3, R4	4.7 k Ω resistor
R2	1 k Ω resistor
IC1	low noise op amp IC

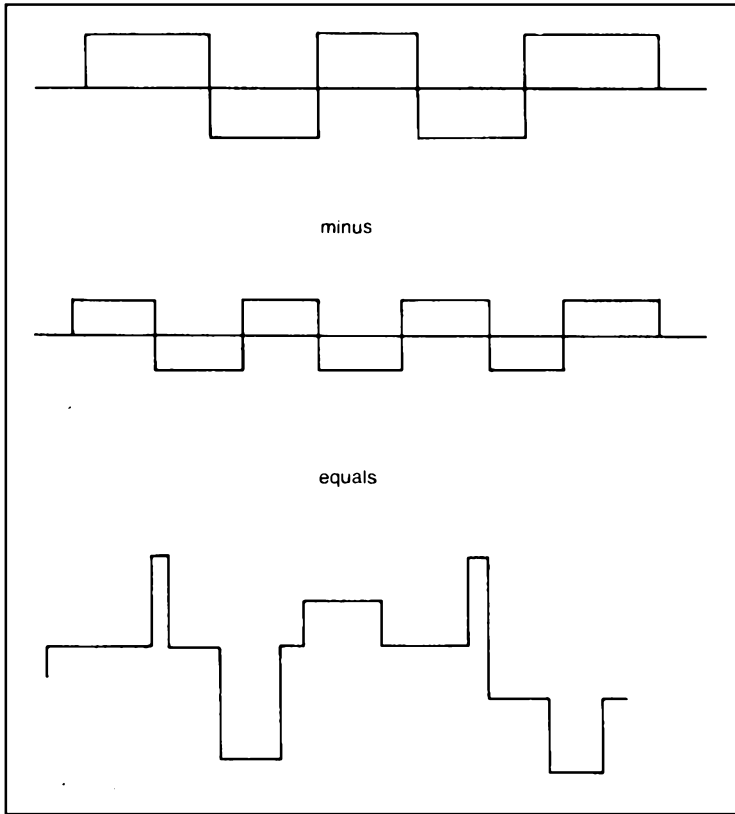


Fig. 4-11. Effect of the subtractor circuit in Fig. 4-10 on a waveforms.

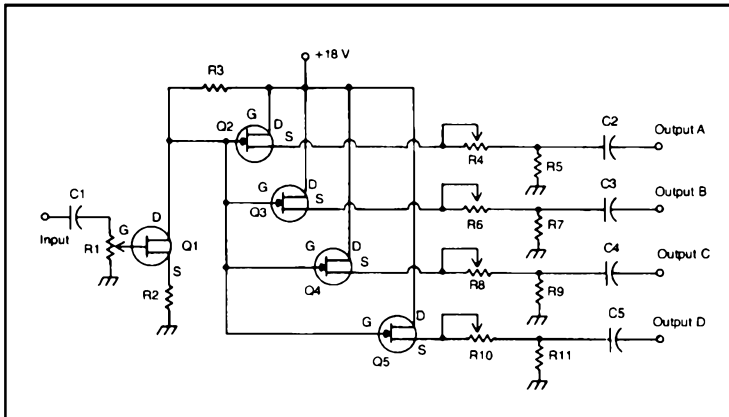


Fig. 4-12. Active signal splitter.

Table 4-10. Parts List for Fig. 4-12.

R1	1 M Ω potentiometer
R2	1 k Ω resistor
R3	10 k Ω resistor
R4, R6, R8, R10	100 k Ω potentiometer
R5, R7, R9, R11	4.7 k Ω resistor
C1, C2, C3, C4, C5	0.1 μ F capacitor
Q1, Q2, Q3, Q4, Q5	FET (2N5457, Radio Shack RS-2028, or similar)

notice any signal degradation. If, however, you try to feed a 10,000 ohm impedance input with a 500 ohm impedance output, you are likely to run into some problems.

The impedances of the circuits in this book should coincide fairly closely unless otherwise noted in the text. These circuits were selected to work together as well as possible.

The sound effects circuits built around the SN76477/88 sound effects generator IC (discussed in Chapter 3) are designed to feed an 8 ohm speaker directly. If you are modifying the signal with other synthesis circuits, the output impedance may be a little low. If this causes problems and unsatisfactory signal quality, you can either use an impedance matching transformer, or a voltage follower (unity gain amplifier) circuit, such as the one shown in Fig. 4-13.

PANNING AND CROSS FADING

Electronic music becomes even more exciting when it is in stereo or quad (or other multiple speaker arrangement). Various sounds can be located at different apparent points in the room, and can even seem to move about.

Panning is the term used to describe the effect of making a sound source appear to move between speakers. Let's assume a stereo (two speaker) set-up. A sound starts out coming entirely from the left speaker, with no signal being fed to the right speaker at all. Of course, the sound will seem to come from the location of the left speaker, as shown in Fig. 4-14.

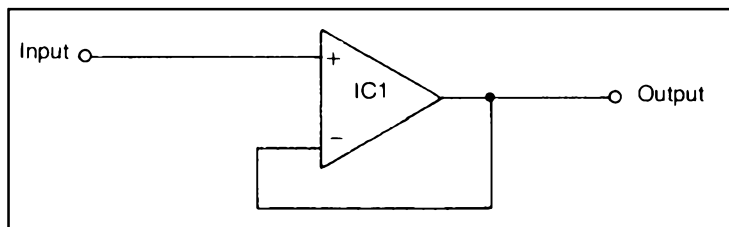


Fig. 4-13. Voltage follower for impedance matching.

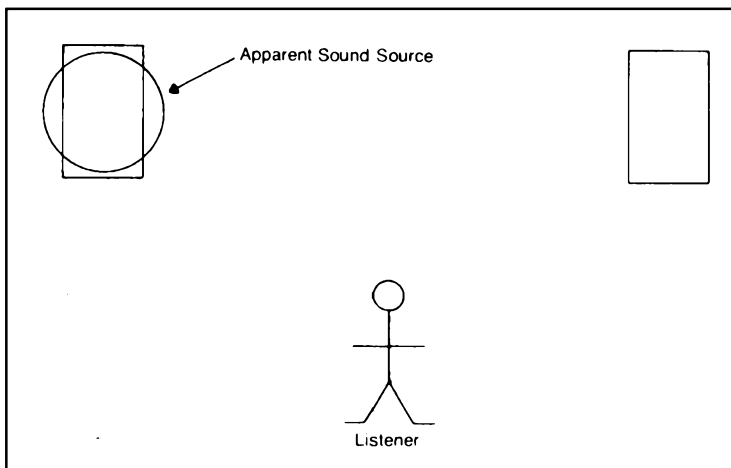


Fig. 4-14. Panning example A.

Now, let's reduce the signal being fed to the left speaker slightly, and feed the subtracted signal to the right speaker. The left speaker is still producing most of the sound, but some is also coming from the right speaker. In this case, the sound will have an apparent source somewhere between the two speakers but to the left of the center point, since the left speaker is being fed a stronger signal than the right. This is illustrated in Fig. 4-15.

Figure 4-16 shows how the apparent location of the sound is affected when both speakers are fed equal level signals. Not sur-

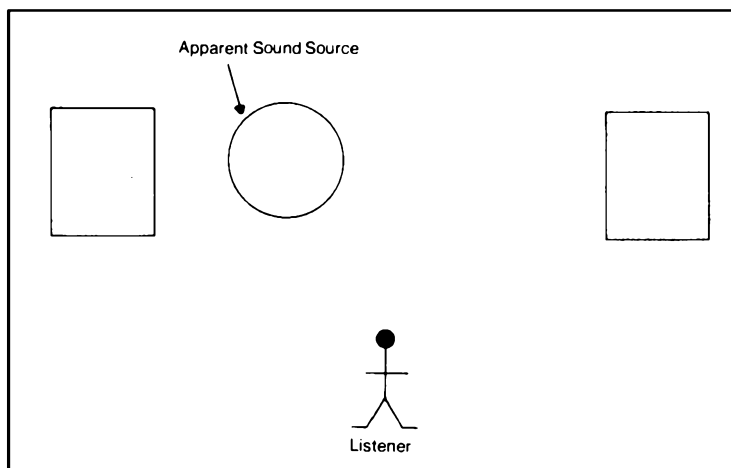


Fig. 4-15. Panning example B.

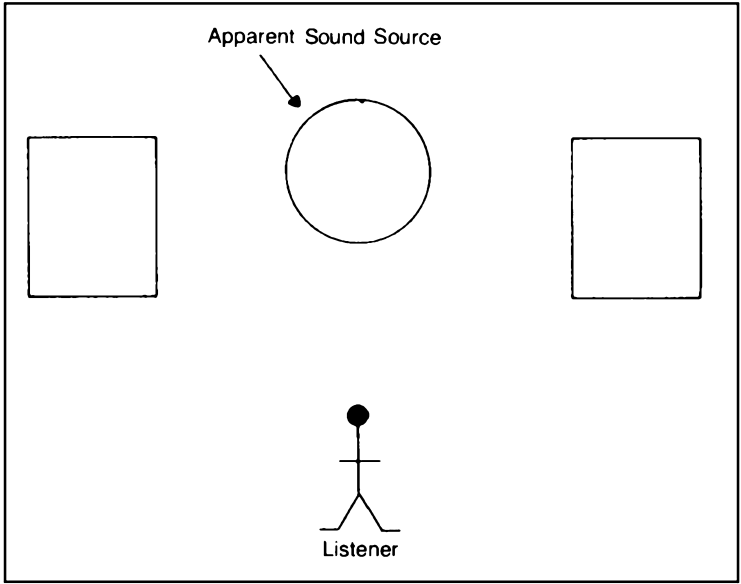


Fig. 4-16. Panning example C

prisingly, the sound appears to come from a point exactly centered between the two speakers.

If all of the signal is fed to the right speaker, and none to the left speaker, the sound will appear to come from the right speaker, as indicated in Fig. 4-17.

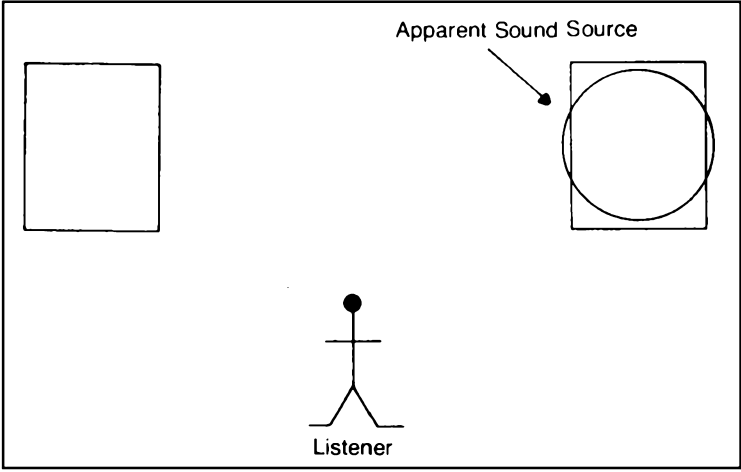


Fig. 4-17. Panning example D.

By controlling the proportion of the signal fed to each speaker, the sound can be illusionarily placed at any desired point between the speakers.

In the four speaker quadraphonic system of Fig. 4-18, the signal can be placed at many additional points throughout the listening area.

In electronic music, moving apparent sound sources can add greatly to the dramatic effect. Many synthesizers have output mixers with *panpots*, or potentiometers that control the relative strength of the signals to be fed to each of the loud speakers. The electronic musician can easily change these control settings and move the apparent sound source while he is performing.

Panning can also be automatically controlled. Figure 4-19 shows a patch for automatic panning. The circle marked lfo is a low frequency oscillator. As the name obviously suggests, this is an oscillator with a very low frequency—usually below the audible range (about 20 Hz). Low frequency oscillators will be described in Chapter 8.

The triangle labeled vca represents a *voltage-controlled amplifier*. This is an amplifier whose gain is set by a control voltage. Voltage controlled amplifiers will be dealt with in Chapter 6.

The inverter (discussed in Chapter 9) is simply a circuit for reversing the polarity of a signal. If a +1.25 volt signal is fed to the

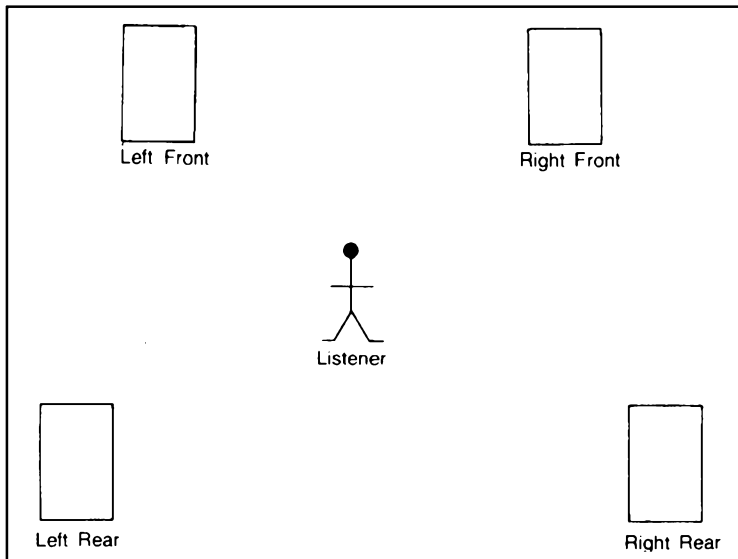


Fig. 4-18. Quadraphonic sound system.

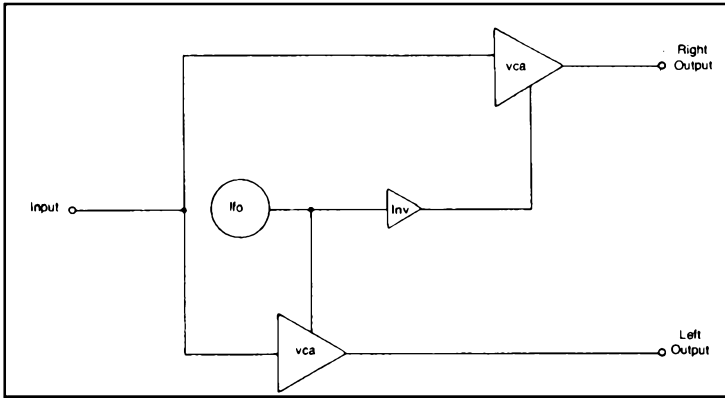


Fig. 4-19. Patch diagram for automatic panning.

input of an inverter, a -1.25 volt signal will appear at the output. Figure 4-20 illustrates the way an inverter can affect an ac waveform.

In the patch shown in Fig. 4-19, let's assume the lfo is generating a very low frequency signal, say about 1 Hz. The apparent sound source will move through the sound field (the space between the speakers) in a pattern determined by the lfo waveform. For example, if the lfo is generating a square wave signal, the sound will ping-pong back and forth between the speakers. For half of each cycle, the signal will come entirely from the right speaker, and for the other half of the cycle, the sound will emerge from the left speaker. The sound will appear to jump back and forth between the speakers bypassing the intermediate points.

If the low frequency oscillator is producing an ascending sawtooth wave, the sound will start at the left speaker, move slowly through the sound field until it is coming entirely from the right speaker, and then it will jump back to the left speaker and start all over again. If you are having trouble visualizing how this works, refer back to the signals shown in Fig. 4-20. Consider the signal labelled Input (the non-inverted signal) as a graph of the volume control setting for the left speaker. The Output (inverted signal) represents the volume control setting for the right speaker. The left speaker starts out at full volume, and the right speaker starts at its minimum. The right signal is gradually turned up, while the left signal is being turned down, until the right speaker is up full and the left speaker is at its minimum setting. Then the controls are instantly set back to their original setting (left—maximum, right—minimum) and the pattern is repeated.

If a sine wave or triangle wave is used in the patch of Fig. 4-19, the sound will appear to swim smoothly back and forth between the speakers.

Some very interesting things can happen if an audio frequency (about 20 Hz) signal is used to control the vcas in this patch. In most cases, the apparent sound source will be the midpoint between the speakers, since it will be moving too fast for the ear to follow. Often a very large apparent sound source will be the result.

In addition, a number of spurious tones not contained in the original signal will be heard. These are called *sidebands*. Sidebands usually give a very complex, somewhat harsh quality to a modulated sound.

This effect is not widely used, and it could wear thin if employed too frequently. However, it can be quite dramatic if tastefully used.

The effect is called *location modulation*, or LM. Its sound is somewhat similar to that of amplitude modulation (AM), which is not surprising, since LM is actually an exotic variation of AM. Amplitude modulation will be examined in some detail in Chapter 6.

In the following section, we are going to look at another common form of modulation used in electronic music.

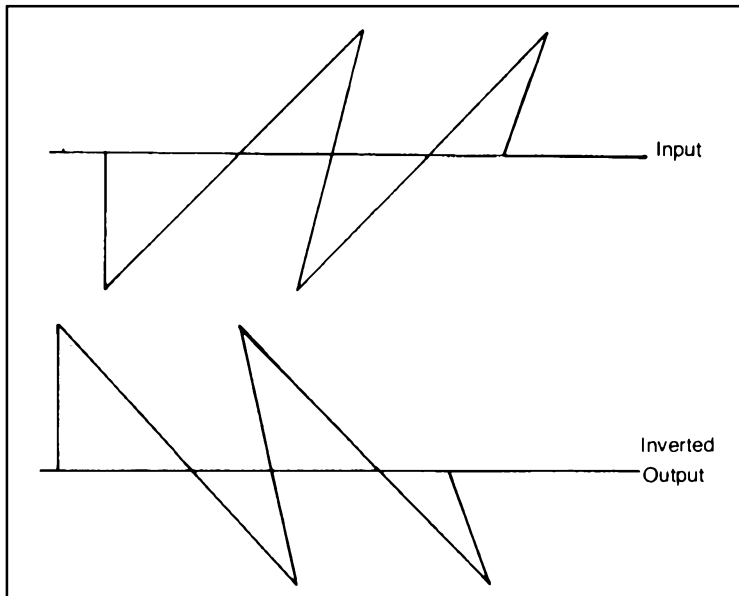


Fig. 4-20. Effect of an inverter on a sawtooth wave.

FREQUENCY MODULATION

If a voltage-controlled oscillator is driven by a periodic ac signal (such as from a second oscillator) in the audible range, the two signals will interact to form a complex new signal. This effect is called frequency modulation, or FM, for short. It is one of the most popular effects in the electronic musician's standard repertoire.

Frequency modulation's most striking effect is the creation of phantom frequencies not present in either of the original signals. These phantom frequencies are called sidebands, and they will probably not be in any harmonic relationship.

The main signal from the vco is called the *carrier*, while the controlling signal is referred to as the *program*. The instantaneous frequency of the carrier will vary in proportion to the instantaneous voltage of the program signal. This is illustrated in Fig. 4-21.

Each sideband is created by adding or subtracting the program frequency (or a multiple thereof) to the carrier frequency. The number of sidebands is determined by the amplitude of the program signal. This number is called the *modulation index*, and it can be calculated by dividing the maximum deviation that the program signal will cause in the carrier frequency (from its nominal, unmodulated value) by the program frequency.

For simplicity, we will work with pure sine waves in the examples. Other waveforms may be used in practice, but the results can get extremely complex because each harmonic in each signal will interact and create its own set of sidebands just as if it was a sine wave by itself.

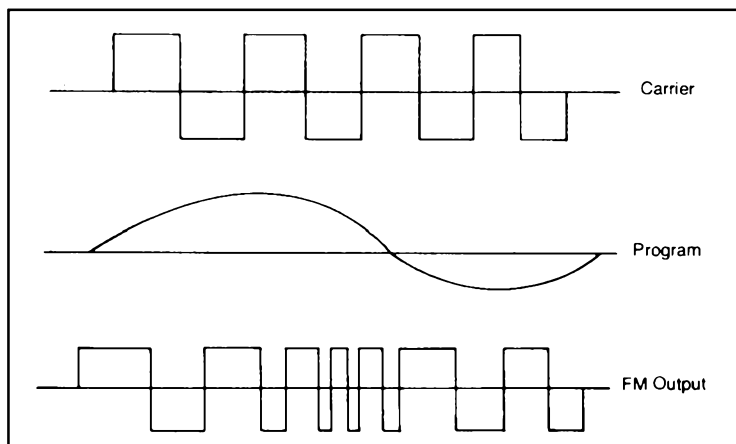


Fig. 4-21. FM waveforms.

Let's assume the carrier signal has a frequency of 1500 Hz, and will be modulated by a 100 Hz program signal. The program signal's amplitude is wide enough to cause the output signal to fluctuate between 1000 Hz and 2000 Hz. This is a maximum frequency deviation of 500 Hz in either direction.

We can calculate the modulation index with the following equation:

$$\text{Modulation Index} = \frac{\text{Maximum Frequency Deviation}}{\text{Program Frequency}}$$

$$\text{or Modulation Index} = \frac{500}{100} = 5$$

This means there will be five sidebands above the nominal carrier frequency and five below it. Each sideband will be removed from the nominal carrier frequency by a factor of the program frequency (1000 Hz).

We can find the upper sidebands simply by adding 100 Hz five times;

1500 Hz	nominal carrier frequency
1600 Hz	first upper sideband
1700 Hz	second upper sideband
1800 Hz	third upper sideband
1900 Hz	fourth upper sideband
2000 Hz	fifth upper sideband

Similarly, the lower sidebands can be found by subtracting the program frequency five times.

1500 Hz	nominal carrier frequency
1400 Hz	first lower sideband
1300 Hz	second lower sideband
1200 Hz	third lower sideband
1100 Hz	fourth lower sideband
1000 Hz	fifth lower sideband

As a second example, let's say we have a 2000 Hz carrier, and a 55 Hz program which causes a peak deviation of 440 Hz (the output signal varies between 1560 Hz and 2440 Hz). First, we calculate the modulation index.

$$\text{Modulation Index} = \frac{440}{55} = 8$$

This gives us a total of 16 sidebands—8 above the nominal carrier frequency, and 8 below. The resulting signal would consist of the following frequencies:

- 1560 Hz
- 1615 Hz
- 1670 Hz
- 1725 Hz
- 1780 Hz
- 1835 Hz
- 1890 Hz
- 1945 Hz
- 2000 Hz nominal carrier frequency
- 2055 Hz
- 2110 Hz
- 2165 Hz
- 2220 Hz
- 2275 Hz
- 2330 Hz
- 2385 Hz
- 2440 Hz

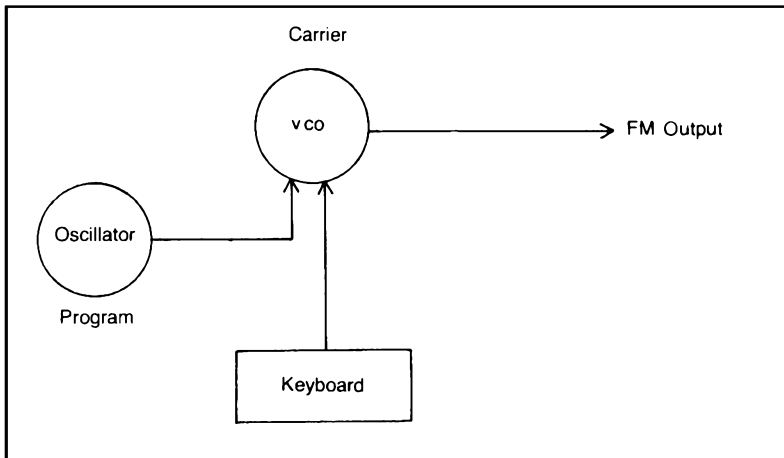


Fig. 4-22. Patch diagram for simple FM.

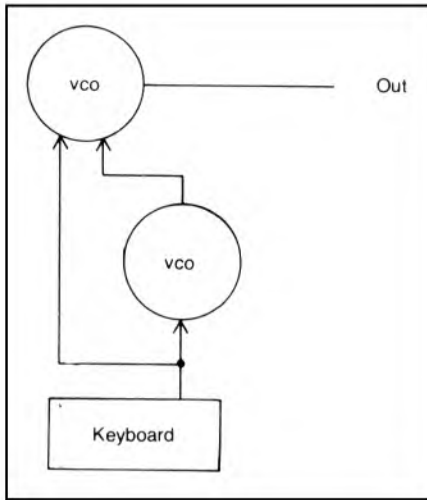


Fig. 4-23. Patch diagram for improved FM.

Another effect of frequency modulation is that as the amount of modulation (number of sidebands) is increased, the strength of the original carrier signal in the output is decreased. This coupled with the nonharmonic relationship of the sidebands can result in a strongly modulated signal losing its definite sense of pitch, especially if complex waveforms are used for the original carrier and program signals.

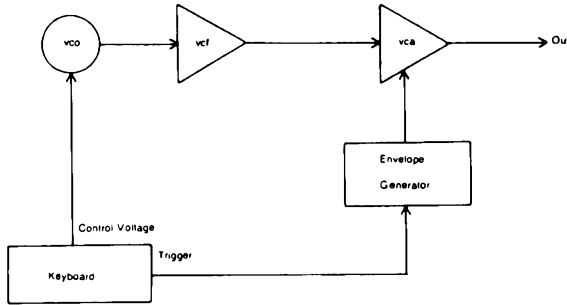
For musical melodic voices, it is a good idea for at least one of the original signals to be a sine wave (or a low-pass filtered triangle wave).

Figure 4-22 shows a simple patch for FM effects. A manually controlled oscillator can be used to modulate the vco. However, this approach leaves something to be desired. Everytime the vco changes frequencies, the tonal quality of the sound will fluctuate somewhat. This may or may not be objectionable.

By using vcOs for both signal sources, as shown in Fig. 4-23, such potential problems may be easily avoided. The program and the carrier will be in a constant relationship with each other, so the tonal quality of the nonharmonic sidebands will be more stable.

Frequency modulation is ideal for synthesizing bells and metallic sounds. If used with restraint and taste, some very satisfying and rich sounds can be generated through FM.

Chapter 5



Filters

So far we have been concentrating on signal sources, but that is only part of electronic sound synthesis. A practical synthesizer system should also include a number of modules that can manipulate or change the signal in some way.

The most common and versatile of the signal manipulation circuit types is the *filter*. A filter is a frequency selective device. It will allow some frequencies to pass through to the output, while blocking (or greatly attenuating) other frequencies. When a complex (harmonic rich) signal is fed into the filter, the structure of the signal will be altered at the output. For example, if a 1500 Hz triangle wave is fed through a filter which blocks all signals above 2000 Hz, the output will be a 1500 Hz sine wave, since all of the harmonics of the original signal (which start at 4500 Hz) will be removed, leaving only the fundamental.

SUBTRACTIVE SYNTHESIS

In Chapter 4, we discussed additive synthesis in which two or more simple signals are combined to create new, complex signals. *Subtractive synthesis* is essentially the opposite of additive synthesis.

In subtractive synthesis, we start out with a complex signal and remove the frequencies that we don't want to create a new, simpler signal. Filters are used to remove frequencies.

Let's say we start out with a 500 Hz square wave. The harmonic content of the signal would be as follows:

500 Hz	fundamental
1500 Hz	third harmonic
2500 Hz	fifth harmonic
3500 Hz	seventh harmonic
4500 Hz	ninth harmonic
5500 Hz	eleventh harmonic
6500 Hz	thirteenth harmonic
7500 Hz	fifteenth harmonic

If we filter out all frequencies between 2000 Hz and 5000 Hz, we will be left with this overtone series:

500 Hz
1500 Hz
5500 Hz
6500 Hz
7500 Hz

This signal will sound quite different than the original square wave. In this example, the tone will have a rather hollow quality.

For a second example, let's again start with the same 500 Hz square wave signal. In this case we will filter out all frequencies below 2000 Hz. Now the signal will consist of the following frequencies:

2500 Hz
3500 Hz
4500 Hz
5500 Hz
6500 Hz
7500 Hz

The resulting signal will appear to have a fundamental frequency of 2500 Hz. Notice that the overtones bear no harmonic relationship to this new fundamental.

Of course there is absolutely no reason why additive and subtractive synthesis techniques can't be combined. To illustrate this, let's add a 600 Hz rectangle wave (1:3 duty cycle) to our 500 Hz

square wave. The rectangle wave has the following harmonic content:

600 Hz	fundamental
1200 Hz	second harmonic
2400 Hz	fourth harmonic
3000 Hz	fifth harmonic
4200 Hz	seventh harmonic
4800 Hz	eighth harmonic
6000 Hz	tenth harmonic
6600 Hz	eleventh harmonic
7800 Hz	thirteenth harmonic
8400 Hz	fourteenth harmonic

The combined additive synthesis signal will now include the following frequencies:

500 Hz
600 Hz
1200 Hz
1500 Hz
2400 Hz
2500 Hz
3000 Hz
3500 Hz
4200 Hz
4500 Hz
4800 Hz
5500 Hz
6000 Hz
6500 Hz
6600 Hz
7500 Hz
7800 Hz
8400 Hz

If we filter out everything below 1000 Hz and everything between 2700 Hz and 5300 Hz, we will be left with the following frequencies:

1200 Hz
1500 Hz
2400 Hz

2500 Hz
5500 Hz
6000 Hz
6500 Hz
6600 Hz
7500 Hz
7800 Hz
8400 Hz

Clearly this complex and unusual signal would be difficult to achieve with just additive synthesis or only subtractive synthesis.

While electronic musicians make use of both synthesis techniques, most modern commercial synthesis equipment tends to emphasize subtractive synthesis over additive synthesis. There are a number of reasons for this, not the least of which is that the equipment required for subtractive synthesis tends to be somewhat less complicated than the equipment needed for extensive additive synthesis.

It is often difficult to get a number of vcos to track (hold the same frequency relationship) each other, especially over a wide range. This can make complex additive synthesis involving more than two or three oscillators tricky and tedious work. Sine wave oscillators seem to be particularly troublesome in this respect.

However, practical voltage-controlled filters can do an acceptable job of tracking most voltage-controlled oscillators, so subtractive synthesis is generally less of a pain in the neck.

FILTER TYPES

Filters are classified in a number of ways. One important distinction is between active and passive filters. An *active filter* requires a power supply, and includes some kind of amplifying device to boost the amplitude of the passed frequencies. Active filters may be built around a transistor, an integrated circuit, or a tube. Op amps are especially popular in modern active filter circuits.

A *passive filter* circuit, on the other hand, is parasitic. All of its operating power is taken from the signal itself. No power supply is provided. A passive filter reduces the amplitude of the entire signal, although the blocked frequencies are attenuated more than the passed frequencies, of course.

Semiconductors and other active components are not used in passive filter circuits. Only resistors, capacitors, and coils are used.

Another common way of classifying filter circuits is by their slope. An ideal filter that is intended to block everything above 1000 Hz, and pass everything below that frequency, should completely block a 1001 Hz signal, and completely pass a 999 Hz signal. The frequency response of such an ideal filter is illustrated in Fig. 5-1.

Such an ideal circuit is impossible to achieve. In a practical filter circuit, there will be a range of frequencies that are partially blocked. A more realistic frequency response chart for a filter is shown in Fig. 5-2.

Obviously, the steeper the slope, the better the filter. A filter's cutoff slope is measured in dB per octave. Let's say our 1000 Hz filter is a passive type circuit that would give at a slope of 3 dB per octave. In other words, 2000 Hz would be attenuated by 3 dB, 4000 Hz would be reduced by 6 dB, 8000 Hz would be attenuated by 9 dB, and so forth.

Some active filters can have slopes as steep as 24 dB per octave. In some cases, steeper slopes may be achieved by using multiple filters. This is called cascading.

The most important way of classifying filters is to define what frequencies it passes. The four basic types are low-pass filters, high-pass filters, band-pass filters, and band-reject (or notch) filters.

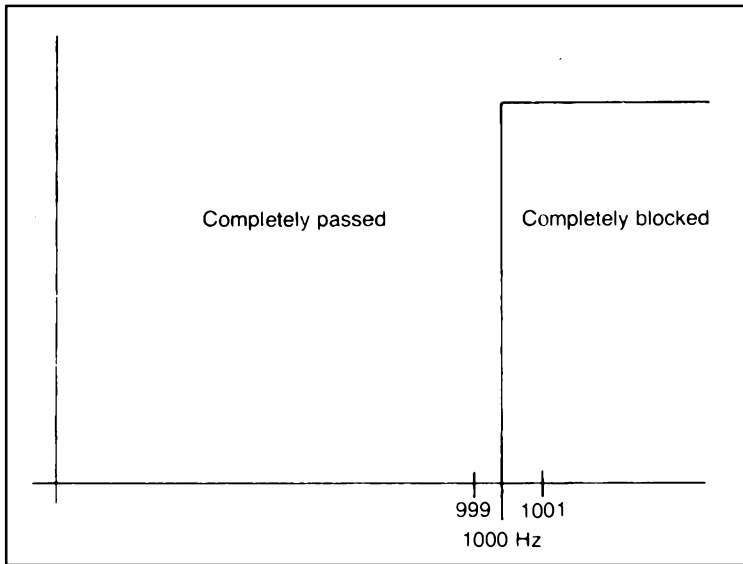


Fig. 5-1. Frequency response graph of an ideal low-pass filter.

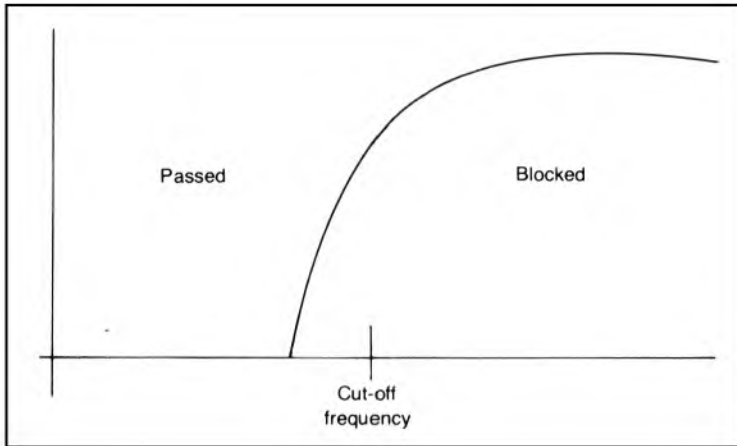


Fig. 5-2. Frequency response graph of a practical low-pass filter.

Passive Filters

Passive filters can come in handy when you need a quick and dirty filter at minimal cost, but they generally aren't good enough for electronic music work. They will be discussed in this chapter anyway, however, because their action is clearer than that of active filters. The passive circuits represent a good introduction to the theory of filters.

Low-Pass Filters. The frequency response graphs shown in Fig. 5-1 and Fig. 5-2 are for low-pass filters. The name is really self-explanatory. Low frequencies are passed by this type of circuit while the high frequencies are blocked.

A simple passive low-pass filter circuit is shown in Fig. 5-3. It consists of just two components—a capacitor and a resistor. If you are familiar with basic electronics theory, you should know that a capacitor offers more impedance to low frequencies than to high frequencies. To a very low frequency or dc (0 Hz), an ideal capacitor behaves like an open circuit, while at high frequencies, it behaves like a short circuit, or a very small resistance. In the circuit of Fig. 5-3, the capacitor shunts high frequencies to ground, while forcing the lower frequency signals to pass on through the output.

The cutoff frequency for this simple low-pass filter circuit is determined by the combined values of the resistor and capacitor. The required values for a specific cutoff frequency can be found with the following formula:

$$R = \frac{159,000}{FC}$$

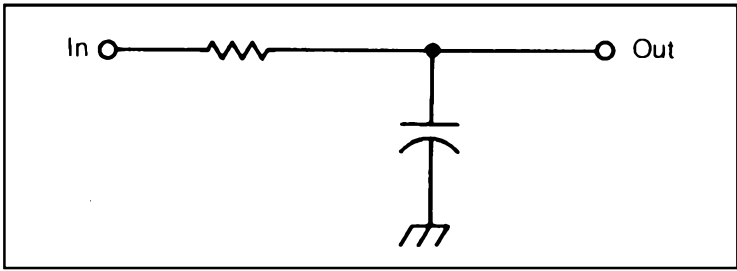


Fig. 5-3. Passive low-pass filter.

R is the resistance in ohms, F is the frequency in hertz, and C is the capacitance in microfarads.

In designing such a filter the best approach is to arbitrarily select a value for C and calculate R. Resistors tend to be available in more intermediate values and readily combine to make up oddball values much easier than capacitors. If necessary, a trimpot could be used. However, such precision is rarely practical in this circuit. The cutoff slope is so gradual (3 dB per octave) that there will be little audible difference between a filter with a 1000 Hz cutoff and one with a 1200 Hz cutoff. There is plenty of room for error.

Let's try a few examples. First, let's design a passive low-pass filter with a nominal cutoff frequency of 1000 Hz. We'll use 0.1 μF as a handy standard value for the capacitor.

$$R = \frac{159,000}{FC} = \frac{159,000}{(1000)(0.1)} = 1590 \text{ ohms.}$$

A 1.5k Ω resistor would certainly be close enough. The exact cutoff frequency (assuming 0 percent tolerance for the components) would be 1060 Hz.

There are many different possible combinations of components for any given cutoff frequency. For instance, if we started the previous problem with a 0.022 μF capacitor, we'd end up with a resistance of 7227 ohms.

$$R = \frac{159,000}{FC} = \frac{159,000}{(1000)(0.022)} = 7227$$

A 6.8 k Ω resistor could be used.

Now let's try a second example. We need a low-pass filter with

a cutoff frequency of 2250 Hz, and we'll again start out with a 0.1 μF capacitor:

$$R = \frac{159,000}{(2250)(0.1)} = 706 \text{ ohms}$$

This result is somewhat of an oddball value. We could use a 680 ohm resistor, of course, but it's a good idea to aim for something a little higher. When this kind of thing comes up, you can always try to recalculate R with a new value of C. We'll try 0.001 μF this time:

$$R = \frac{159,000}{(2250)(0.001)} = 70,666 \text{ ohms}$$

Once again we'd have to round down to 68 k Ω , but at least we'll have a higher resistance.

Passive filter circuits are usually fixed frequency devices, but you could build a low-pass filter with a manually variable cutoff frequency simply by adding a potentiometer to the basic circuit, as shown in Fig. 5-4.

High-Pass Filters. The opposite of a low-pass filter is, not surprisingly, a high-pass filter. Where a low-pass filter passes low frequencies and blocks high frequencies, a high-pass filter passes high frequencies and blocks low frequencies. A frequency response graph for a high-pass filter is shown in Fig. 5-5. Notice how it is a mirror image of the graph for a low-pass filter (Fig. 4-2).

The circuit for a passive high-pass filter is also the mirror image of its low-pass equivalent. The circuit is illustrated in Fig.

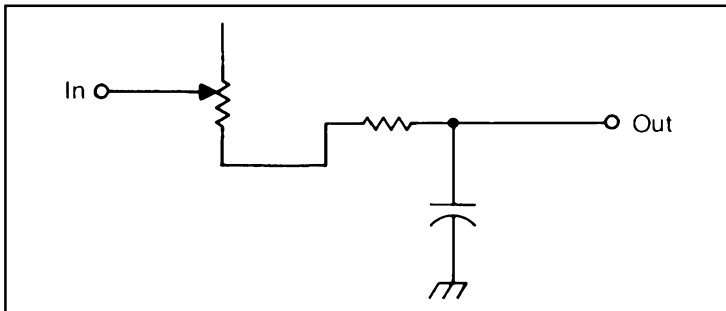


Fig. 5-4. Manually variable passive low-pass filter.

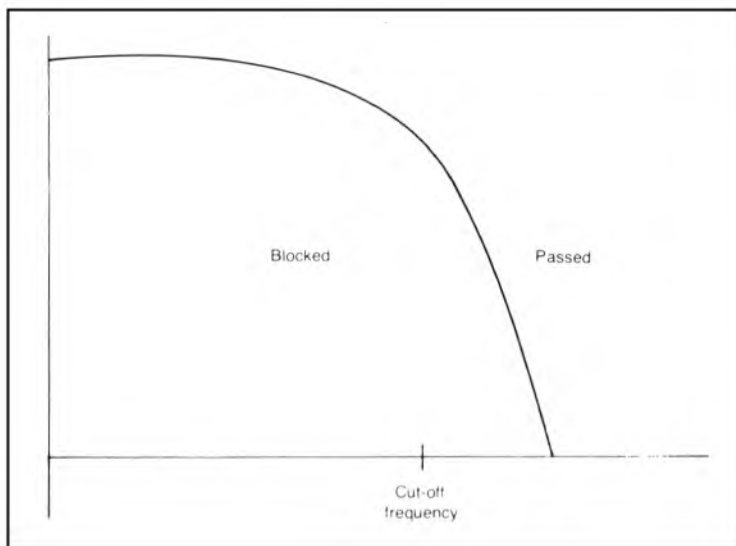


Fig. 5-5. Frequency response graph of a high-pass filter.

5-6. It is just like the low-pass filter circuit of Fig. 5-3 except that the resistor and the capacitor have changed position.

High-pass filters are not as commonly used in electronic music work as low-pass filters are. A low-pass filter will remove some of the harmonics, but the fundamental (and the apparent pitch of the filtered sound) will remain unchanged. A high-pass filter, on the other hand, will delete the original fundamental frequency. One of the original signal's harmonics will become the new apparent fundamental, and it is not likely that the remaining overtones will have a harmonic relationship to this new fundamental.

As an example, let's assume we have a 500 Hz square wave, and a low-pass filter and a high-pass filter. Each filter will have a

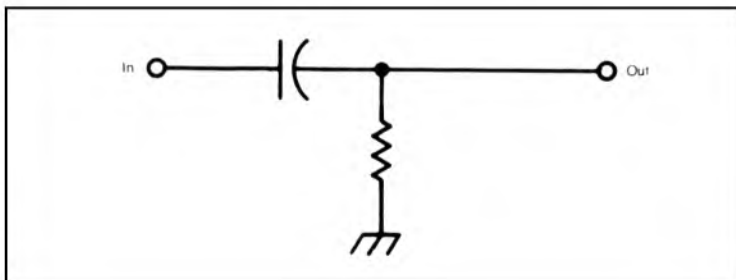


Fig. 5-6. Passive high-pass filter.

cutoff frequency of 3000 Hz. For simplicity, we will assume the filters are ideal and have infinitely steep slopes.

The original signal will consist of the following frequencies:

500 Hz	fundamental
1500 Hz	third harmonic
2500 Hz	fifth harmonic
3500 Hz	seventh harmonic
4500 Hz	ninth harmonic
5500 Hz	eleventh harmonic
6500 Hz	thirteenth harmonic
7500 Hz	fifteenth harmonic

The output of the low-pass filter will not be too far removed from the original signal. It will contain the following three frequency components:

500 Hz	fundamental
1500 Hz	third harmonic
2500 Hz	fifth harmonic

The output of a high-pass filter, on the other hand, will be quite different. In the case of our example, the output signal would include the following frequencies:

3500 Hz	apparent fundamental
4500 Hz	nonharmonic overtone
5500 Hz	nonharmonic overtone
6500 Hz	nonharmonic overtone
7500 Hz	nonharmonic overtone

Obviously, the output of the high-pass filter will sound less like the original signal than will the output of the low-pass filter.

Band-Pass Filters. So far we have been dealing with filters that remove all frequencies either above or below a specific point. But what if we want to filter out the lowest and the highest frequencies in a complex signal and keep some of the middle frequencies. In this case, a band-pass filter is needed.

A *band-pass filter* can be created by connecting a low-pass filter and a high-pass filter in series, as shown in Fig. 5-7. The cutoff frequency of the low-pass filter should be higher than that of the high-pass filter. Only those frequencies that are passed by both

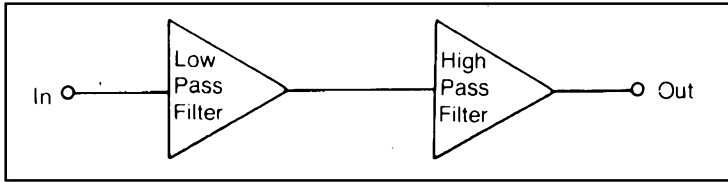


Fig. 5-7. Combining a low-pass and a high-pass filter in series to create a band-pass filter.

filters will reach the output. The effect of a band-pass filter is illustrated in the frequency response graph of Fig. 5-8.

A band-pass filter can be defined by giving both the upper and the lower cutoff frequency points, but the center frequency and the Q are the parameters most commonly used.

The center frequency, obviously, is the frequency that is in the middle of the passed band. It can be found by taking the average of the upper and lower cutoff frequencies. For example, if the high-pass filter has a cutoff frequency of 1000 Hz, and the low-pass filter has a cutoff frequency of 3500 Hz, the center frequency would be 2250 Hz. The passed band would be 2500 Hz wide. The bandwidth is found simply by subtracting the lower cutoff frequency from the upper cutoff frequency.

Q is a mathematical factor that determines the bandwidth. Practical filters have Q values ranging from less than one on up into the hundreds.

Band-pass filters can be made by cascading low-pass and high-pass filters, as illustrated in Figure 5-7, but it is also possible to build a single circuit that will produce exactly the same results.

For a passive band-pass filter, you can simply add a coil across

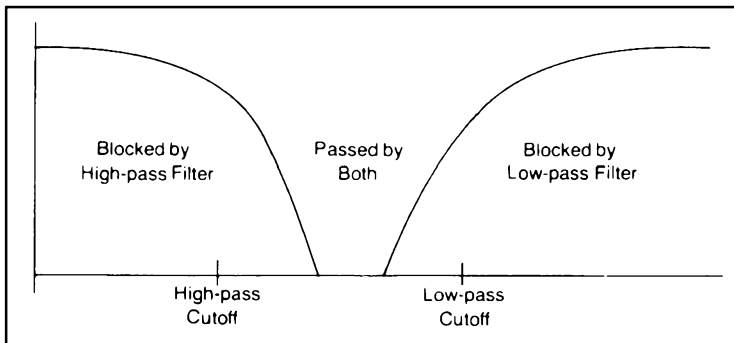


Fig. 5-8. Frequency response graph of a band-pass filter.

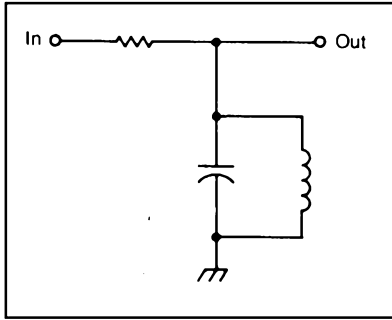


Fig. 5-9. Passive band-pass filter.

the capacitor of a low-pass filter. This circuit is shown in Fig. 5-9. The bandwidth of this circuit is determined by the resistance and capacitance, using the same formula used to determine the cutoff frequency of a low-pass filter.

$$BW = \frac{159,000}{RC}$$

The center frequency, on the other hand, is set by the resonant frequency of the coil/capacitor combination. The formula is as follows:

$$F = \frac{159}{\sqrt{LC}}$$

As an example, let's say we have a passive band-pass filter consisting of a 2200 kΩ resistor, a 0.05 μF capacitor, and a 50 mH coil. First off, we calculate the bandwidth:

$$BW = \frac{159,000}{RC} = \frac{159,000}{(2200)(.05)} = \frac{159,000}{110} = 1445 \text{ Hz}$$

The bandwidth is approximately 1500 Hz wide. Next we calculate the center frequency:

$$F = \frac{159}{\sqrt{LC}} = \frac{159}{\sqrt{(.05)(.05)}} = \frac{159}{\sqrt{.0025}} = \frac{159}{.05} = 3180 \text{ Hz}$$

The center frequency is 3180 Hz, and the bandwidth is 1445 Hz. This means that all frequencies within 722.5 Hz of 3180 Hz will be passed by this filter. In other words, this combination of components will filter out all frequencies except those between 2457.5 Hz and 3902.5 Hz.

Of course, in practical applications, you will know what center frequency and bandwidth you need and will want to choose components that will work out to those parameters.

Let's work a sample problem from this approach. We'll assume we need a bandpass filter that will leave us with the frequencies from 1200 Hz to 2700 Hz. This is a bandwidth of 1500 Hz (2700 - 1200). The center frequency is the average of the two extremes, or 1950 Hz in this case.

First we'll work out the low-pass filter section to give us the desired band-width. We'll try it with a 0.022 μF capacitor:

$$R = \frac{159,000}{(\text{BW})(C)} = \frac{159,000}{(1500)(.022)} = \frac{159,000}{33} = 4818.1818 \text{ ohms} \approx 4700 \text{ ohms}$$

Now, we can calculate the necessary inductance to give us a center frequency of 1950 Hz. But first we must rearrange the formula to solve for L:

$$F = \frac{159}{\sqrt{LC}} \text{ so } L = \frac{1}{C} \left(\frac{159}{F} \right)^2$$

So, plugging in the values for our sample problem, we find the following:

$$L = \frac{1}{.022} \left(\frac{159}{1950} \right)^2 = 45.4545 (.0815)^2 = 45.4545 \times 0.00665 = 0.3022 \text{ H} \approx 300 \text{ mH}$$

For a second example, let's design a band-pass filter that will pass all frequencies from 700 Hz to 3600 Hz. This works out to a bandwidth of 2900 Hz, and a center frequency of 2150 Hz. We'll use a 0.1 μF capacitor. Solving for R first we get:

$$R = \frac{159,000}{(\text{BW})(C)} = \frac{159,000}{(2900)(.1)} = \frac{159,000}{290} = 548.28 \text{ ohms}$$

The nearest standard value is 560 ohms.

Next we calculate the value for the coil:

$$L = \frac{1}{C} \left(\frac{159}{F} \right)^2 = \frac{1}{.1} \left(\frac{159}{2150} \right)^2 = 10(.74)^2 = 10 (.00547)$$

$$= .0547 \text{ H} \cong 50 \text{ mH}$$

Band-pass filters are extensively used in electronic music. Low-pass filters are probably the most common type, but band-pass filters certainly run a close second. If they have a tight enough Q they can be used to isolate any harmonic, or group of harmonics. A band-pass filter can even impart a strong sense of pitch on a complex signal like white noise.

To better illustrate the functioning of a band-pass filter, let's look at how the 2150 Hz center frequency/2900 Hz bandwidth band-pass filter we've just designed would affect a 400 Hz square wave. For simplicity we will assume the filter is an ideal one with an infinitely steep cutoff slope, even though a passive filter certainly doesn't come anywhere close to this ideal.

The original square wave signal would have the following harmonic content:

400 Hz	fundamental
1200 Hz	third harmonic
2000 Hz	fifth harmonic
2800 Hz	seventh harmonic
3600 Hz	ninth harmonic
4400 Hz	eleventh harmonic
5200 Hz	thirteenth harmonic
6000 Hz	fifteenth harmonic

The output of the filter would consist of the following frequency components:

1200 Hz	new apparent fundamental
2000 Hz	nonharmonic overtone
2800 Hz	nonharmonic overtone
3600 Hz	third harmonic

A band-pass filter could be used in place of a low-pass filter by adjusting the bandwidth and center frequency so that the lower cutoff frequency is below the audible range (about 20 Hz). Similarly, a band-pass filter can simulate the action of a high-pass filter if the upper cutoff frequency is above the audible range (about 20,000 Hz, or 20 kHz).

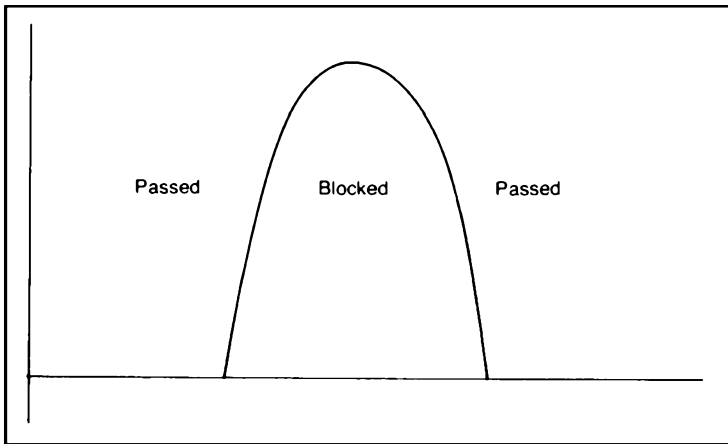


Fig. 5-10. Frequency response graph of a band-reject filter.

Band-Reject Filters. Just as the opposite of a low-pass filter is a high-pass filter, the opposite of a band-pass filter is a band-reject filter. Band-reject filters are sometimes called notch filters.

A *band-reject filter*, as the name clearly suggests, blocks a specific band of frequencies, while passing all those above or below that band. A typical frequency response chart for a band-reject filter is shown in Fig. 5-10.

A passive band-reject filter can be made by reversing the components of a band-pass filter. The circuit is illustrated in Fig. 5-11. The equations are the same as for a band-pass filter.

Actually, band-reject filters are not widely used in electronic music. They are most commonly employed for noise reduction. A very tight Q band-reject filter centered around 60 Hz (or 120 Hz) can be used to reduce ac hum from power supply transformers, or picked up by interconnecting wires.

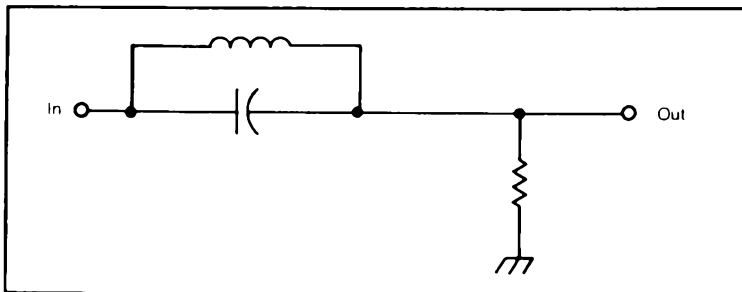


Fig. 5-11. Passive band-reject filter.

Impedance Matching In Passive Filters. In an earlier chapter, we mentioned that best results can be achieved when the interconnected input and output impedances of various modules match fairly closely.

In most cases, the impedance through a passive filter won't be all that important, since this kind of simple circuit is severely limited in quality anyway. However, you may want to even out the impedances somewhat. The resistor is the key component here. For the best impedance match, the value of the resistor should be equal to the square root of the product of the input (source) and output (load) impedances.

For instance, let's assume we are feeding the input of the filter with an oscillator which has an output impedance of 2000 ohms. The output of the filter is being fed into an 8 ohm speaker. We could then calculate the ideal value of the resistor as follows:

$$R = \sqrt{R_s R_L} = \sqrt{(2000)(8)} = \sqrt{16000} = 126.49111 \text{ ohms}$$

A 120 ohm resistor would be close enough.

If you calculate the resistance in this manner, you will then have to calculate the capacitance. The formula given earlier can easily be reworked into the needed form:

$$C = \frac{159,000}{FR}$$

So, if the filter in the previous example is to have a 1000 Hz cutoff, we can find the capacitance to be equal to:

$$C = \frac{159,000}{(1000)(120)} = 1.325 \mu\text{F}$$

A 1 μF or a 1.5 μF capacitor could be used. The problem with this method is that there is a greater jump between standard capacitor values than between standard resistor values, so rounding off to the nearest available value may introduce a greater degree of error from the nominal cutoff frequency.

Active Filters

Earlier in this chapter, the disadvantages of passive filters were discussed. For practical electronic music work, active filter circuits are generally required.

Most modern active filter circuits are designed around op amp

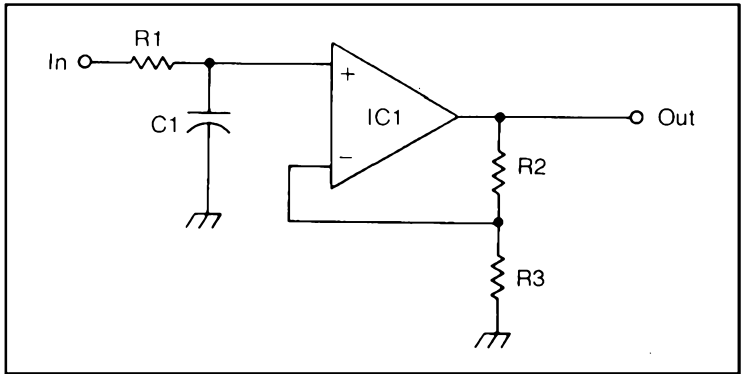


Fig. 5-12. Active, first order, low-pass filter.

ICs. Low cost devices (like the popular 741) may be used in non-critical applications, but the serious electronic musician will usually insist on low noise units. Any semiconductor device will generate some amount of noise (in the undesirable sense). 741s can generate an unacceptable amount of noise especially when high levels of amplification are used.

Low-Pass Filters. A relatively simple active low-pass filter circuit is shown in Fig. 5-12. Of course, the component values used will change, depending on the desired cutoff frequency. The parts lists for three typical cutoffs are listed in Table 5-1, Table 5-2, and Table 5-3.

Resistors R2 and R3 set the gain of the circuit. For the values shown, the passed frequencies will be amplified by a factor of two.

R1 and C1 are the frequency determining components in this circuit. The formula is straightforward:

$$F = \frac{1}{2\pi RC}$$

This filter circuit has a relatively shallow slope.

A better active low-pass filter circuit is shown in Fig. 5-13.

Table 5-1. Parts List for Fig. 5-12 at a Cutoff Frequency of 500 Hz.

R1	6.8 kΩ resistor (see text)
R2	10 kΩ resistor
R3	4.7 kΩ resistor
C1	0.047 μF capacitor (see text)
IC1	low noise op amp IC

Table 5-2. Parts List for Fig. 5-12 at a Cutoff Frequency of 1000 Hz.

R1	3.3 kΩ resistor (see text)
R2	10 kΩ resistor
R3	4.7 kΩ resistor
C1	0.047 μF capacitor (see text)
IC1	low noise op amp IC

Table 5-3. Parts List for Fig. 5-12 at a Cutoff Frequency of 2000 Hz.

R1	8.2 kΩ resistor (see text)
R2	10 kΩ resistor
R3	4.7 kΩ resistor
C1	0.01 μF capacitor (see text)
IC1	low noise op amp IC

This circuit is called a voltage-controlled-voltage-source, or vcvs filter. It should not be confused with the voltage-controlled filter (vcf) that is discussed later in this chapter.

In this circuit, the frequency is dependent on four components—R1, R2, C1, and C2. Generally, R1 will be equal to R2, and C2 should have a value twice that of C1. The cutoff frequency can be calculated with the following formula:

$$F = \frac{1}{2\pi \sqrt{2R1 \times 3C1}}$$

Parts lists for three typical cutoff frequencies are given in Table 5-4, Table 5-5, and Table 5-6.

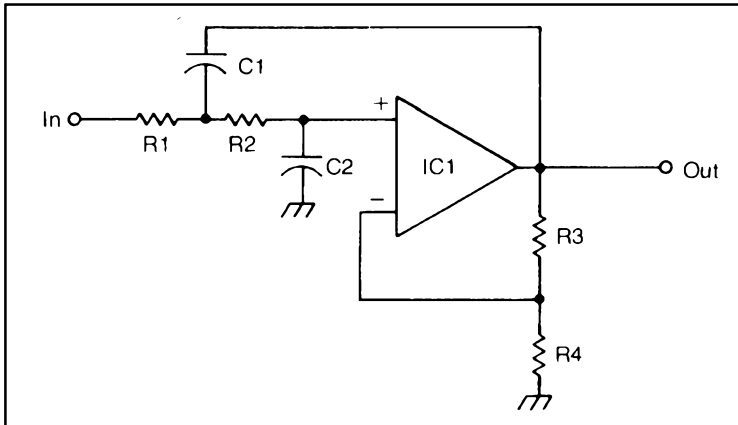


Fig. 5-13. Active, second order, vcvs low-pass filter.

Table 5-4. Parts List for Fig. 5-13 at a Cutoff Frequency of 500 Hz.

R1, R2	1.5 k Ω resistor (see text)
R3	10 k Ω resistor
R4	4.7 k Ω resistor
C1	0.022 μ F capacitor (see text)
C2	0.01 μ F capacitor (see text)
IC1	low noise op amp IC

Table 5-5. Parts List for Fig. 5-13 at a Cutoff Frequency of 1000 Hz.

R1, R2	470 Ω resistor (see text)
R3	10 k Ω resistor
R4	4.7 k Ω resistor
C1	0.022 μ F capacitor (see text)
C2	0.01 μ F capacitor (see text)
IC1	low noise op amp IC

Table 5-6. Parts List for Fig. 5-13 at a Cutoff Frequency of 2000 Hz.

R1, R2	1 k Ω resistor (see text)
R3	10 k Ω resistor
R4	4.7 k Ω resistor
C1	0.0022 μ F capacitor (see text)
C2	0.001 μ F capacitor (see text)
IC1	low noise op amp IC

This filter is called a *second order filter* because its slope is twice as steep as the basic filter circuit that was illustrated in Fig. 5-12.

Filters with steeper slopes can be created by cascading lower-order filter circuits.

High-Pass Filters. As with passive filters, active high-pass filters can be created by swapping the frequency determining resistors and capacitors of a comparable low-pass filter circuit. The high-pass equivalent of the low-pass circuit shown in Fig. 5-12, is illustrated in Fig. 5-14. Figure 5-15 shows a vcvs high-pass filter. The equations are identical to those given for the low-pass filter in the previous section.

Band-Pass Filters. Figure 5-16 shows the schematic for an active band-pass filter. Usually, in working with this circuit, a designer will start out by defining the desired center frequency, Q , and the amplifier gain. Next, a capacitor value is selected—typically in the range of 0.005 μ F to 0.05 μ F. Other values may also be used, but this limited range gives a good starting point. The two capacitors (C1 and C2) should have equal values.

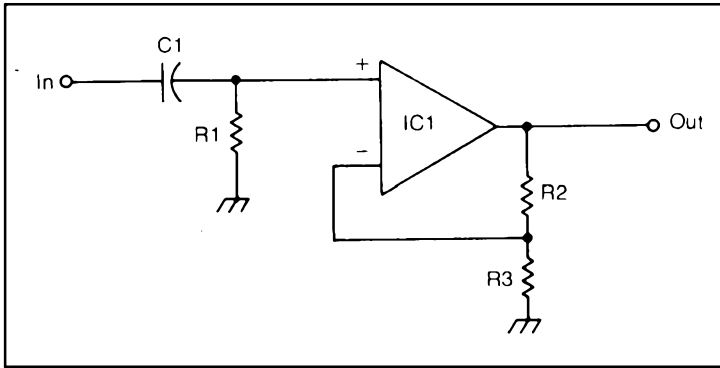


Fig. 5-14. Active, first order, high-pass filter.

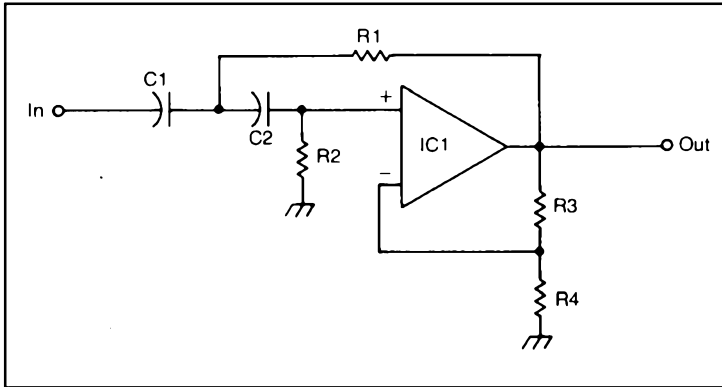


Fig. 5-15. Active, second order, vcvs high-pass filter.

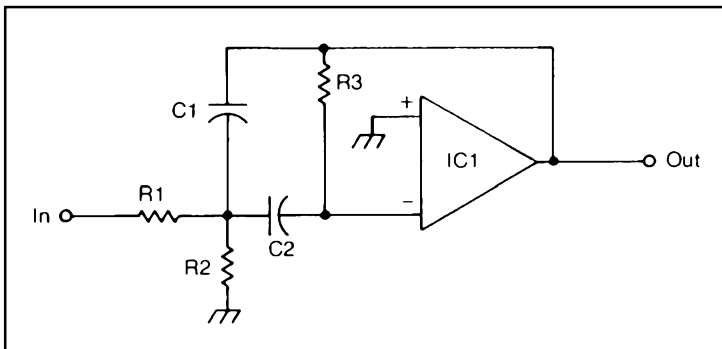


Fig. 5-16. Active band-pass filter.

Table 5-7. Parts List for Fig. 5-16 at a Center Frequency of 500 Hz.

R1	120 kΩ resistor
R2	330 kΩ resistor
R3	3.3 kΩ resistor
C1, C2	0.01 μF capacitor
IC1	low noise op amp IC

The resistance values are closely related to each other and the values mentioned above. Resistor R1, for example, may be calculated with the following formula:

$$R1 = \frac{Q}{2\pi GCF}$$

where G is the amplifier gain, C is the value of C1, and F is the center frequency. Q, of course, represents the Q of the filter. 2π is approximately equal to 6.28.

The value of R2 is determined by R1 and the amplifier gain with the following formula:

$$R2 = 2R1G$$

The formulas for calculating the value of resistor R3 is as follows:

$$R3 = \frac{Q}{2\pi CF(2Q^2 - G)}$$

Some typical component values for center frequencies of 500 Hz, 1000 Hz, and 2000 Hz are given in Table 5-7 through Table 5-9. In calculating these values, I assumed a Q of 5 and a gain (G) of just over 1.25. The values were rounded off for convenience.

For tighter Qs (narrower band-widths) two or more of these filters could be cascaded, or connected in series, as illustrated in Fig. 5-17. The components in each filter section should have identical values. That is, R1sh

Table 5-8. Parts List for Fig. 5-16 at a Center Frequency of 1000 Hz.

R1	62 kΩ resistor
R2	180 kΩ resistor
R3	1.5 kΩ resistor
C1, C2	0.01 μF capacitor
IC1	low noise op amp IC

Table 5-9. Parts List for Fig. 5-16 at a Center Frequency of 2000 Hz.

R1	33 kΩ resistor
R2	91 kΩ resistor
R3	820 Ω resistor
C1, C2	0.01 μF capacitor
IC1	low noise op amp IC

Band-Reject Filters. A band-reject or notch filter circuit is shown in Fig. 5-18. The calculations for this circuit are somewhat more complex than the other circuits discussed in this chapter. This is not particularly unfortunate, since band-reject filters are not very widely used in electronic music. I have simplified the formulas as much as possible here.

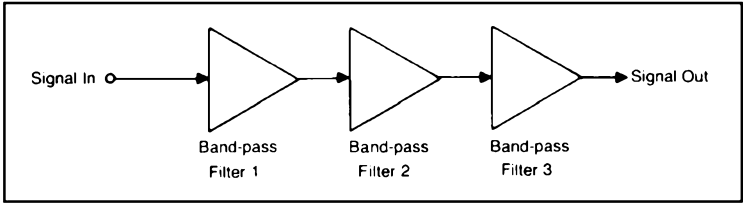


Fig. 5-17. Cascading band-pass filters to achieve a tighter Q.

First, the capacitors are calculated. C1 and C2 should be of equal value. The formula for determining this value is as follows:

$$C1 = \frac{1}{2000000\pi (F/20\pi)}$$

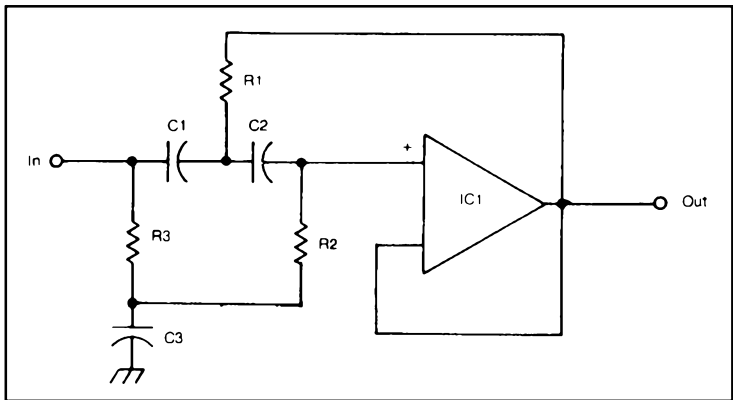


Fig. 5-18. Active band-reject filter.

Table 5-10. Parts List for Fig. 5-18 at a Center Frequency of 60 Hz.

R1	91 Ω resistor
R2	10 k Ω resistor
R3	100 Ω resistor
C1, C2	0.15 μ F capacitor
C3	0.3 μ F capacitor
IC1	low noise op amp IC

where C is the capacitance in farads (multiply by 1,000,000 to get the value in microfarads (μ F)), F is the center frequency in hertz, and π is approximately equal to 3.14.

The value of capacitor C3 is very easy to calculate. It is simply double the value of C1. That is $C3 = 2C1$.

Next we need to calculate the values of the three resistors. The formula for R1 is as follows:

$$R1 = \frac{1000(F/20\pi)}{2Q + (1/2Q)}$$

R2 is found by the following formula:

$$R2 = \frac{1000(F/20\pi)}{(1/2Q)}$$

Finally, we can calculate the value of R3:

$$R3 = \frac{1000(F/20\pi)}{2Q}$$

For all of these equations the amplifier gain is assumed to be 1, or unity. Some typical values for this circuit are listed in Table 5-10 through Table 5-12. A Q of 5 is assumed in each case.

FIXED FILTER BANKS

A number of band-pass filters and manually controlled am-

Table 5-11. Parts List for Fig. 5-18 at a Center Frequency of 120 Hz.

R1	180 Ω resistor
R2	18 k Ω resistor
R3	180 Ω resistor
C1, C2	0.082 μ F capacitor
C3	0.15 μ F capacitor
IC1	low noise op amp IC

Table 5-12. Parts List for Fig. 5-18 at a Center Frequency of 1000 Hz.

R1	1.5 k Ω resistor
R2	150 k Ω resistor
R3	1.5 k Ω resistor
C1, C2	0.001 μ F capacitor
C3	0.0022 μ F capacitor
IC1	low noise op amp IC

plifiers can be combined into a fixed filter bank, or equalizer. The pass bands of the filters should overlap somewhat, as illustrated in Fig. 5-19.

One possible use for such a filter bank would be to even out (or equalize) problem frequencies in your sound systems response. If you are recording, you should not, however, compensate for problems created by the contours of the room. These concepts will be dealt with in more detail in Chapter 6.

A typical filter bank will consist of band-pass filters spaced an octave apart. Some inexpensive systems use fewer bands (each being wider, of course), and some deluxe systems break up the audio range into 1/3 octave bands. More bands give you greater control, but they increase the cost and complexity of the system. In most cases, octave wide bands are an acceptable compromise.

Assuming you want to cover the entire audio range (20 Hz to 20,000 Hz), you will need 10 filters to have each octave individually controllable. The octaves break down as follows:

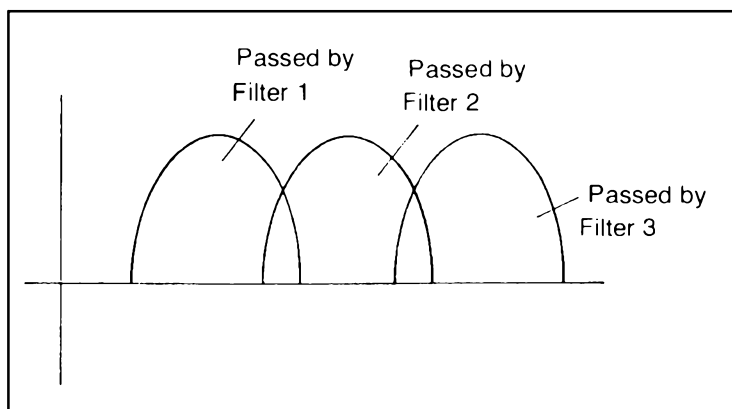


Fig. 5-19. Frequency response graphs of overlapping passbands of a filter bank.

center frequency	band	octave number
30 Hz	20-40	1
60 Hz	40-80	2
120 Hz	80-160	3
240 Hz	160-320	4
480 Hz	320-640	5
960 Hz	640-1280	6
1920 Hz	1280-2560	7
3840 Hz	2560-5120	8
7680 Hz	5120-10,240	9
15,360 Hz	10,240-20,480	10

These filters will have fixed center frequencies, so signals with different fundamentals may have quite different harmonic content and, therefore, varying sound qualities. For example, let's assume we have an octave filter bank as described above. Filters 1,3,5,7, and 9 will be set to pass all of the signal in their passbands. Filters 2,4,6,8, and 10 will be cut off completely. For simplicity we will assume all of the filters have ideal cutoff characteristics.

The first signal that will pass through the filter bank will be a 100 Hz square wave. A square wave, you should recall, contains all odd harmonics (third, fifth, seventh, etc.) of the fundamental frequency. With the filtering setup described in the previous paragraph, the output signal would consist of the following frequencies:

100 Hz	fundamental
500 Hz	fifth harmonic
1300 Hz	thirteenth harmonic
1500 Hz	fifteenth harmonic

(We will ignore the harmonics above the fifteenth because of their relative low levels.

Now let's take a look at what happens if we increase the square wave's frequency to 200 Hz.

600 Hz	third harmonic
1400 Hz	seventh harmonic
1800 Hz	ninth harmonic
2200 Hz	eleventh harmonic

Obviously these two signals will sound quite different. Filter banks generally aren't suitable for use on melodic lines because of

the fluctuating tonal quality the fixed filters will produce. For background and continuous (or drone) tones, however, a good filter bank can allow the electronic musician a great deal of freedom in customizing the harmonic content of the signal.

Figure 5-20 shows a simple circuit for use in a filter bank. Of course you will need one of these circuits for each band to be controlled. The unmarked components are selected for the desired center frequency, as discussed earlier in this chapter. The circuitry around the second op amp is simply an amplifier. The potentiometer is used to control how much of that filter's passed signal will reach the output of the filter bank.

THE VOLTAGE-CONTROLLED FILTER

A major problem with fixed filters was touched upon in the previous section. As the fundamental frequency of a complex signal changes, the harmonic content of the output signal may change drastically because the filter blocks and passes specific predefined frequencies regardless of their harmonic relationships. This can be quite a problem. In effect, it prohibits the use of filters on melodic lines unless you stop and readjust the filter for each note played. There has to be an easier way!

Fortunately there is. The solution is the voltage-controlled filter. The concept is similar to that of the voltage-controlled oscillator. A dc voltage is used to adjust the cutoff frequency (and

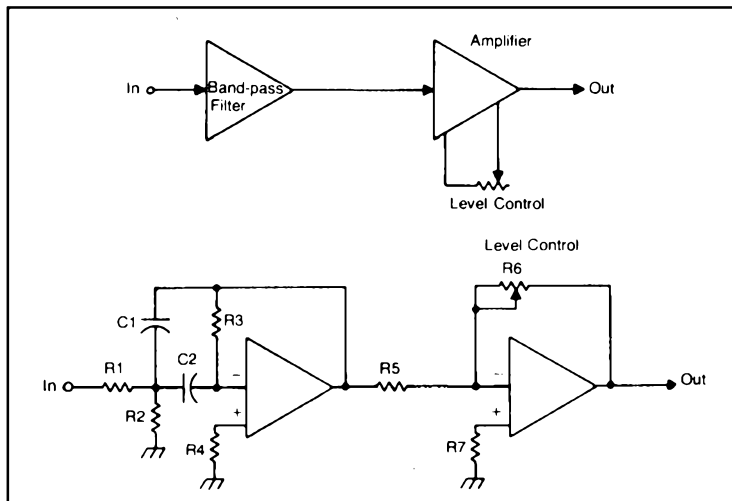


Fig. 5-20. Simple circuit for use in a filter bank.

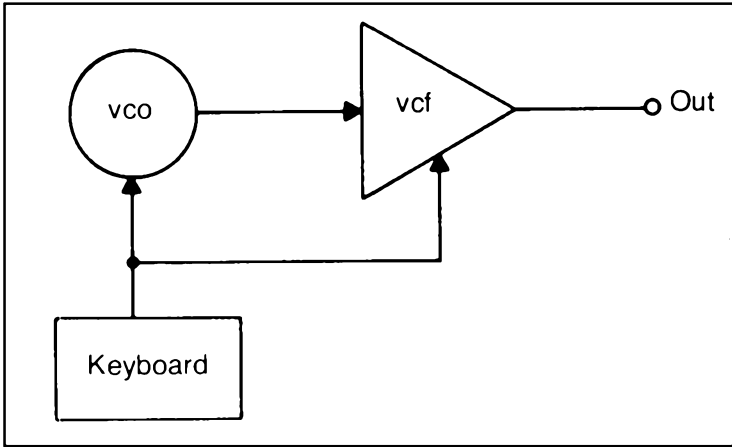


Fig. 5-21. Basic patch for using a vcf.

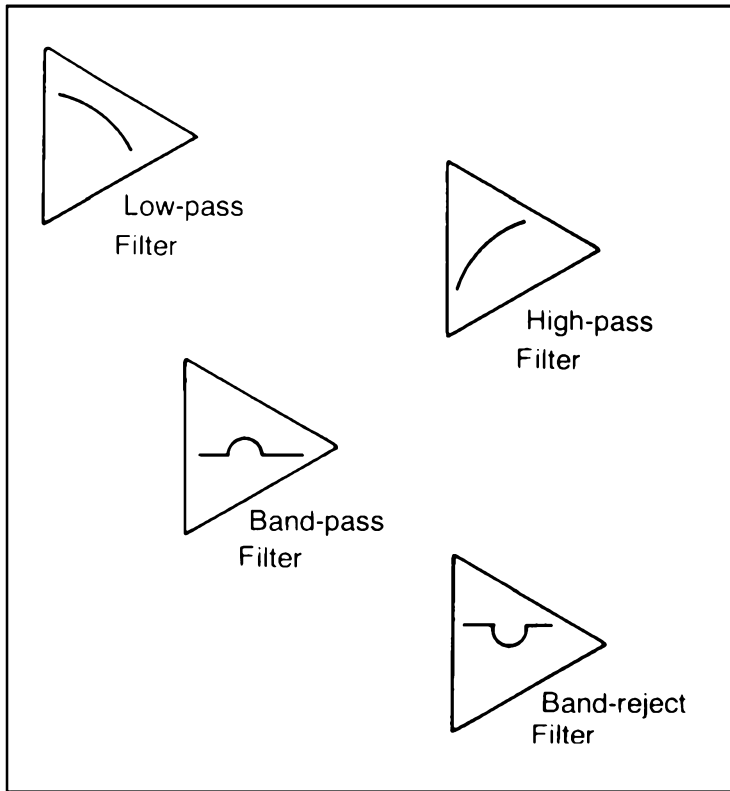


Fig. 5-22. Symbols for the basic filter types.

occasionally the Q , or band-width) of the filter. Ac voltages may also be used for special effects.

By controlling both a voltage-controlled oscillator (vco) and a voltage-controlled filter (vcf) with the same voltage, they will be synchronized with each other, and the harmonic content of the output signal will remain more or less constant, even though the specific frequencies involved may change drastically. The basic patch diagram is illustrated in Fig. 5-21.

As shown in this patch diagram, a triangle is used to represent a filter in patch diagrams. Usually a simplified frequency response graph will be drawn within the triangle to indicate the filter type. This is shown in Fig. 5-22. Alternatively, letters may be used to define the filter type, such as LP for a low-pass filter. This is demonstrated in Fig. 5-23.

Discrete Circuits

A relatively simple low-pass vcf circuit is illustrated in Fig. 5-24. The parts list is given in Table 5-13; although you may want to

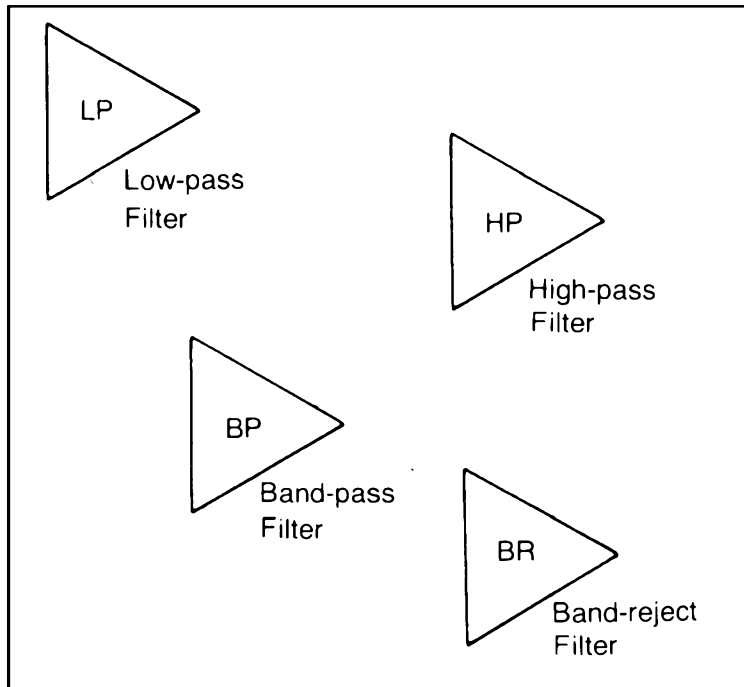


Fig. 5-23. Alternate symbols for the basic filter types.

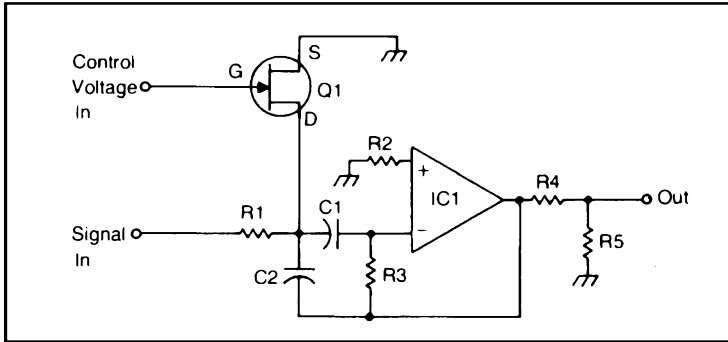


Fig. 5-24. Low pass vcf circuit.

experiment with other component values. The values given will result in a fairly shallow cutoff slope.

The control voltage applied to this circuit should be negative, with levels down to -5 volts. The lower the voltage, the lower the cutoff frequency. With a control voltage input of 0 volts, the cutoff frequency should be about 3000 Hz.

Despite the simplicity of this circuit, it can be extremely useful in an electronic music system.

A somewhat more complex voltage-controlled filter circuit is shown in Fig. 5-25. The primary output of this circuit is a band-pass filter, but low-pass and high-pass operation may be tapped off at the outputs of IC1 and IC3, respectively. The parts list for this circuit is given in Table 5-14.

Voltage-controlled filters are almost as important to the electronic musician as voltage-controlled oscillators. They are essential for subtractive synthesis.

Integrated Circuits

Dedicated integrated circuits for voltage-controlled filters are also available. Generally, any of the four basic filter types (low-

Table 5-13. Parts List for Fig. 5-26.

R1	27 k Ω resistor
R2, R3	1 M Ω resistor
R4	100 Ω resistor
R5	560 Ω resistor
C1, C2	0.0047 μ F capacitor
IC1	low noise op amp IC
Q1	FET (MPF-102, Radio Shack RS-2036, or similar)

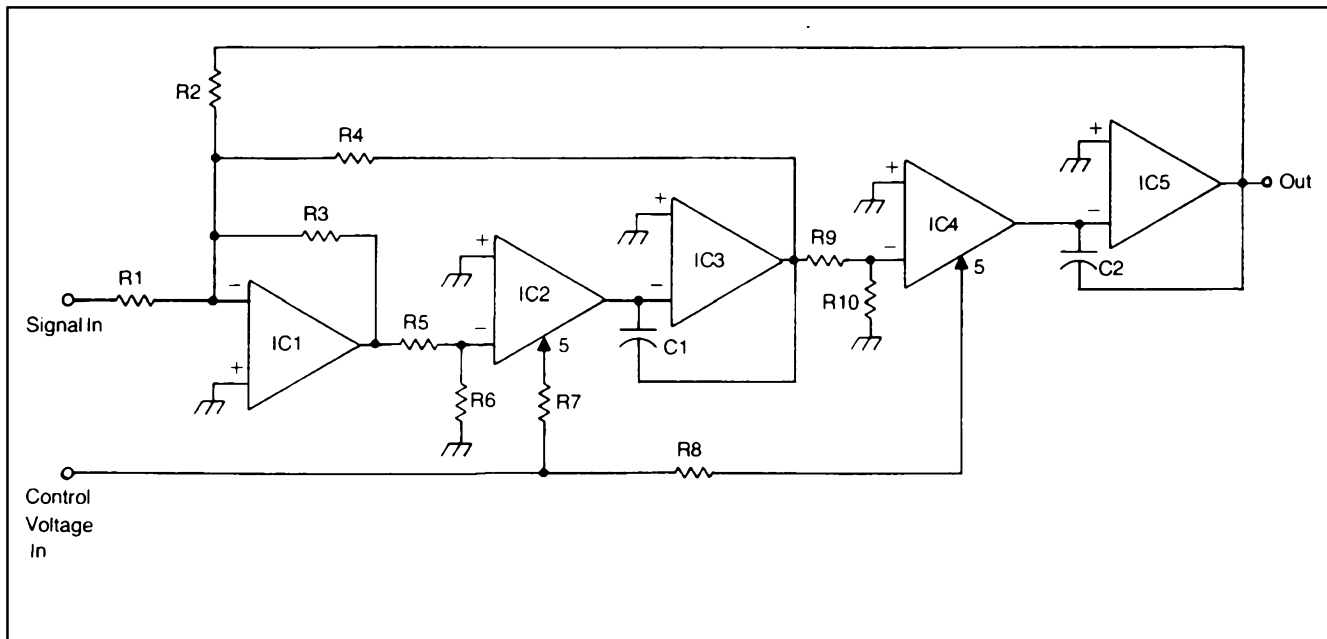


Fig. 5-25. A more complex vcf circuit.

Table 5-14. Parts List for Fig. 5-27.

R1, R2, R3, R5, R9	100 k Ω resistor
R4	22 k Ω resistor
R6, R10	220 Ω resistor
R7, R8	33 k Ω resistor
C1, C2,	0.01 μ F capacitor
IC1, IC3, IC5	low noise op amp IC
IC2, IC4	CA 3080 IC

pass, high-pass, band-pass, or band-reject) can be built around one of these chips.

One manufacturer of these devices is Curtis Electromusic Specialties, Inc. The pin-out diagram for the Curtis CEM 3350 vcf IC is shown in Fig. 5-26. This chip contains two complete filters. Only a handful of resistors and capacitors are needed to create two practical voltage-controlled filters. Both the cutoff frequency (or center frequency) and the Q of the filter may be voltage controlled, making a wide range of effects possible.

The other leading manufacturer of electronic music ICs is Solid State Music. The pin-out of their SSM 2040 voltage-controlled filter is shown in Fig. 5-27. This device contains four filter sections which

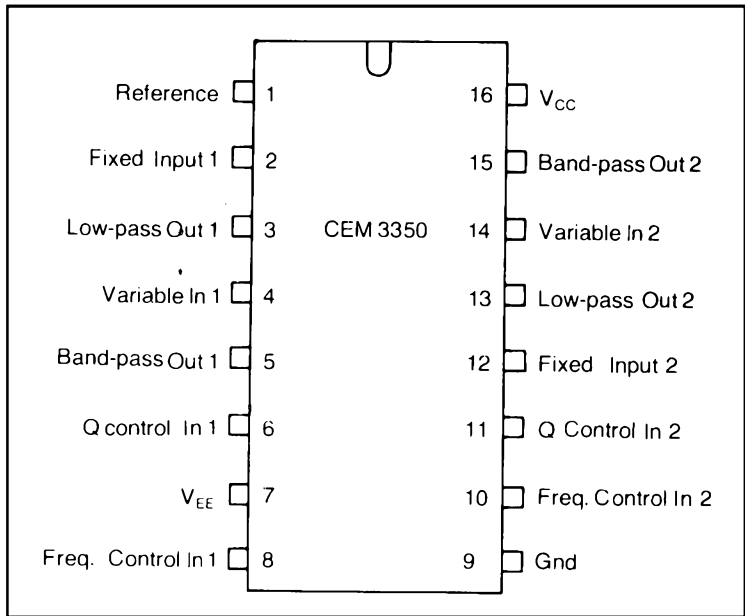


Fig. 5-26. Pin-out diagram for Curtis CEM 3350 vcf IC

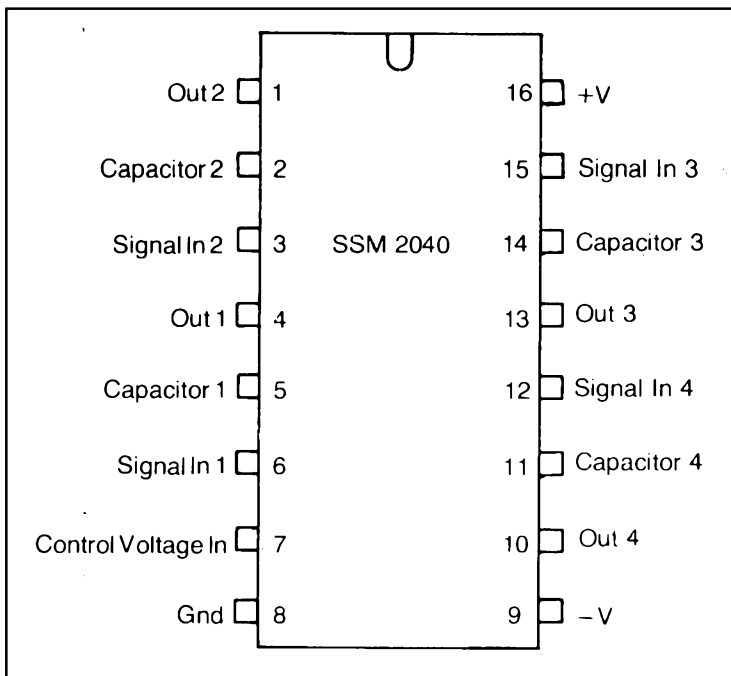
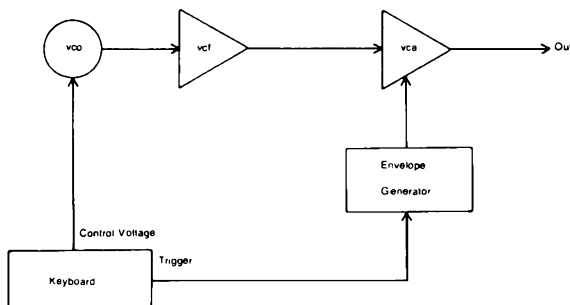


Fig. 5-27. Pin-out diagram for the SSM 2040 vcf IC

may be used separately, or combined into almost any desired filtering pattern. All four sections are controlled with a single control voltage input (pin 7).

Both of these manufacturers make special purpose chips that are used in high quality commercial synthesizers. Extensive data sheets are available from the manufacturers, or their distributors.

Chapter 6



Amplifiers

One of the most basic of electronic circuits is the amplifier. An amplifier is a device that can take a relatively weak signal and boost it to a higher level. In the previous chapters, we have already encountered several amplifiers that were part of more complex circuits. An oscillator is an amplifier with positive feedback. An active filter includes an amplifier to compensate for insertion losses. In this chapter, we will deal more directly with amplifiers as amplifiers.

An electronic music synthesis system includes a number of amplifiers in the form of oscillators, active filters, and so forth, but it will also include a number of devices for straight amplification. There are three basic types of amplification devices in a synthesizer. They are the monitoring sound system, preamps, and voltage-controlled amplifiers. Each of these will be discussed in this chapter.

THE SOUND SYSTEM

An electronic music synthesizer simply generates and modifies electrical signals. Some kind of sound system is required to convert these electrical signals into sounds. Basically a sound system consists of a relatively high-powered amplifier and one or more loudspeakers.

Amplifier Considerations

The first step in selecting a sound system for your synthesizer

is to determine whether you will be working primarily in the studio, or if you are going to be performing live in concert.

Obviously if you're performing live, you will need a relatively portable sound system. In a permanent studio, you can use heavier equipment that might even be built into immovable cabinets.

Ironically, while a live performance sound system should be smaller to increase portability, more power is required in a live performance situation. A studio will generally be a fairly small room, and you only need enough sound for you to hear it clearly. In a live concert, however, you will need to fill an auditorium that may be quite large. Moreover, there will be a number of people in the room while you're playing (at least you hope so). Since people's bodies absorb sound, greater power is required from an amplifier in a crowded room than in an empty room. A crowd may also be noisy. And in many cases (especially in rock music) you will want to play loudly.

This all boils down to a need for a relatively powerful amplifier. Just how powerful will depend on the size of the rooms you play in, how loud you like to play, and the condition of your wallet. Make sure you have enough power, but don't go overboard. A few years ago one manufacturer came out with a kilowatt (1000 watt) performance amplifier. Such a monster would come in handy if you were playing in the Grand Canyon. In more practical performance situations, however, it represents an awful case of overkill. Remember, the greater the wattage of an amplifier, the larger and bulkier it will be, creating transportation problems. In addition, everything else being equal, cost goes up steeply with amplifier power. Finally, a mammoth amplifier will eat up a lot of electrical power, and may tend to pop a lot of fuses.

Ideally your amplifier should be capable of putting out slightly more power than you need. Assuming the volume control is calibrated from 0 to 10, if you have to set it above 8, you probably need a larger amplifier. On the other hand, if you keep it set below 4, it's a pretty good bet that you're lugging around more amplification capability than you need.

While some musicians prefer tube amplifiers, I strongly recommend solid state equipment to the synthesist. Solid state amplifiers tend to be far lighter, less expensive, and more reliable than their tube equivalents. More importantly, as long as they aren't overdriven, they tend to produce less distortion than tube equipment. The warm, mellow sound of a tube amplifier comes from distorted signals. It may be fine for an electrical guitar, but a

synthesizer should be amplified as purely (distortion free) as possible. All effects on the signal should be fully under the musician's control. If you want the tube sound, you should be able to synthesize it without much trouble.

This is not to say solid state amplifiers are distortion free—not by a long shot. But the audible effects of transistor distortion (in properly selected, properly used equipment) tends to be far less noticeable than that of tube units.

In the studio, the situation is somewhat different. Huge amounts of power are not required, and might even cause problems. The ear tends to distort at high volumes, and when working on a tape composition, you should be able to hear precisely what you are putting onto the tape. In some cases, a very loud sound system can cause feedback problems throughout the entire system. And since few of us can afford a fully soundproofed studio, excessive volume from the amplifier can cause severe problems with family and neighbors.

In a live performance situation, you can compensate somewhat for mild distortion problems in your sound system. But in a recording studio, your monitoring equipment should pass as clean a signal as possible. The amplifier should have a reasonably flat frequency response across the entire audio range (20 Hz to 20,000 Hz). Some experts even recommend a wider range, say from 10 Hz to 50,000 Hz.

Ideally, the amplifier should be acoustically transparent. The only change in the signal at the output should be that it is at a higher level than at the input. Fortunately, modern amplification equipment is quite good, and the slight distortion that is produced is inaudible with normal signals.

Many people who are into electronic music as a hobby use their regular stereo system as a monitor. This is fine as long as it is convenient. However, it can quickly become a major nuisance if you have to keep plugging and unplugging the various accessories of your stereo system and your synthesizer. This can especially be true if you have to lug the stereo amplifier into another room for your electronic music work.

Loudspeaker Considerations

Actually, the amplifier is only half of the sound system. We also need one or more loudspeakers to convert the amplified electrical signals into actual sounds. Any sound system will only be as good as its weakest component, which is usually the speakers. Amplifiers

with distortion ratings of less than 0.1 percent are not uncommon or overly expensive (it is highly doubtful that anyone can actually hear distortion below about 1 percent), but a good loudspeaker generally has approximately a 3 percent distortion rating.

There are a number of things you can do to minimize the inaccuracies of practical speakers. The first and most obvious is to match impedances. Most modern amplifiers and speakers are rated for 8 ohms, but you may occasionally run into 4 ohm or 16 ohm units. I've encountered a few with impedances of 32 ohms, or even higher. The amplifier and the speaker should ideally have identical impedances for most efficient energy transfer between the devices. Unfortunately, impedance varies with frequency, so the rated impedance is really just an average. An 8 ohm speaker might actually have impedances ranging from less than 2 ohms to over 12 ohms. Ideally, the amplifier's impedance should fluctuate in the same pattern as the speaker so that the impedances will always match. This information is not easy to come by, however, so the best practical approach is extensive listening tests of the speakers with the amplifier you intend to use before you buy.

That advice is primarily for perfectionists. Modern solid state amplifiers are pretty forgiving in this area. Most amplifiers with 8 ohms rated output impedances will work just fine with 4 ohm or 16 ohm rated speakers.

Multiple speaker systems are a must for electronic music work. No single speaker can adequately cover the entire audible spectrum (20 Hz to 20,000 Hz). It is always a good idea to set up a division of labor approach, using a relatively large woofer to handle the low frequencies, and a tweeter to handle the high frequencies. A midrange speaker to cover the middle ground between the two extremes is also highly desirable. Remember, an electronic music synthesizer puts out a lot of harmonic loaded complex signals. Many of the sounds you will synthesize could be a real torture test for your speakers, so make sure your speakers can handle it.

The wattage of the speakers should be at least 25 percent higher than the rated output wattage of the amplifier. Most amplifiers can exceed their rated output wattage with increased distortion, and you can easily blow a speaker if you're not careful. Tweeters are especially susceptible in electronic music systems. A strong high frequency complex waveform will have a lot of harmonics beyond the audible range, and the tweeter could be damaged trying to reproduce high amplitude ultrasonic signals. To prevent this kind of problem, you could include a fixed cutoff low-pass filter

(cutoff frequency at about 20,000 Hz) as the last stage of your synthesis system before it reaches the amplifier. This will block the ultrasonic signals you can't hear anyway. Some less expensive amplifiers have a frequency response that rolls off at extremely high frequencies anyway, but you can't always rely on that.

If you are running a stereo, quadraphonic, or any other multiple source sound system, each of the speakers in the system should be identical. Mismatched speaker systems will result in a very unbalanced sound.

In a live performance situation, the need for high quality speakers is obvious. Good speakers are equally important in a studio monitor. Cheap speakers can have a drastic effect on many sounds. You have no way of knowing exactly what you are putting onto your tapes unless your speakers are capable of accurate reproduction. Believe it or not, some synthesized sounds may actually sound better through a cheap set of speakers. The same sound may seem far too thin and unpleasant when the tape is later played through a good sound system. There's no need to go overboard with super deluxe \$500 speakers in your studio, but you should insist on speakers that will let you be sure what you're hearing is what is going onto the tape.

Many studio electronic musicians prefer headphones. This is largely a matter of personal preference. However, you should give some consideration to how the finished tape is most likely to be heard. Electronic compositions—especially if they involve a lot of spatial effects—can seem entirely different when heard through headphones rather than through open air speakers.

PREAMPS

Preamp is short for preamplifier. The name is pretty much self-descriptive. A *preamplifier* is used to boost a weak signal up to a usable level before it is treated by other circuits. In a sound synthesis system, preamps are most frequently used with external signals that are fed into the synthesizer for modification. Microphones and tape heads produce very low level signals that are too weak for most synthesis modules to work with. The result is a very noisy signal, or no signal at all. Electric guitar pickups, on the other hand, generate relatively high level signals, so preamplification often isn't needed. In fact, in some cases, the signal from an electric guitar may have to be attenuated before being processed by a synthesizer.

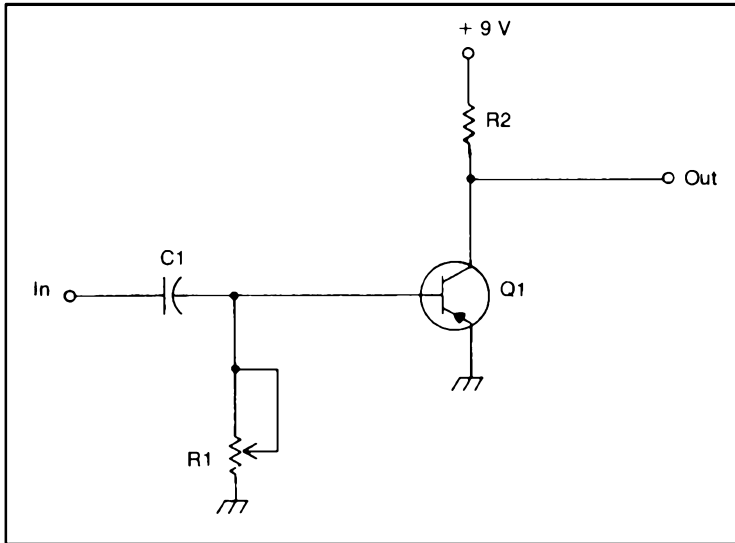


Fig. 6-1. Transistor preamp.

Active mixers, as discussed in Chapter 4, are essentially multiple-input preamps with independent level controls.

A fairly simple transistorized preamp circuit is shown in Fig. 6-1. The parts list is given in Table 6-1.

Another preamplifier circuit is illustrated in Fig. 6-2. This one is built around a low noise op amp IC. The parts list is given in Table 6-2.

Preamps are relatively mundane and unexciting modules in an electronic music synthesizer, but they are often needed. And since these circuits are so simple and inexpensive, there is no reason not to include a couple in your sound synthesis system just in case.

THE VOLTAGE-CONTROLLED AMPLIFIER

In previous chapters we have encountered the voltage-controlled oscillator (vco), and the voltage-controlled filter (vcf), so

Table 6-1. Parts List for Fig. 6-1.

R1	10 k Ω potentiometer
R2	1 k Ω resistor
C1	0.047 μ F capacitor
Q1	PNP transistor (Radio Shack RS2007, or similar)

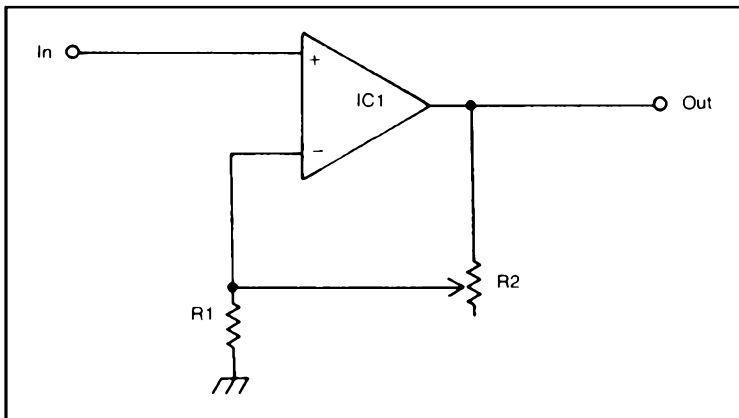


Fig. 6-2. Op amp preamp.

you should not be surprised to learn that the third most important module in a sound synthesis system is the voltage-controlled amplifier.

In a vco, the control voltage sets the oscillator frequency., In a vcf, the control voltage determines the cutoff frequency. In a vca, the control voltage sets the signal level, or amplitude. It essentially behaves like an electric volume control. A high control voltage would place a strong signal at the output of a vca, while a low control voltage would attenuate the output signal. This may not sound like very much, but it is an invaluable tool in the synthesist's bag of tricks.

Circuits

Before discussing some of the many ways voltage-controlled amplifiers can be used in electronic sound synthesis, let's examine a few typical circuits.

Figure 6-3 shows one of the simplest vca circuits I've encountered. It is made up of just five components. The parts list is given in Table 6-3. While many different NPN transistors may be used, the diode must be a germanium type. Silicon diodes will not work in this application.

R1	1 k Ω resistor
R2	100 k Ω potentiometer
IC1	low noise op amp IC

Table 6-2. Parts List for Fig. 6-2.

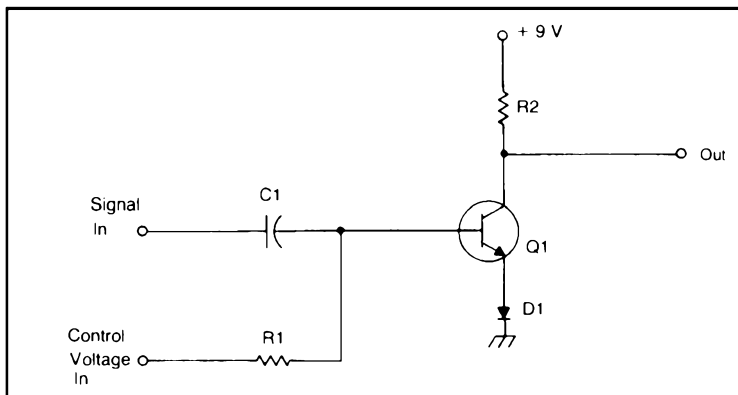


Fig. 6-3. Simple vca.

This circuit has a fairly poor low frequency response, and it is not recommended for frequencies below about 100 Hz to 200 Hz. However, it works fine with higher frequencies. This circuit's frequency response extends well past the upper end of the audible range.

Two additional vca circuits are illustrated in Fig. 6-4 and Fig. 6-5. Their parts lists are given in Table 6-4 and Table 6-5 respectively.

Special purpose IC vcas are also available from Curtis Electromusic Specialties, Inc. and Solid State Music. Figure 6-6 shows the pin-out diagram for the SSM 2010. Clearly this is a fairly simple device to use. The control voltage provides linear (see Fig. 6-7) or exponential (see Fig. 6-8) control. The human ear detects changes of volume in an exponential fashion, but linear control can be useful for certain special effects.

Uses

There are many ways a voltage-controlled amplifier can be used in electronic music. A control voltage can adjust the output

Table 6-3. Parts List for Fig. 6-3.

R1	470 k Ω resistor
R2	22 k Ω resistor
C1	0.68 μ F capacitor
D1	germanium diode (1N34A or similar)
Q1	NPN transistor (2N3564, GE-10, RCA SK 3019, Motorola HEP-54 Radio Shack RS-2011, or similar)

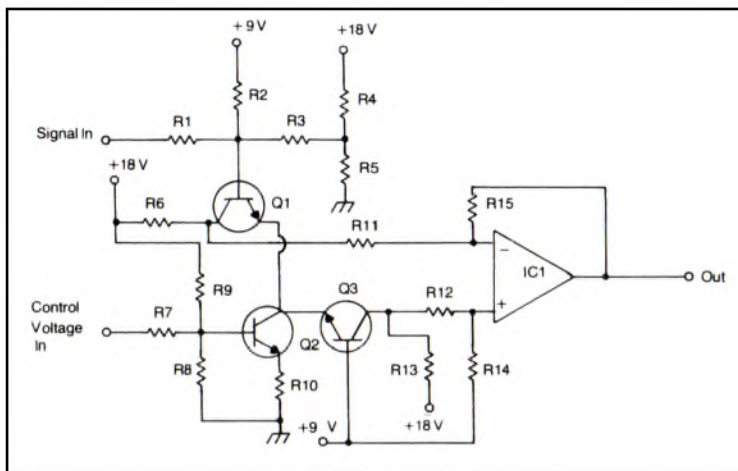


Fig. 6-4. Improved vca.

level faster and with far greater accuracy than any human being possibly could. The four most common uses of vcas are gating, creation of new timbres, tremolo, and amplitude modulation. Each of these will be discussed in the following pages.

Gating. A vca can be used to temporarily block a continuous signal. This can help prevent annoying clicks that can be caused by switching the signal source on and off.

A typical gating patch is shown in Fig. 6-9. Notice that two outputs are taken off of the keyboard. One is the variable voltage that controls the pitch of the oscillator. The other signal is simply a straight voltage that is fed to the output whenever one or more keys are held down. Most synthesizer keyboards have at least two sets of switch contacts for these types of output signals. (In a later section

Table 6-4. Parts List for Fig. 6-4.

R1	33 k Ω resistor
R2, R10	2.2 k Ω resistor
R3	10 k Ω resistor
R4	100 k Ω resistor
R5, R7	120 k Ω resistor
R6, R11, R12, R13	6.8 k Ω resistor
R8	22 k Ω resistor
R9	68 k Ω resistor
R14, R15	15 k Ω resistor
Q1, Q2, Q3	NPN transistor (2N3904, Radio Shack RS-2009, or similar)
IC1	low noise op amp IC

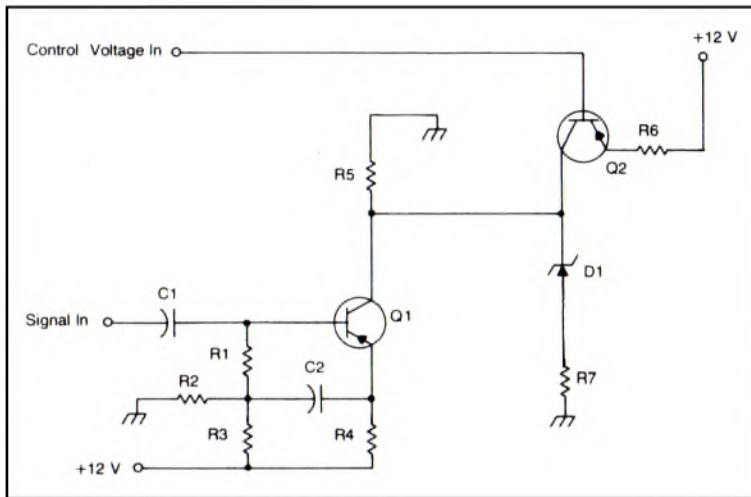


Fig. 6-5. Another vca.

Table 6-5. Parts List for Fig. 6-5.

R1, R5	1 k Ω resistor
R2	22 k Ω resistor
R3	4.7 k Ω resistor
R4	1.8 k Ω resistor
R6	120 Ω resistor
R7	56 Ω resistor
C1	0.1 μ F capacitor
C2	1 μ F capacitor
D1	6 V zener diode
Q1, Q2	PNP transistor (2N1384, QE-9, RCA SK3008, Motorola HEP-2, or similar)

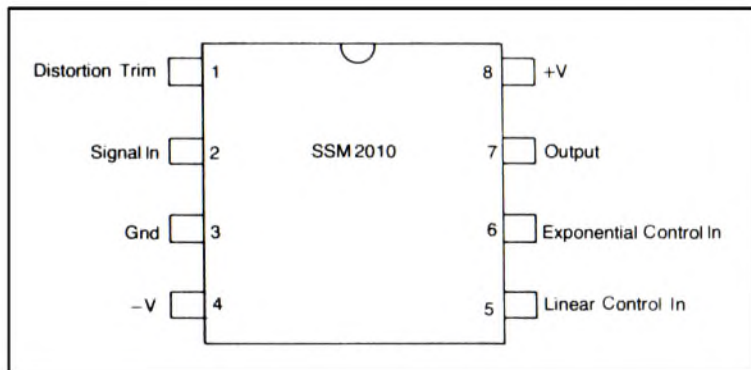


Fig. 6-6. Pin-out diagram for the SSM 2010 vca IC

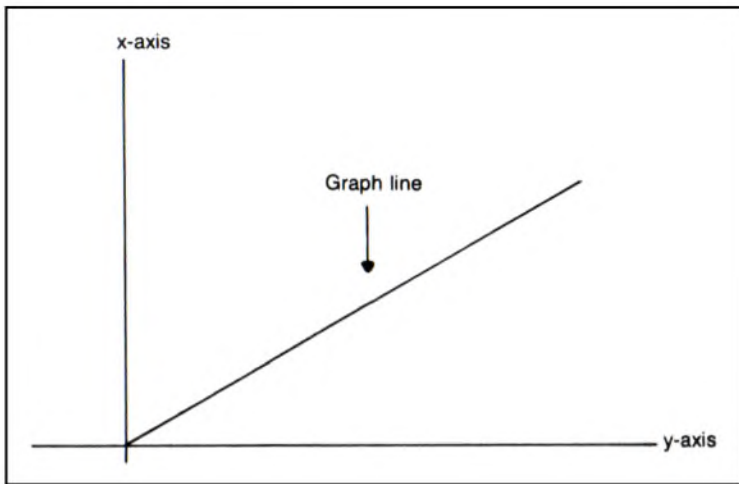


Fig. 6-7. Linear graph.

we will deal with a modification of the gate signal called the trigger pulse).

When no key in this patch is depressed, the oscillator will continue to generate a signal, probably with a downward drifting frequency. This could obviously make playing any kind of melodic

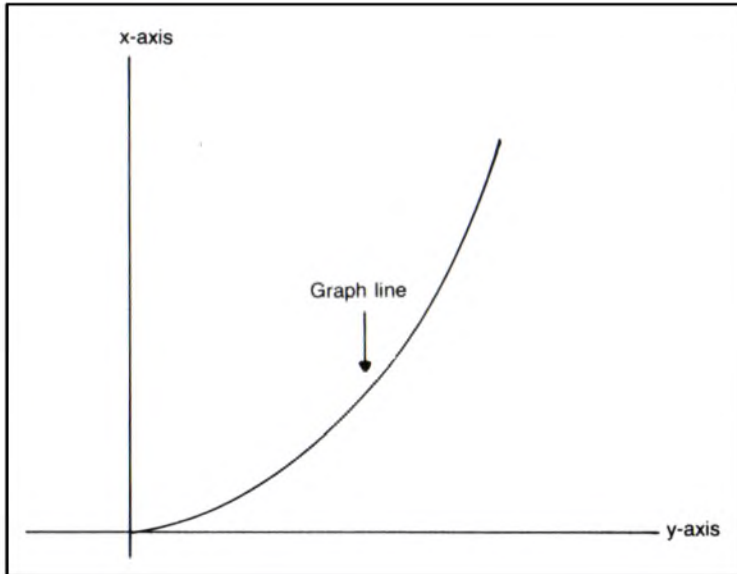


Fig. 6-8. Exponential graph.

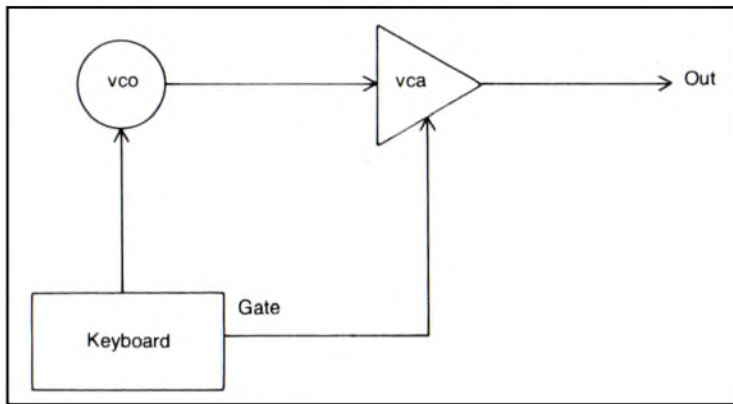


Fig. 6-9. Gating patch.

line exceedingly difficult, and rests (momentary silences) would be impossible unless you manually switched the oscillator on and off.

However, when no key is depressed, there will not be a gate signal, so the vca's level would be held to an output amplitude of 0, or silence. When a key is pressed, the gate signal will turn up the volume on the vca, and the tone will be heard at the output for as long as the gate signal is present (i.e., a key is held down).

Timbres. In the simple gating discussed above, the output signal is either at full volume, or cut off. Most natural sound sources take some finite time for the volume to build up from the minimum level to the maximum level and a finite amount of time to die back down to the minimum level again. These times are often very brief, and in many cases we can't hear them as changing amplitudes, but they do have a very definite effect on the sound. The changing amplitude pattern of a sound is called its *envelope*, and its audible effect is called *timbre* (pronounced tam-ber).

An amplitude envelope consists of at least two parts. The attack (the time it takes the sound to build from its minimum level to its maximum level) and the decay (the time it takes the sound to die down from the maximum level to the minimum level). Some typical attack/release (AR) envelopes are illustrated in Fig. 6-10. Figure 6-10A would give a very percussive effect. Figure 6-10B has a short attack and a long release or decay, and would sound rather like a guitar or a piano. Figure 6-10C has a long attack and a long decay, which would tend to sound like some kind of wind instrument. A long attack/short decay envelope, like the one shown in Fig. 6-10D, has a very unnatural electronic effect. Such envelopes are not found in nature.

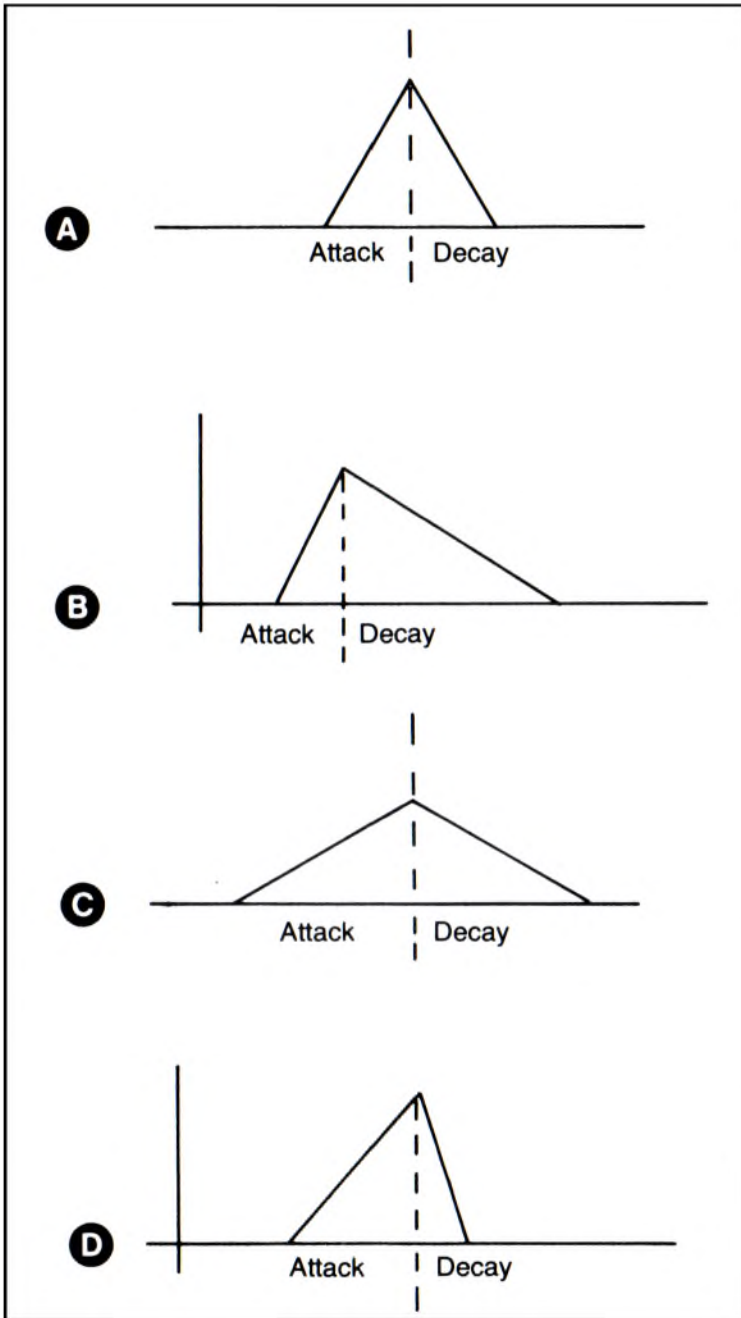


Fig. 6-10. Some typical attack/decay envelopes.

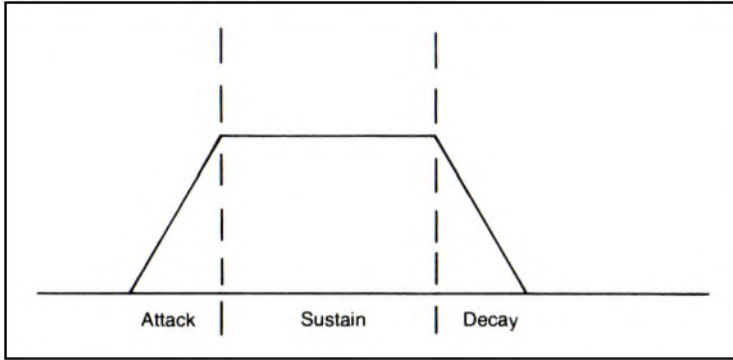


Fig. 6-11. An attack/sustain/decay envelope.

Most sounds have more complex envelopes than the simple AR types shown in Fig. 6-10. For one thing, many sound sources can hold the tone at the maximum level for some time. This is called *sustain*. An attack/sustain/decay envelope is shown in Fig. 6-11.

A still more complex envelope is illustrated in Fig. 6-12. First there is an attack, then an initial decay down to a specific sustain level which is held until the final release that fades back down to the minimum level. This is the most commonly used envelope format in electronic music. Varying voltages that create these patterns are produced by special circuits called *envelope generators*. Sometimes they are referred to as function generators. These devices will be discussed in Chapter 8. For now, just accept the fact that these

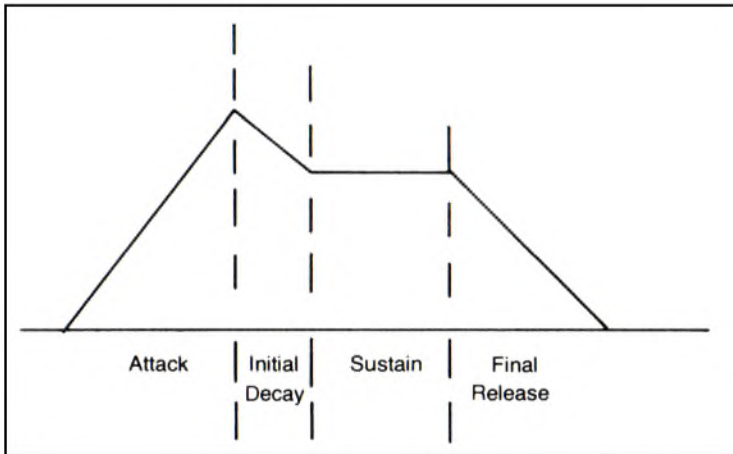


Fig. 6-12. An attack/decay/sustain/release envelope.

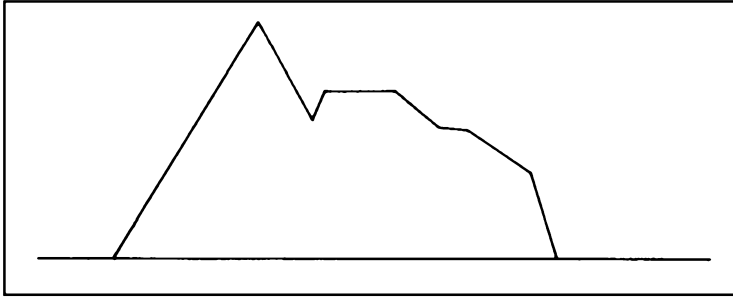


Fig. 6-13. A complex envelope.

envelope signals can be easily generated in an electronic music synthesis system.

Many sounds in the real world have extremely complex envelopes that may be difficult (although not necessarily impossible) to simulate electronically. An example is illustrated in Fig. 6-13. Moreover, each harmonic in a complex signal may have a different envelope, an effect that can be recreated only through some form of additive synthesis (See Chapter 4).

Fortunately, the ear is not all that precise, especially when several sounds are heard simultaneously, so the standard, easily generated AR, or the ADSR (attack/decay/sustain/release) envelopes may be used.

The basic envelope generator patch is shown in Fig. 6-14. The envelope generator can be controlled by either the keyboard's gate signal (discussed earlier) or its trigger pulse signal. As long as the

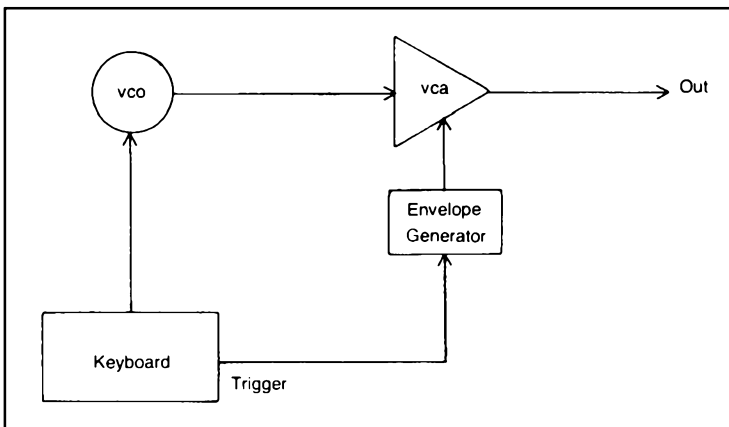


Fig. 6-14. Basic envelope generator patch.

key is held down, the gate signal will be at its maximum level, so the envelope is held at its sustain level as long as the key is depressed.

The trigger pulse on the other hand, is of a fixed length. As shown, when a key is depressed, a brief voltage burst is fed through to the envelope generator, starting the attack. But the trigger pulse rapidly dies out so that there is no sustain—the envelope moves immediately from the attack portion to the decay portion.

Envelope generators and their use will be described in more detail in Chapter 8.

Tremolo. You should recall that in discussing voltage-controlled oscillators and voltage-controlled filters, it was mentioned that ac signals (such as the output of an oscillator) can control a voltage-controlled module as well as a dc voltage. If a low frequency (4 Hz to 10 Hz) oscillator controls a vca, there will be a warbling effect to the sound. This warbling effect is called *tremolo*. It is somewhat similar to vibrato, although not quite as warm sounding. Sine waves or triangle waves work best.

Amplitude Modulation. When a vco is controlled by a low frequency oscillator signal, we got the effect known as vibrato. As the controlling frequency is increased into the audio range, however, phantom signals called sidebands are produced, and we got the effect known as frequency modulation, or FM.

Similarly, if the oscillator producing a tremolo effect by controlling a vca is increased in frequency to the audible range, a new complex tone is produced. This effect is called *amplitude modulation*, or AM.

Amplitude modulation produces sidebands somewhat like frequency modulation, although not as many of them.

Let's assume that the main signal oscillator is feeding the vca with a 2000 Hz sine wave, and the modulation oscillator is controlling the vca with a 500 Hz sine wave. The output will include both of these frequencies, and their sum and difference. In other words, the output signal will contain the following four frequencies:

500 Hz	modulating frequency
1500 Hz	difference
2000 Hz	main signal
2500 Hz	sum

Any pair of sine waves will produce just two sidebands regardless of the signal strength. A set of sum and difference frequencies will be produced for each harmonic in either of the input signals.

As a second example, we'll say the main signal is a 600 Hz square wave. A low-pass filter cuts off everything above 5000 Hz, so the main signal consists of the following frequencies:

600 Hz	fundamental
1800 Hz	third harmonic
3000 Hz	fifth harmonic
4200 Hz	seventh harmonic

The modulating signal will still be a 500 Hz sine wave, which will modulate (and create a sum and difference frequency pair of sidebands) with each of the harmonics. The patch diagram for this is illustrated in Fig. 6-15. The sum and difference frequencies are summarized in Table 6-6. The output signal would therefore consist of thirteen frequencies.

100 Hz	difference—fundamental
500 Hz	modulating frequency
600 Hz	fundamental
1100 Hz	sum—fundamental
1300 Hz	difference—third harmonic
1800 Hz	third harmonic
2300 Hz	sum—third harmonic
2500 Hz	difference—fifth harmonic
3000 Hz	fifth harmonic

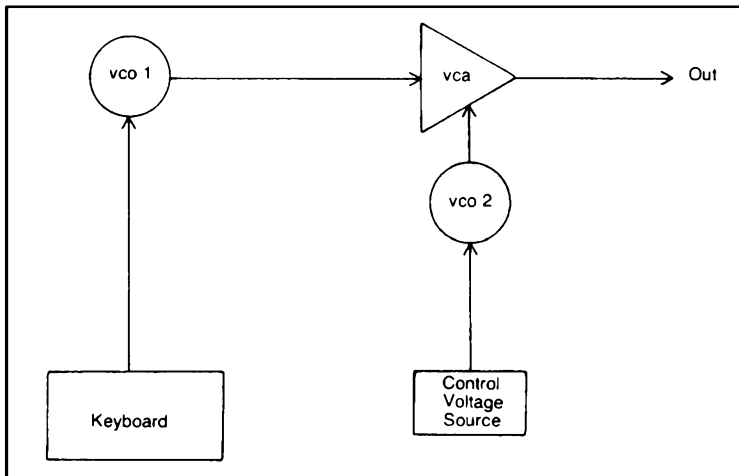


Fig. 6-15. AM example.

Table 6-6. AM Example.

Harmonic	Fundamental	Third	Fifth	Seventh
Program	600 Hz	1800 Hz	3000 Hz	4200 Hz
signal				
Difference	100 Hz	1300 Hz	2500 Hz	3700 Hz
(- 500 Hz)				
Sum	1100 Hz	2300 Hz	3500 Hz	4700 Hz
(+ 500 Hz)				
Modulating frequency = 500 Hz				

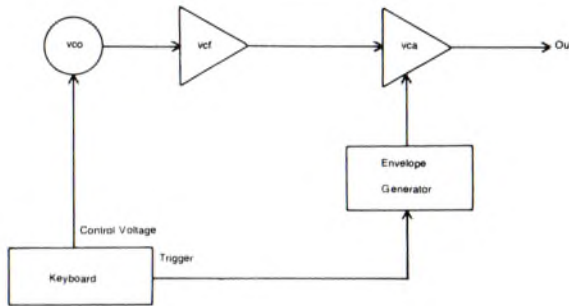
3500 Hz	sum—fifth harmonic
3700 Hz	difference—seventh harmonic
4200 Hz	seventh harmonic
4700 Hz	sum—seventh harmonic

If the modulating signal consists of a number of harmonics too, the situation becomes even more complex. For best results, at least one of the signals in amplitude modulation should be a sine wave or a triangle wave (preferably low-pass filtered). More complex signals tend to create muddy sounding outputs with no definite sense of pitch. Of course, in some cases, this may be just what you want.

Notice that unless the main signal and the modulating signal are harmonically related, the various overtones and undertones will not be harmonics of the perceived fundamental.

In most cases, the perceived fundamental of an amplitude modulated signal will be the same as the fundamental of the original main program signal. However, if the modulating signal is very strong (high level) its fundamental may take precedence.

Chapter 7



Manual Controllers

Of course the most fantastic array of synthesis modules in the world aren't going to be worth much if you have no way to control them. In this chapter, we will explore a few of the many ways a musician can control various modules. In the next chapter, we will look at some automatic controllers that allow a synthesizer to more or less play itself.

These two chapters do not describe every possible method of controlling sound synthesis equipment. There is almost infinite room for imaginative new ideas. We will be concentrating on the more commonly used controllers.

KEYBOARDS

Without a doubt, the most familiar manual controller for electronic music synthesizers is the keyboard. Generally, a standard 12 notes to an octave keyboard, similar to those used on organs or pianos, is used. This can be both an advantage and a disadvantage.

The familiarity of the keyboard format can make an electronic sound synthesis system seem far less strange and intimidating. Many people are a bit afraid of the technology and a standard keyboard allows them to approach the synthesizer more like a standard musical instrument. Of course musicians who play other keyboard instruments should have little difficulty adapting to a synthesizer equipped with a standard keyboard.

Another advantage of using standard keyboards on a synthesizer is that music can be easily notated on standard music manuscript paper. Other forms of synthesizer control (discussed shortly) generally require the composer to create some new form of notation—usually a fairly complex system that can't be easily read by other musicians without special training.

In other words, the major advantage of using a standard organ-like keyboard to control an electronic music synthesizer is the familiarity that allows the musician to approach the synthesizer like a traditional musical instrument. The chief disadvantage is its familiarity that allows the musician to approach the synthesizer like a traditional musical instrument. That is not a misprint. The comforting familiarity of a standard keyboard is both an advantage and a disadvantage.

If the musician approaches the synthesizer too much as if it were a traditional musical instrument, he will cut off many of the unique possibilities the synthesizer allows. An electronic music synthesizer is capable of far, far more than any traditional instrument.

While a traditional format keyboard makes playing and notation easy, there is no reason why it has to be used traditionally. For one thing, a synthesizer keyboard can be tuned to different scales. For example, a quartertone scale or a whole step scale. Of course with a nontraditional scale, the notes and octaves won't correspond directly with the pattern of black and white keys. For instance, a four octave keyboard could be tuned so that it covers only a single octave, with many more in-between notes. Or, the scale could be compressed so that a four octave keyboard covers eight octaves—two real octaves to each keyboard octave.

Even when a nontraditional scale is used, the keys played can be notated as if the standard equally tempered twelve note to an octave scale was being used.

Nor is there any law that says a keyboard can only control pitch. Any voltage-controlled module can be controlled by the keyboard. A keyboard can control a voltage-controlled filter or a voltage-controlled amplifier as well as a voltage-controlled oscillator. It may take a little time getting used to playing a keyboard that controls the volume of the sound, but it may well be just the effect the piece you're working on needs. The whole point of a synthesizer is the wide variety of new possibilities that are not practical on more traditional instruments.

Electrical Requirements

In an electronic musical instrument like a synthesizer, each key on the keyboard is essentially a switch. Most synthesizer keyboards have two switches per key. One switch determines the voltage to be fed through to the control output. This is usually accomplished by a string of switch selectable resistors, as illustrated in Fig. 7-1. These resistors are generally variable trimpots for fine tuning.

The second set of switches make connections to a common bus line, as shown in Fig. 7-2. Notice that as long as one or more keys is held down, the bus voltage is fed to the output. This is the gate

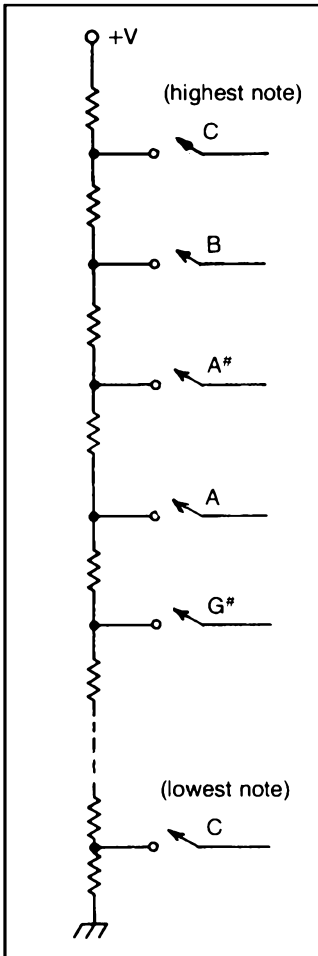


Fig. 7-1. Switch selectable resistors for a keyboard.

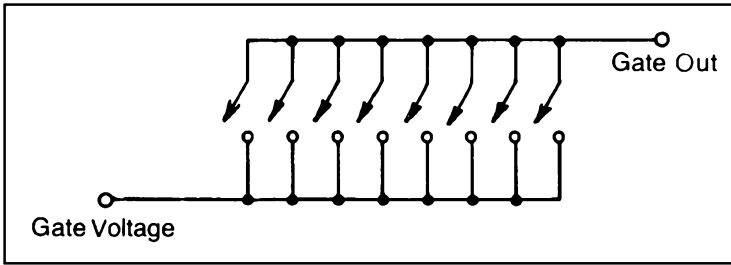


Fig. 7-2. Gate bus.

signal described in Chapter 6. This signal is generally used to control voltage-controlled amplifiers (Chapter 6) or to trigger envelope generators (Chapter 8). The gate voltage is either at its maximum level, or at 0 volts. There are no intermediate levels.

A trigger pulse (also used to trigger envelope generators) may be created from a gate signal with the circuit shown in Fig. 7-3. The parts list is given in Table 7-1.

A trigger pulse is a brief voltage burst that is fed to the output when a key is depressed. It is of a fixed length (usually very brief), no matter how long the key is held down. To generate a new trigger pulse, the key must be released and a new key (or the same one) must then be depressed.

A gate signal is illustrated in Fig. 7-4, and a trigger pulse is shown in Fig. 7-5. The ways these signals are used will be dealt

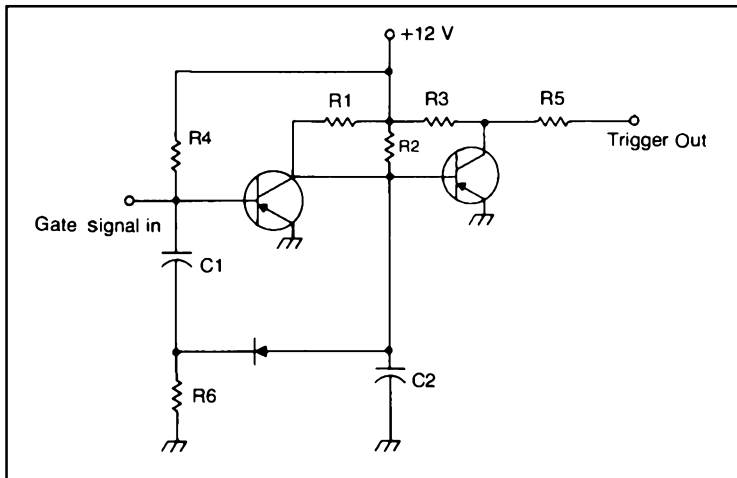


Fig. 7-3. Gate to trigger pulse converter.

Table 7-1 Parts List for Fig 7-3

R1	33 k Ω resistor
R2	150 k Ω resistor
R3	10 k Ω resistor
R4	330 k Ω resistor
R5	10 k Ω resistor
R6	3.9 M Ω resistor
C1, C2	0.0047 μ F capacitor
Q1, Q2	NPN transistor (2N3904, Radio Shack RS-2016, or similar)

with in more detail in the section on envelope generators in Chapter 8.

Some synthesizers use additional switch contacts on each key for special purposes. Some keyboards generate different voltages depending on how far each key is depressed. Many commercial synthesizers have velocity sensitive keyboards. This means an extra voltage is fed to an output line. This voltage is proportional to how fast (or how hard) each key is struck. This allows for greater musical expression in playing the keyboard, giving the synthesizer more of the sensitivity of a piano. Unfortunately, the required circuits are a bit complex, and the required components are hard to get and rather expensive, putting velocity sensitive keyboards out of the reach of most experimenters, at least for the time being. But with the technology constantly evolving, who knows what may be possible a few years from now.

Polyphonic Keyboards

The basic keyboard switching arrangement just described is a purely monophonic system. That is, only one note (or whatever)

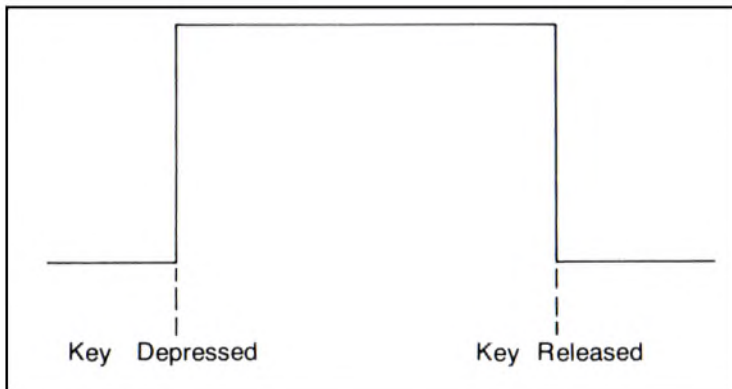


Fig. 7-4. Gate signal.

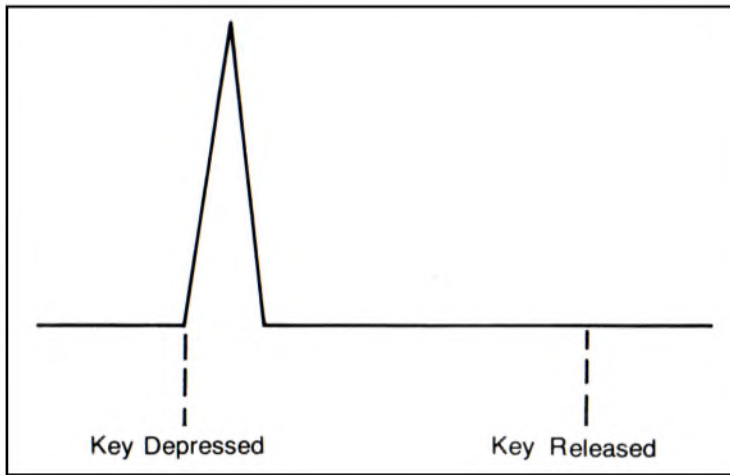


Fig. 7-5. Trigger pulse signal.

may be played at a time. Depressing multiple keys would have no effect on the gate or trigger pulse signals, and the output voltage would be some oddball composite because of the effects of series and parallel resistances.

Many musicians prefer polyphonic systems, and this is understandable. The ability to play chords, or more than a single melodic line is quite desirable musically. Many commercial synthesizers now feature limited or fully polyphonic keyboards. Unfortunately, the circuitry involved is rather tricky, and I feel outside the scope of this book.

The usual technique for polyphonic keyboards is to rapidly scan each key several times a second. Essentially, a specialized circuit repeatedly checks each key in turn to see if it is currently depressed. If it is not, the scanning simply moves on to the next key, or if it is, the appropriate output signal is generated, and the scanning continues. The scanning is done at a very high rate to avoid missing anything during fast musical passages. Scanning rates are often in the range of a million times per second—that's considerably faster than any human musician could play.

If you feel a strong need for polyphonic capabilities, you could borrow from organ technology and use a top octave circuit. The 50240 is a dedicated IC for just this purpose. The pin-out diagram for this 16-pin chip is shown in Fig. 7-6.

A high speed clock (high frequency square wave) signal is fed into the top octave generator. The clock frequency is divided by

appropriate constants to create twelve equally tempered frequencies, or a single octave. Any one or more of these outputs may be fed to the output of the circuit. For instance, if switches S1, S5, and S8 are closed, a major triad will be fed to the output.

By changing the frequency of the clock, the output pitches can be raised or lowered into different keys, while remaining perfectly in tune with each other.

Generally a single octave of pitches is not enough for musical purposes. Additional octaves of each note can be tapped off with a circuit known as a flip-flop, which essentially divides each frequency by two. The direct output of the top octave generator will be the highest octave.

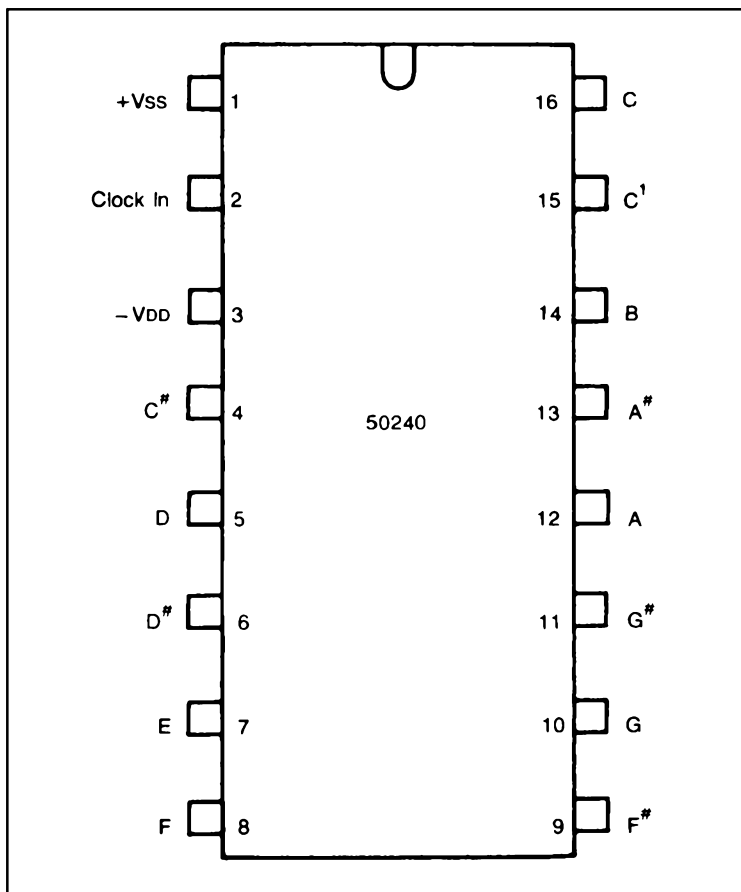


Fig. 7-6. Pin-out diagram for the 50240 top octave generator IC.

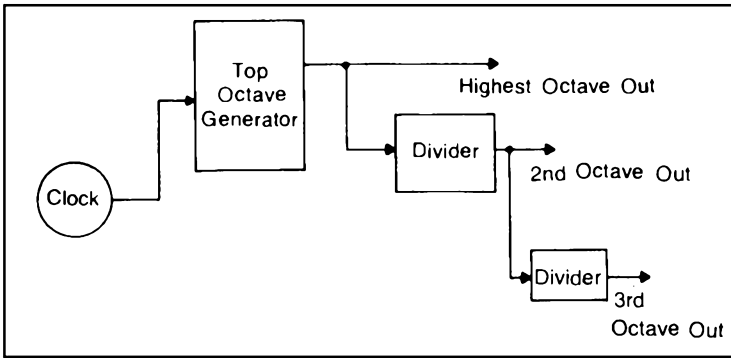


Fig. 7-7. Top octave circuit block diagram.

A block diagram of a complete top octave circuit is shown in Fig. 7-7. A practical schematic is illustrated in Fig. 7-8, with Table 7-2 providing the parts list.

There are some disadvantages to using the top octave generator approach in an electronic music synthesizer, however.

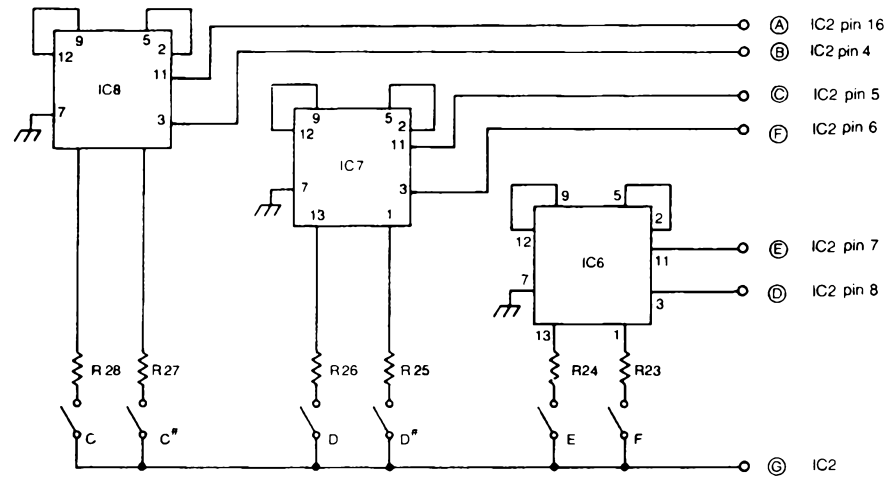
First off, only the internally generated square wave signals may be controlled by the keyboard. You can not operate this keyboard with voltage-controlled modules like vcfs, vcas, or vcOs that generate waveforms other than square waves. In addition, only standard equally tempered scales may be produced.

These restrictions obviously limit the versatility of the synthesizer. For many musicians, however, the advantages of polyphonic capability far outweighs these problems. This is especially true for musicians who use their synthesizers to play more or less traditional types of music.

If these limitations do bother you, but you still want polyphonic capability without a great deal of expense, you could always use two keyboards with your synthesizer. A top octave keyboard for polyphonic work, and a standard monophonic keyboard as previously described for controlling other voltage-controlled modules.

Table 7-2. Parts List for Fig. 7-8.

R1	2.7 k Ω resistor
R2	10 k Ω potentiometer
R3	10 k Ω resistor
R4-R28	18 k Ω resistor
C1	100 pF capacitor
IC1	CD4001 quad NAND gate IC
IC2	50240 top octave generator IC
IC3-IC8	CD4013 dual flip-flop IC



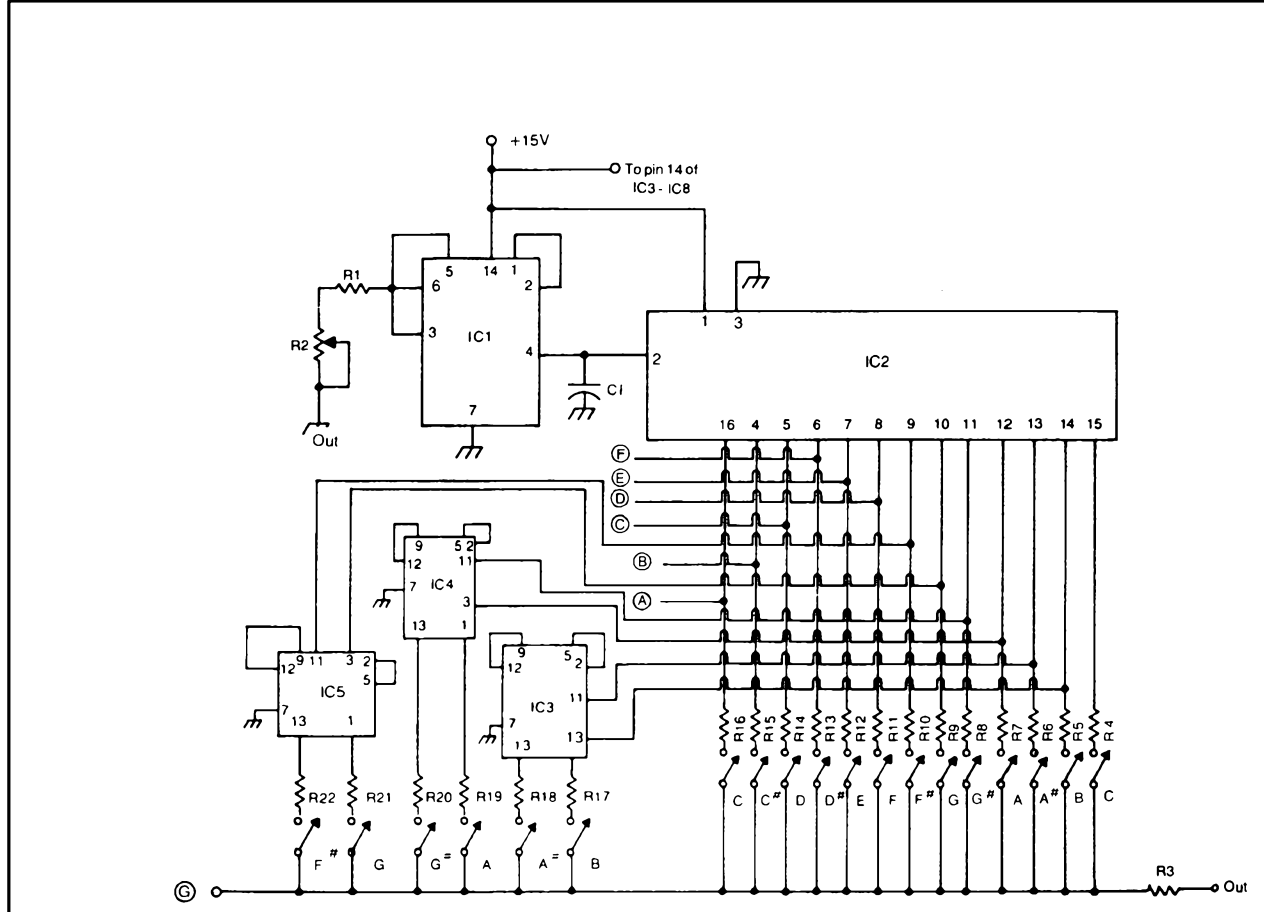


Fig. 7-8. Top octave circuit.

This is a good compromise since polyphonic capability is seldom needed when using a keyboard to control a vcf or a vca.

Obtaining A Keyboard

Often the biggest problem facing the experimenter building his own electronic music synthesizer is finding a keyboard. This isn't as easy as buying resistors and capacitors at your neighborhood Radio Shack.

Occasionally a few surplus organ or synthesizer keyboards will be advertised in one of the surplus house catalogs. Addresses may be obtained from the advertising sections of the electronics hobbyist magazines. Surplus keyboards are usually available only in small quantities and generally sell out very quickly.

A few companies do sell new keyboards as regular items. PAIA, a company that specializes in sound synthesis kits currently offers three keyboards with two switches to a key, as discussed above. They sell one and a half octave (18 note), three octave (37 note) and 5 octave (61 note) keyboards. Their address is 1020 W. Wilshire Blvd., Oklahoma City, OK 73116.

You may be able to get ahold of a defective organ or synthesizer from a local musical instrument dealer. How much you have to pay will depend on the dealer and the condition of the instrument. One advantage of using this source for a keyboard is that you can usually re-use the case to house at least part of your customized instrument. Sometimes you can even use some of the original circuitry.

Keyboards can also be adapted from toy instruments. These instruments are generally inexpensive enough to make buying a new one just for the keyboard economically feasible. Keyboards from toy musical instruments will rarely have switch contacts already installed. You will have to come up with some customized arrangement of your own. One possible approach is illustrated in Fig. 7-9. Figure 7-10 shows another possibility. Sometimes adding the switches to the keys can be difficult and frustrating. You should be prepared for some tedium before buying a toy organ for its keyboard.

Some experimenter's build keyboards from scratch. While this can be relatively inexpensive, it requires quite a bit of dedication to the project in many cases. If you build your own keyboard, you are not limited to the traditional twelve notes to an octave arrangement. However, bear in mind the fact that if you opt for a nontraditional format, you will defeat many of the advantages of using a keyboard in the first place. These advantages were discussed earlier in this

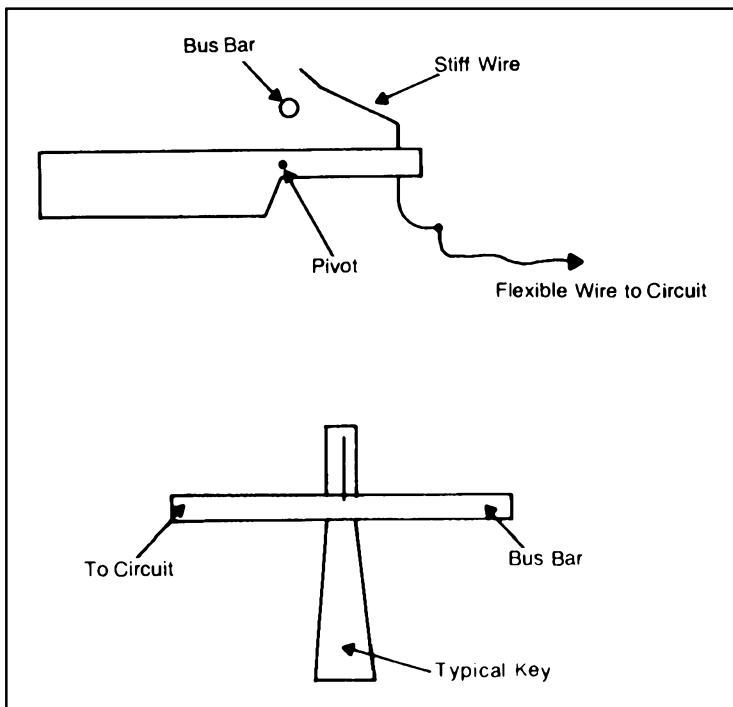


Fig. 7-9. One method of adding switches to a keyboard

chapter. The decision depends on your individual needs and how you intend to use your synthesizer.

A simple homebrew keyboard arrangement is illustrated in Fig. 7-11 and Fig. 7-12. A keyboard pattern is etched onto a printed circuit board. The common connection for the switches is connected

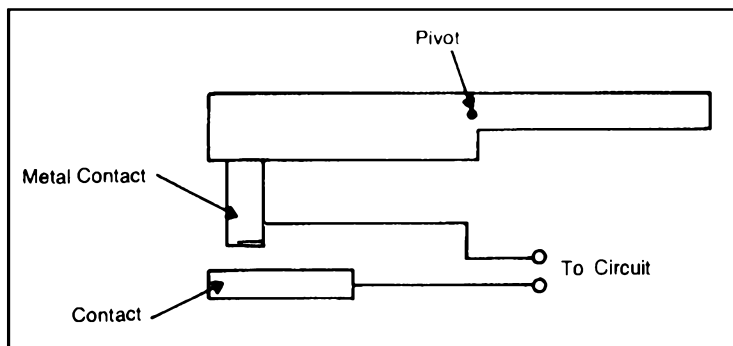


Fig. 7-10. A second method of adding switches to a keyboard.

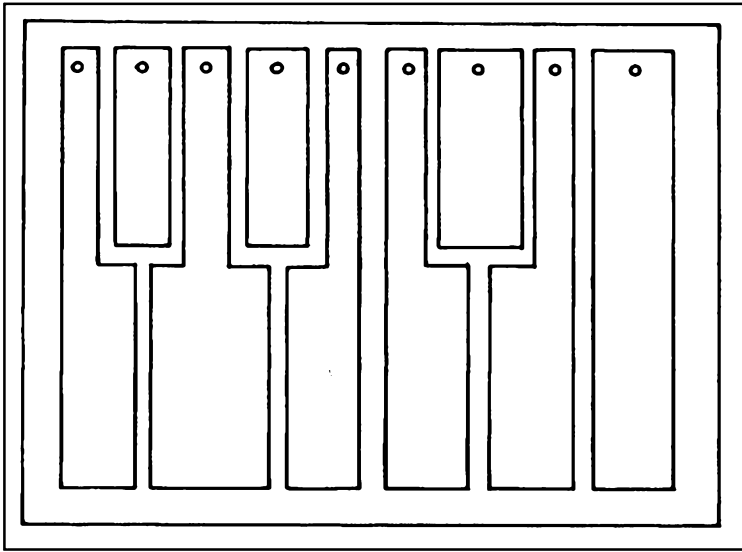


Fig. 7-11. Homebrew printed circuit keyboard.

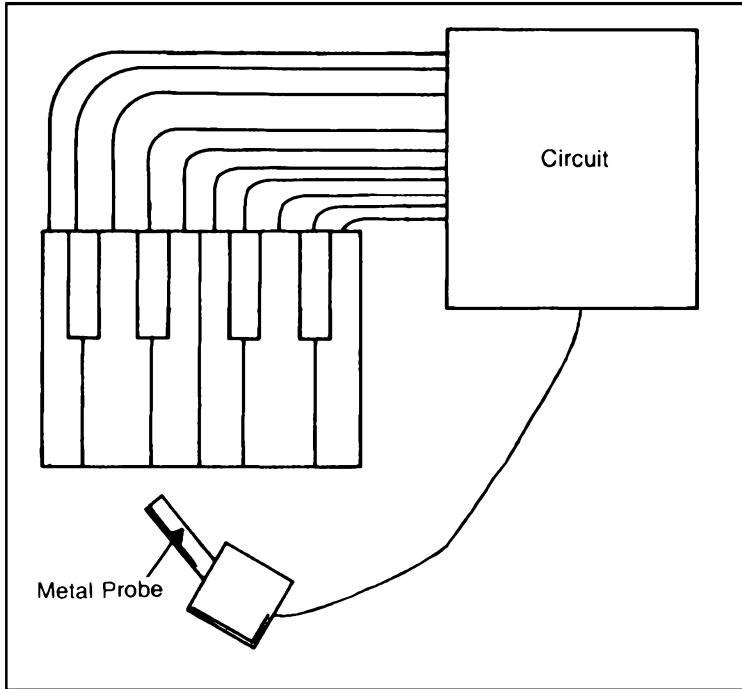


Fig. 7-12. Total printed circuit keyboard setup.

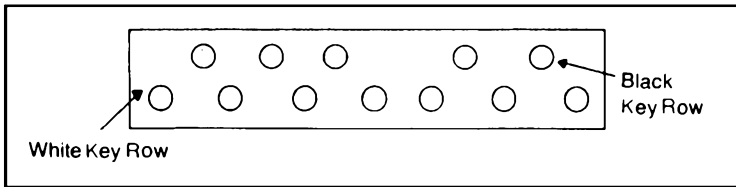


Fig. 7-13. Pushbutton keyboard.

to a probe (a mini plug, or something of that nature) at the end of a flexible wire. Touching the probe to one of the copper key images will play the appropriate note. There are two major disadvantages to this approach. First, playing the keyboard with a probe can be rather awkward. Second, only one set of switch contacts per note is practical with this system. While it is possible to design a keyboard that can be played with a double section probe to close two sets of switch contacts, but most musicians would find such an arrangement almost impossible to play.

Since a synthesizer keyboard essentially works as a series of normally open switches, a logical home brew approach would be to build a keyboard out of a bunch of NO DPST push button switches (see Fig. 7-13). You can stagger the switches to mark the position of the white keys and the black keys. For a somewhat better appearance, you can cover the switches with heavy cardboard strips that represent the keys. This is illustrated in Fig. 7-14.

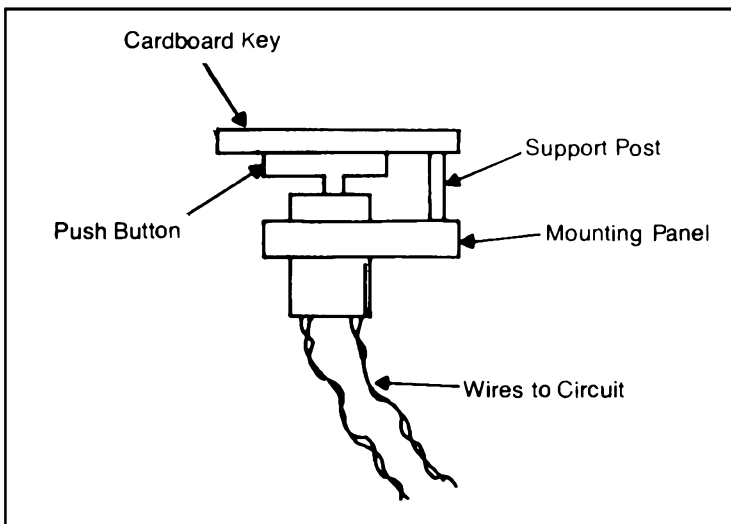


Fig. 7-14. Pushbutton keyboard with cardboard strips to simulate keys.

It is absolutely essential to use high quality switches here. Cheap push button switches have a high failure rate after a few hundred closures. That won't allow to play much music before you have to start replacing switches. Good switches are expensive, so unless you are satisfied with a very short keyboard, one of the other alternatives outlined in the section may be more practical economically.

A push button keyboard will work OK, but you must be careful in positioning the switches. They must be evenly spaced. A keyboard of this type will inevitably have an odd feel to it, which might throw some musicians.

Experimenters with advanced craftsman skills might consider carving or molding keys, then piercing them through a common pivot point as illustrated by Fig. 7-15. However, extreme accuracy must be used in the dimensions to prevent the keys from catching on one another and snagging. This method really isn't too highly recommended, unless it's the kind of thing you consider fun. Adapting a keyboard from a toy musical instrument is more practical.

OTHER CONTROLLERS

While keyboards are unquestionably the most popular form of

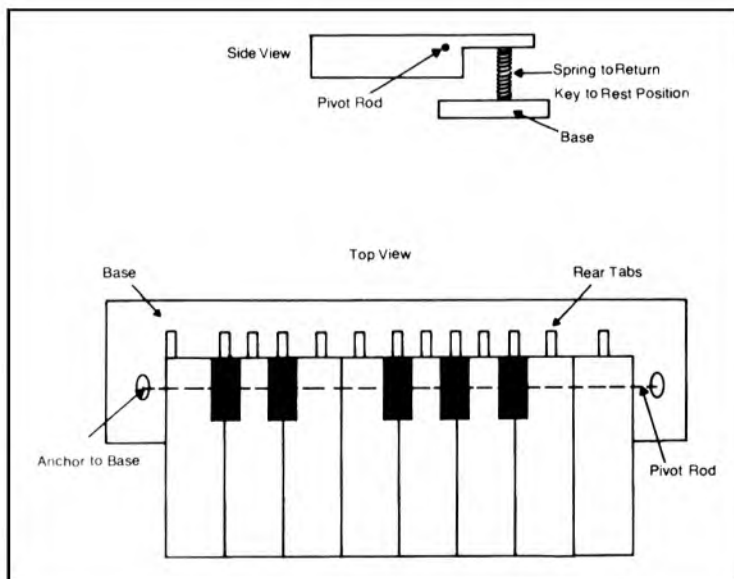


Fig 7-15 Construction of a keyboard.

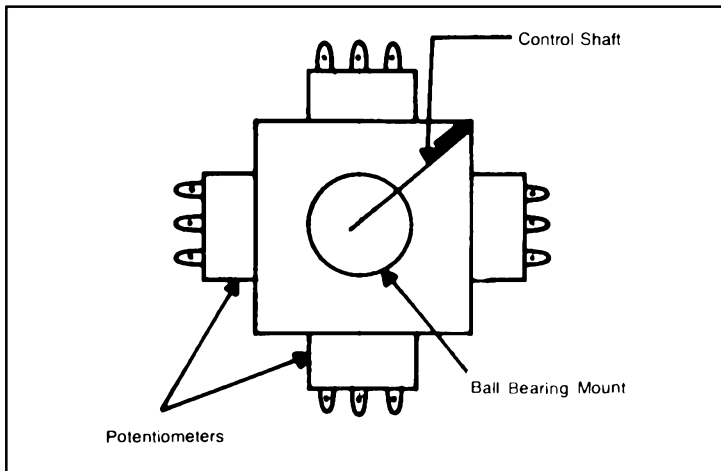


Fig. 7-16. A typical joystick.

manual control for an electronic music synthesizer, they are not the only methods available.

Traditional Methods

Many synthesizers today are driven by traditional instruments. Guitar synthesizers, driven by electric guitars were a natural development, since they already generate electrical signals similar to those used within a synthesizer. Percussionists can control synthesizers with special pressure sensitive pads that can be struck with drumsticks to produce specific effects. There are even sensors that can be placed within the bell of wind instruments so they can be used to control various synthesis functions. Microphones can convert virtually any sound into an electrical voltage that can drive a synthesizer (some kind of preamplification is usually required here).

Nontraditional Methods

But while a synthesizer can be controlled in a manner similar to a traditional instrument, it is important to remember that it is not a traditional instrument with traditional limitations. Therefore, many new manual controllers have been devised for use with electronic music synthesizers. We will examine just a few of the most common of these here.

Joysticks. *Joysticks* are often used to control synthesizer functions. These devices are simply two or four potentiometers

controlled by a single shaft in two (or occasionally three) dimensions. A typical joystick is shown in Fig. 7-16.

These joysticks are similar to those used to control video games. The control shaft may be moved up, down, right, left, diagonally, or in a circle. Each shaft position sets a different combination of resistances on the potentiometers. Figure 7-17 illustrates some typical positions of the shaft on a four potentiometer joystick.

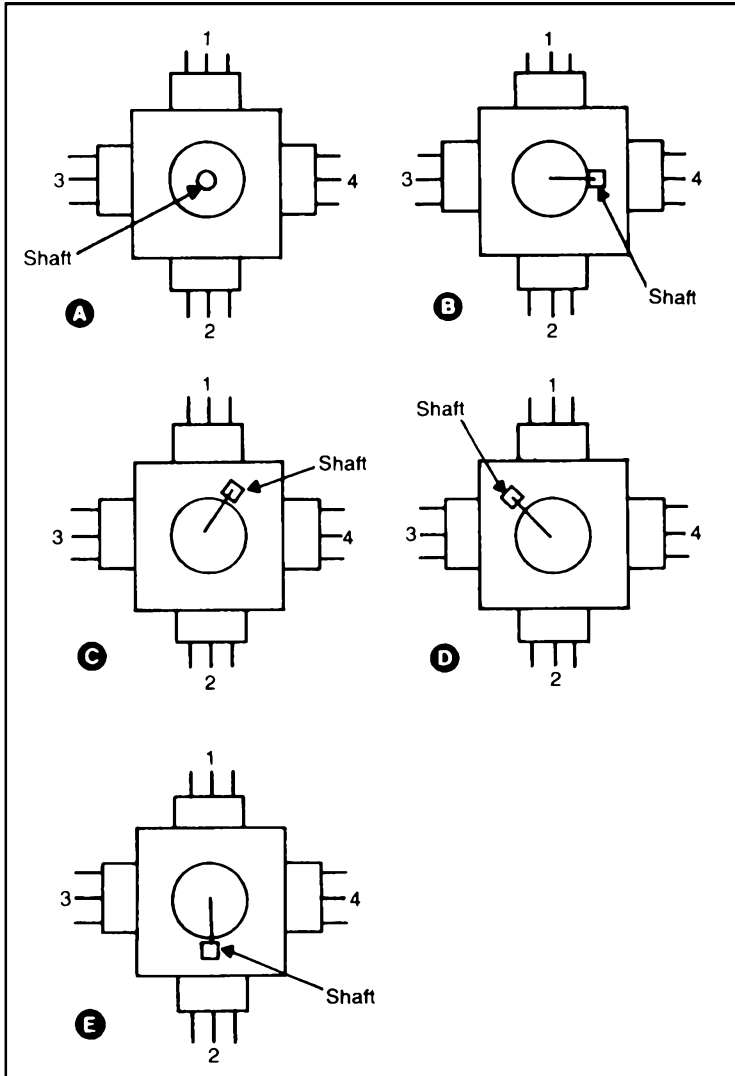


Fig 7-17. Shaft positions for a joystick.

When the shaft is at position A (dead center), all four potentiometers are set to their center positions. Moving the shaft to the right as shown at b, turns potentiometer 4 up, potentiometer 3 down, and leaves the other two potentiometers at their center settings. Position c has the shaft moved up 45° from position b. Potentiometers 1 and 4 are both set to their maximum resistances, while potentiometers 2 and 3 are at their minimum points.

In position d, the control shaft is moved all the way to the left from center (raising potentiometer 3 to its maximum) and slightly upwards (increasing potentiometer 1 from its center somewhat, and decreasing potentiometer 2 from its center setting by the same amount). Potentiometer 4 is at its minimum setting.

Finally, in position e, the control shaft is moved straight down from the center position. In this position, potentiometer 1 is at its minimum setting, potentiometer 2 is at its maximum setting, and potentiometers 2 and 3 are at their mid-points.

Notice that at any given position of the control shaft, potentiometers directly across from each other (1 and 2/3 and 4) have mirror image settings—as one increases, the other decreases.

A joystick of this type can control up to four separate synthesis functions. In most cases, similar functions are controlled by the joystick to prevent excessive confusion. Operating a joystick with divergent functions, or where precision is required, can be quite difficult.

A popular application that is well suited to the joystick is to control four vcas, each driving one channel of a quadraphonic (four speaker) sound system or recorder. The joystick can be used to place the apparent sound source anywhere within the sound field between the speakers (see Chapter 4). The position of the joystick control shaft corresponds to the location of the apparent sound source within the field.

In synthesizer work, one end of each potentiometer in a joystick is connected to a fixed voltage source. Moving the control shaft changes the amount of voltage that will appear at the other end of each potentiometer. Obviously this variable voltage can be used to operate any voltage-controlled synthesis module.

Joysticks can be nice for certain special effects, but in most cases, they are difficult to relate to, and almost impossible to set precisely. Joysticks certainly shouldn't be the primary manual control device for your synthesizer system, but they are good for supplemental controllers.

Ribbon Controllers. A number of commercially available

synthesizers feature *ribbon controllers*. A ribbon controller is simply a long, narrow ribbon of a special material that will produce a resistance that is proportional to where along its length it is touched. The ribbon controller can be used in a similar fashion as a keyboard. The electronic musician can slide his finger along the ribbon to produce smooth glides from frequency to frequency (or whatever parameter is being controlled) at any speed he chooses. Microtones (in-between notes) that are not possible on a traditional keyboard can be readily played on a ribbon controller.

A switch can be added along the entire length of the ribbon, so a gate or trigger pulse can be generated whenever it is touched.

Ribbon controllers are nice, although they are not as easy to play as a traditional keyboard. It is also difficult to notate effects played on a ribbon controller. On the other hand, the unfamiliarity of a ribbon controller is not as likely to limit the electronic musician's imagination as a traditional keyboard.

Unfortunately for the experimenter building a custom electronic music synthesizer, the conductive ribbon material is not readily available, especially in the small quantities needed by the experimenter. However, you may be able to find some in one of the surplus house catalogs, or you may be able to salvage a strip from a junked synthesizer.

Foot Pedals. One of the big advantages of an electronic music synthesizer is the large number of parameters of a sound that can be controlled by the musician. This can also be something of a disadvantage, especially in a live performance situation. After all, the average musician only has two hands to play a keyboard, or ribbon controller, flick switches, or twiddle potentiometers or joysticks.

To increase the amount of control an electronic musician has, foot pedals can be used to control some functions.

Two types of foot pedals are available. One type is nothing more than a foot operated switch for on/off effects. The other type is a foot controlled potentiometer. This second type is illustrated in Fig. 7-18. In an electronic music synthesizer, this kind of foot pedal is generally used with a fixed voltage source to produce a varying voltage for voltage-control.

Of course, it is difficult for most people to precisely position their foot on a control pedal, but this control method is just fine for relatively crude effects. Foot pedals are often used to control volume, or the depth of vibrato or tremolo effects. Figure 7-19 shows a patch that can be used for foot controlled vibrato depth and

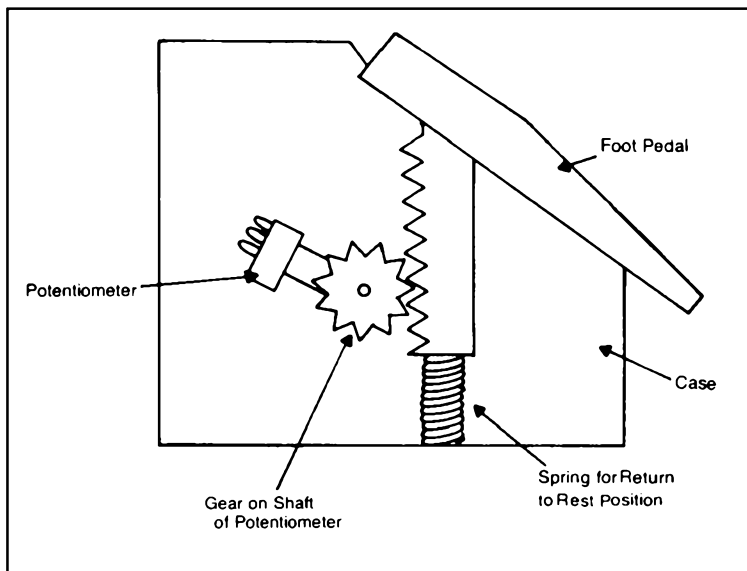


Fig. 7-18. Potentiometer foot pedal.

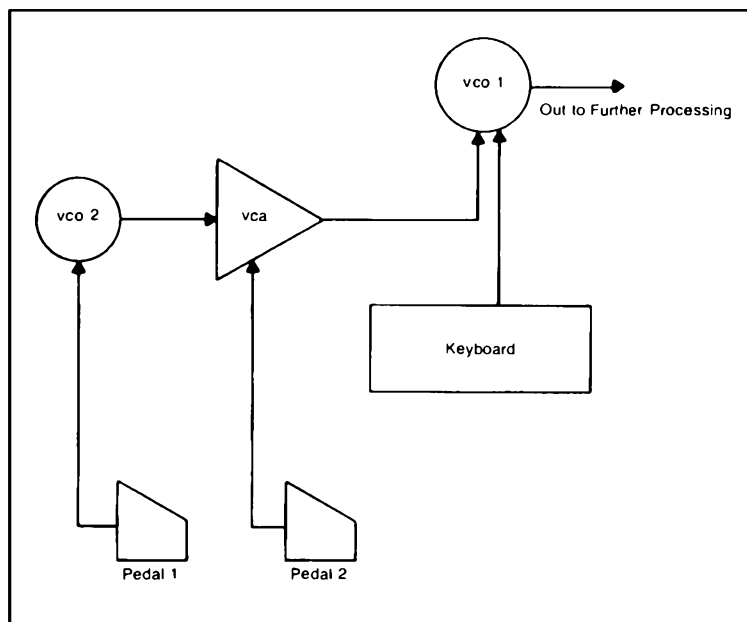


Fig. 7-19. Patch diagram for foot controlled vibrato speed and depth.

speed. Two foot pedals are used—one for each foot. Naturally it is assumed that the musician is seated.

Foot pedal control is not unique to electronic music synthesizers. Many organs have pedals to control the volume of the instrument. Large organs often have a keyboard of foot pedals to play bass notes. Electric guitars have popularized a number of special effects devices that are usually housed in foot pedals since both of a guitarist's hands are occupied. Even standard acoustic pianos have loudness pedals that work in on/off fashion.

Moog even makes a synthesizer called Taurus which is primarily controlled by foot pedals and foot operated switches. The musician's hands are free to play another instrument simultaneously.

Mercury Switches. Many unique control techniques can be concocted using mercury switches. A mercury switch is essentially a glass tube containing two contacts and a globule of mercury. When the tube is tilted so that the mercury is touching both of the contacts, the switch is closed. In other positions (when the mercury is touching just one or neither of the contacts), the switch is open (see Fig. 7-20).

Obviously mercury switches are suitable only for on/off effects, rather than continuously varying control voltages like a potentiometer. Still, there are many applications.

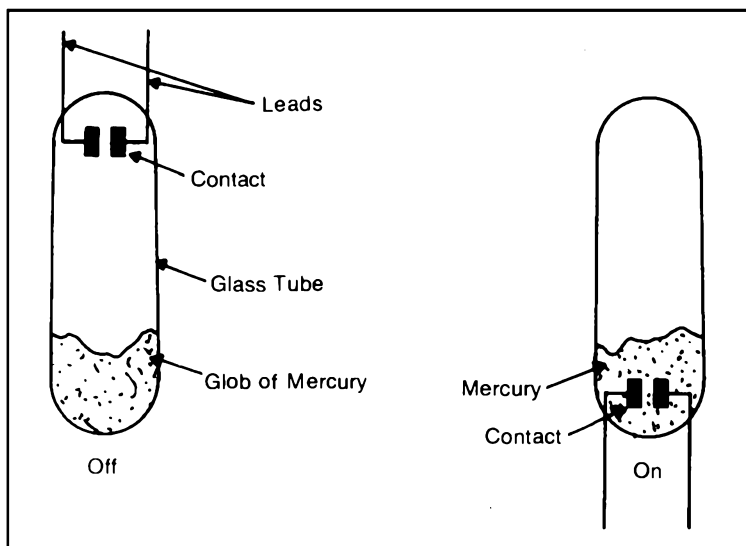


Fig 7-20 Construction of a mercury switch.

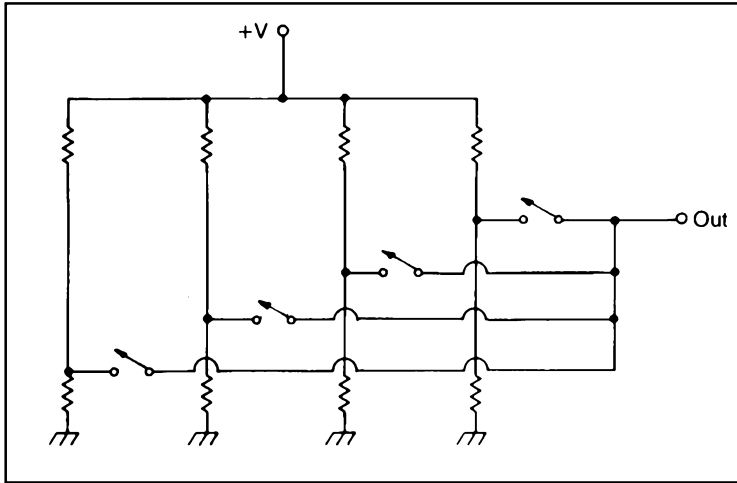


Fig. 7-21. Switched stepped-voltage control

Remember, a keyboard is just a series of switches too. A bank of mercury switches could be connected to generate stepped control voltages using a circuit something like the one in Fig. 7-21.

Mercury switches are controlled by positioning them. Many unorthodox arrangements could be used. It would probably be rather tricky learning how to control your synthesizer using mercury switches, but it can be done. Of course, pseudo-random effects would be the easiest to create this way.

A number of mercury switches could be mounted on a panel that can be turned, rolled, or tossed about to open and close various switches. One possible arrangement of the switches on such a panel is illustrated in Fig. 7-22.

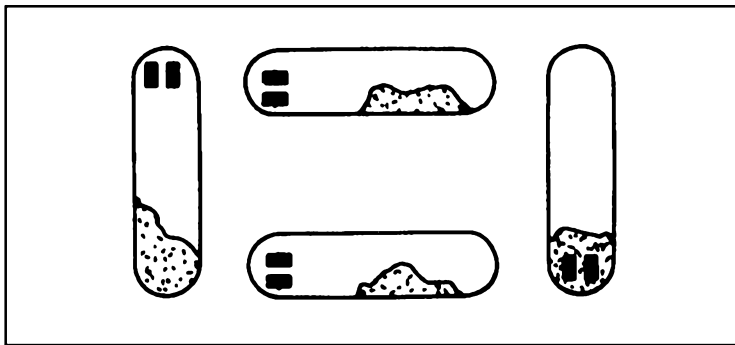


Fig. 7-22. Possible mercury switch controller.

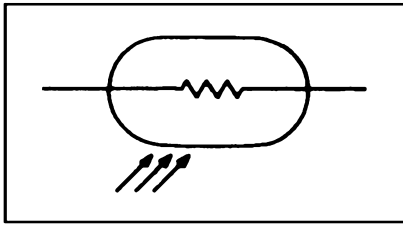


Fig. 7-23. Photoresistor symbol.

A dancer could have several mercury switches sewn to his or her costume. Each time the dancer moved, the synthesizer would generate different sounds. Of course, you would have to be careful about the cables running from the synthesizer to the switches on the dancer's costume. Advanced experimenters might want to couple this idea with a small radio transmitter that is also attached to the dancer's costume and a receiver for the synthesizer.

For really strange random effects, how about connecting mercury switches to a mobile or a set of wind chimes? This would give you a wind controlled synthesizer.

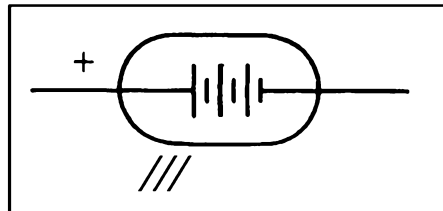
There are many other possibilities for using mercury switches to control an electronic music synthesizer. These techniques are undeniably off the wall, but fun to work with, and they can often produce some very striking effects you might not come up with by other methods.

Light Activated Controllers. Current interest in solar power technology has made a number of light sensitive devices available to the experimenter. These can be used to control your electronic music synthesizer too.

There are two primary ways to play a light sensitive synthesizer. You can either place the controller in a well lighted area and create varying shadows on its surface by waving your hands over it, or you can place the controller in a darkened area, and shine a flashlight or some other small, movable light source on the controller's surface.

There is some confusion about the names of these light sensitive devices. The term "photocell" is often applied to photoresis-

Fig. 7-24. Photocell (photobattery) symbol.



tors, A *photoresistor* is obviously a device that varies its resistance in proportion to the amount of light striking its surface. Another name for this device is light dependant resistor or LDR. The schematic symbol for the photoresistor is shown in Fig. 7-23.

The term photocell is more properly applied to a segment of a photobattery (or solar battery). A *photobattery* generates a current proportional to the amount of light striking its surface. The schematic symbol for this device is illustrated in Fig. 7-24.

Photobatteries are primarily used to power circuits with low voltage requirements. They are not particularly suited to controlling synthesizer modules since their output voltage is more or less constant. Only their current handling capability varies with the light intensity.

Photoresistors, on the other hand, can be very useful to the electronic musician. Since they are a varying resistance device, they can be used almost anywhere a potentiometer is used.

By using a fixed voltage source (a photobattery, for instance) and a photoresistor, a light controlled variable voltage source can easily be built. A schematic is shown in Fig. 7-25.

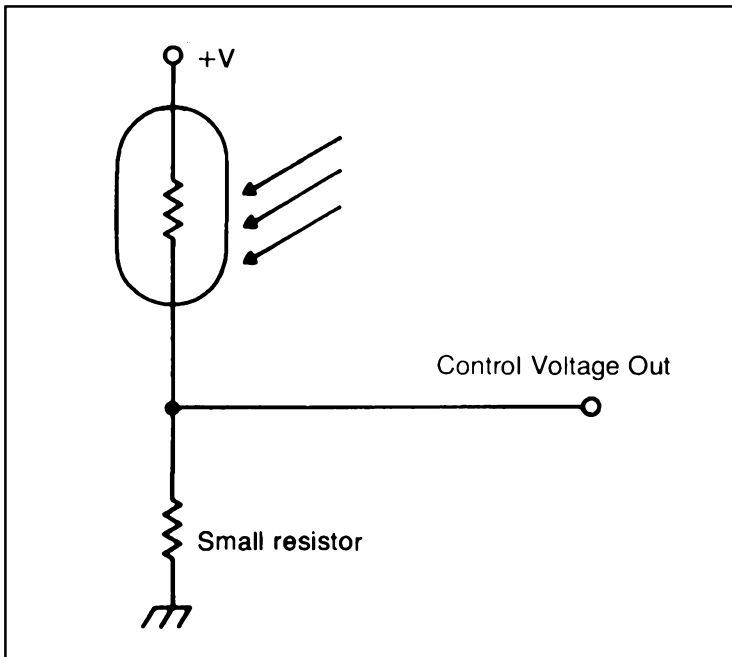


Fig. 7-25. Photoresistor control voltage source.

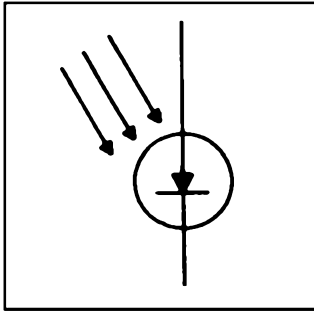


Fig. 7-26. Photodiode symbol.

Other light sensitive devices are also available, and I am not going to devote a lot of space to their possible applications here. There are too many possibilities. Use your imagination.

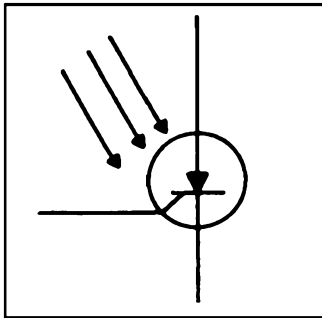


Fig. 7-27. LASCR symbol.

The schematic symbol for a photodiode is shown in Fig. 7-26. Figure 7-27 shows the schematic symbol for a LASCR or light activated silicon controlled rectifier. The schematic symbol for a

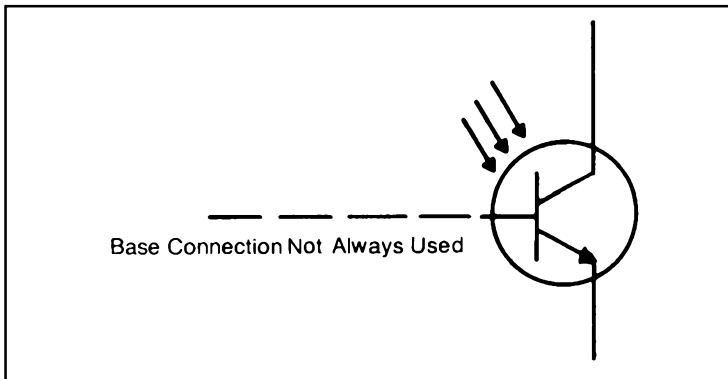


Fig 7-28 Phototransistor symbol.

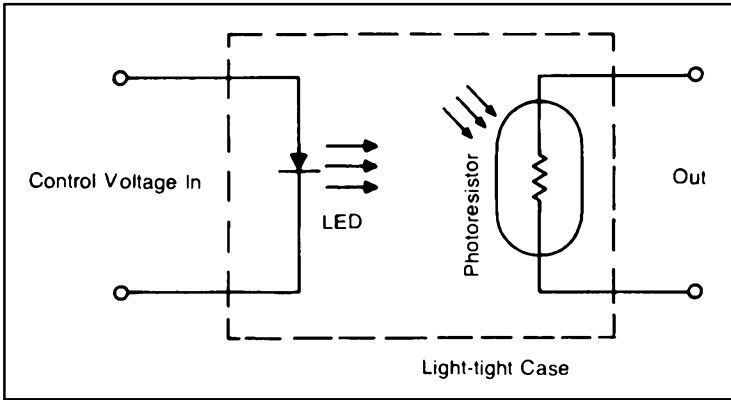


Fig. 7-29. Construction of a homebrew optoisolator.

phototransistor is illustrated in Fig. 7-28. There is often no direct connection to the base of a phototransistor. The light striking the device's surface determines the effective base voltage.

You can use light sensitive devices to build voltage control capabilities into almost any circuit. For example, a potentiometer or resistor can be replaced by a photoresistor. A small lamp or LED (light emitting diode) is positioned to shine on the face of the photoresistor, as shown in Fig. 7-29. Both of these components should be shielded from all outside light. Different voltages applied to the lamp (or LED) will vary the brightness of the light shining on the photoresistor, changing its resistance and the operating parameters of the circuit. The same idea can be used with photodiodes, photoresistors, or LASCRs.

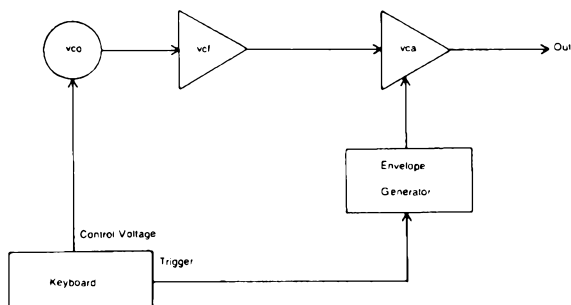
Commercially available devices based on this idea are available. These units are called *optoisolators*. An LED and a light sensitive device are encapsulated in a light-tight package resembling an 8-pin DIP IC

One big advantage of the optoisolator idea is that the control circuit is isolated electrically from the circuit it is controlling. This prevents loading, impedance mismatches, and other potential problems.

Your Own Ideas. There are many, many other ways to manually control your electronic music synthesizer. Many haven't even been thought of yet. Use your own ideas. The possibilities are almost limitless, and no idea is too far out.

Anything that opens and closes a switch, changes a resistance, or generates a varying voltage can be put to use in your synthesis system. Let your imagination flow, and experiment.

Chapter 8



Automatic Controllers

In electronic music synthesis, many parameters of each sound can be dynamically varied. For example, pitch, harmonic content, timbre, and volume, to name just a few. Unfortunately, there is an obvious physical limitation to the number of controllers a musician can operate at one time. To take full advantage of your synthesizer's capabilities, you will need to control some of the sound parameters with automatic devices.

In some patches, automatic controllers may operate all of the parameters, letting the instrument essentially play itself.

There is another major advantage to using automatic controllers. An electronic circuit can generate and change a control voltage far faster and with much greater accuracy than any human being could. This makes many special effects possible. Can you imagine having to set timbre (the amplitude envelope—see Chapter 6) manually for each and every note?

In this chapter, we will examine a number of automatic controller circuits of various types. Once again, we will only be able to cover a few of the more common circuits. As with so many other things in electronic music synthesis, the possibilities are virtually limitless.

LOW FREQUENCY OSCILLATORS

An oscillator can be used to control a voltage-controlled module. An audio range oscillator can be used to control a vco for FM

effects or a vca for AM effects. It can also be used to control a vcf. The effect will be a sort of cross between AM and FM.

Generally, however, oscillators used as automatic controllers have subaudible frequencies. A circuit used this way is called a low-frequency oscillator, or lfo. An lfo may be voltage-controlled, but in most cases the frequency is manually set.

If the frequency of an lfo is slow enough, its waveshape will be audible. Most commercial synthesizers have lfos that generate sine waves. The smooth, gliding shape of this waveform makes it perfect for effects like tremolo (see Chapter 6) or vibrato (see Chapter 5). If a sine wave oscillator is not available, triangle waves may be substituted. For best results when using triangle waves, use a low-pass filter to remove the harmonics.

Unusual effects can be achieved by controlling a voltage-controlled module with a lfo that generates sawtooth waves or rectangle waves.

In most cases, when a lfo is referred to, it is assumed to be a sine wave type unless otherwise noted.

For circuits that can be used in this application, refer back to Chapter 3.

ENVELOPE GENERATORS

Envelope generators (sometimes called function generators) were discussed in Chapter 6.

To review, there are two basic types. The AR (Attack/Release) type allows the synthesist to determine how long the voltage will take to build up to its maximum level, and how long it will take to die down. As long as there is a voltage at the envelope generator's trigger input after the attack portion is completed, the maximum voltage level will be sustained. If the trigger signal ends before the attack portion is completed, the release (or decay) will begin immediately after the attack.

Some typical AR envelopes are illustrated in Fig. 8-1. Figure 8-2 shows how the control panel of an AR envelope generator might look.

A more deluxe kind of envelope generator is the ADSR type. This type of envelope generator produces four stage envelopes. First there is an attack (as with the AR type), which is immediately followed by an initial decay which drops the output voltage down from its maximum level to a manually set sustain level. The sustain is again held for as long as the trigger signal is present. Then the final release lets the voltage drop back down to zero.

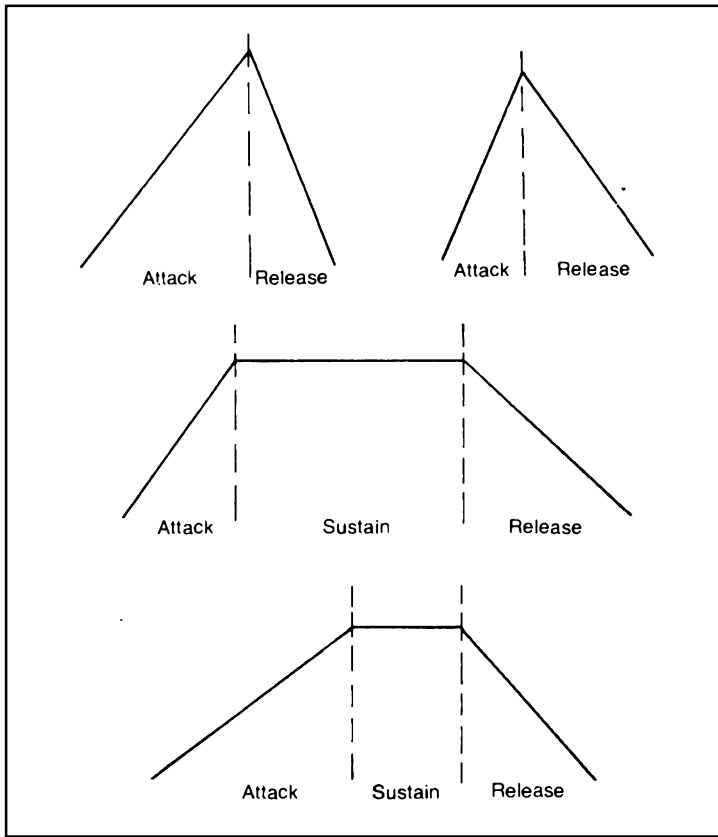


Fig. 8-1. AR envelopes.

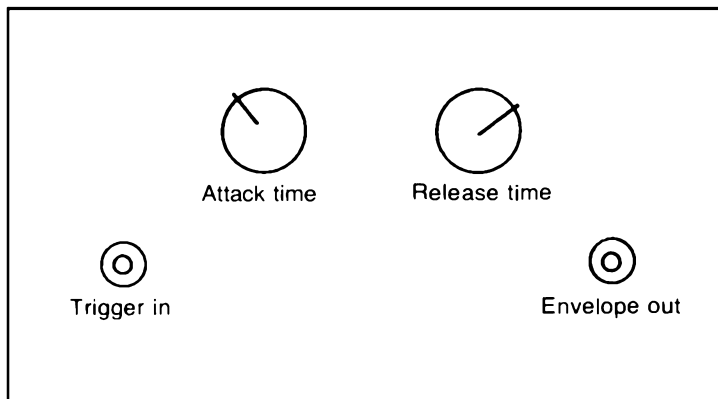


Fig. 8-2. Control panel of an AR envelope generator.

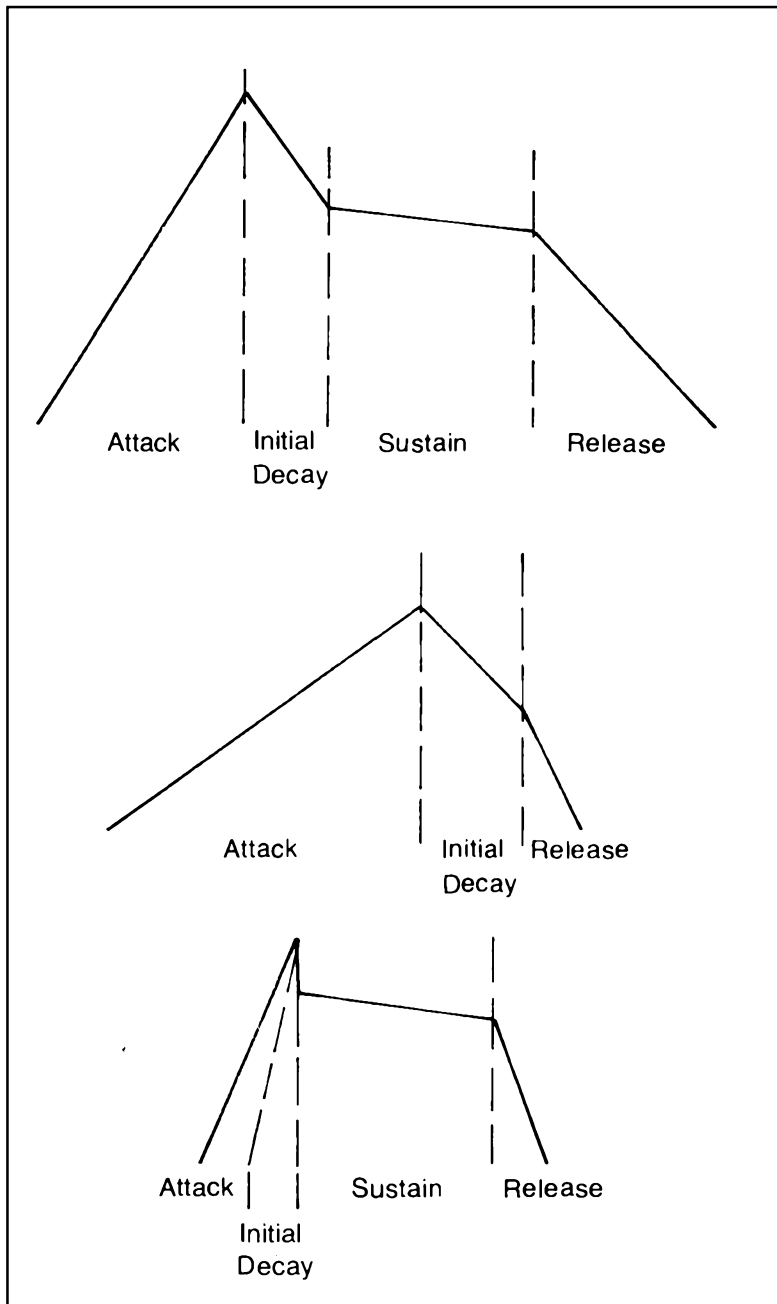


Fig. 8-3. ADSR envelopes.

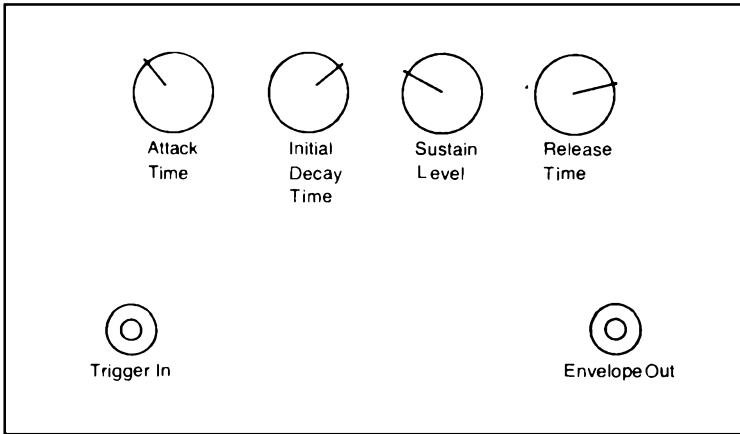


Fig. 8-4. Control panel of an ADSR envelope generator.

Some typical ADSR envelopes are shown in Fig. 8-3. The control panel of a typical ADSR envelope generator is illustrated in Fig. 8-4.

Circuits

Envelope generators are most frequently used to control v_{cs} to set the timbre of the sound, as discussed in Chapter 6. However, like any other voltage source, they may be used to operate any

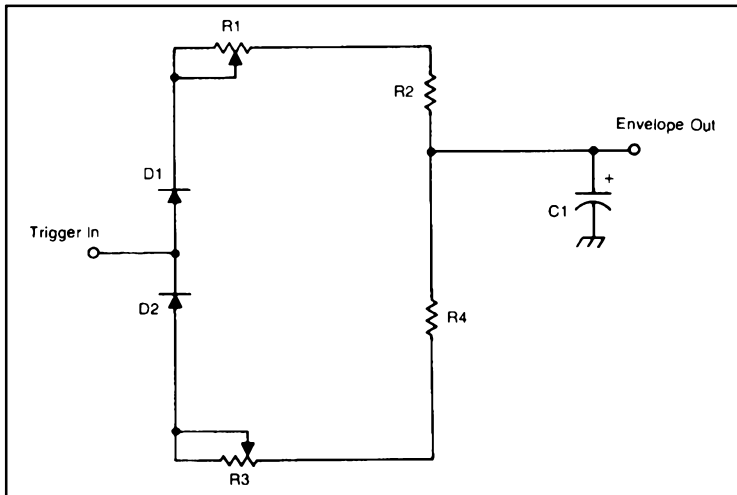


Fig. 8-5. Simple AR envelope generator.

Table 8-1. Parts List for Fig. 8-5.

R1, R3	1 M Ω potentiometer
R2, R4	47 k Ω resistor
C1	2.2 μ F electrolytic capacitor
D1, D2	diode (1N4148 or similar)

voltage-controlled module. Some very interesting effects can be achieved by controlling a vco or a vcf with an envelope generator.

Simple AR Type. A simple AR type envelope generator is shown in Fig. 8-5. This is a passive circuit, taking its power from the trigger signal. It is just about as simple as an envelope generator circuit can get, requiring only 7 components. The parts list is given in Table 8-1. Potentiometer R1 controls the attack time, while potentiometer R3 is used to adjust the release time. The maximum level will be sustained as long as the trigger signal is present.

One disadvantage of this circuit is that only gate signals can really be used to trigger it. Trigger pulses are too short. The trigger

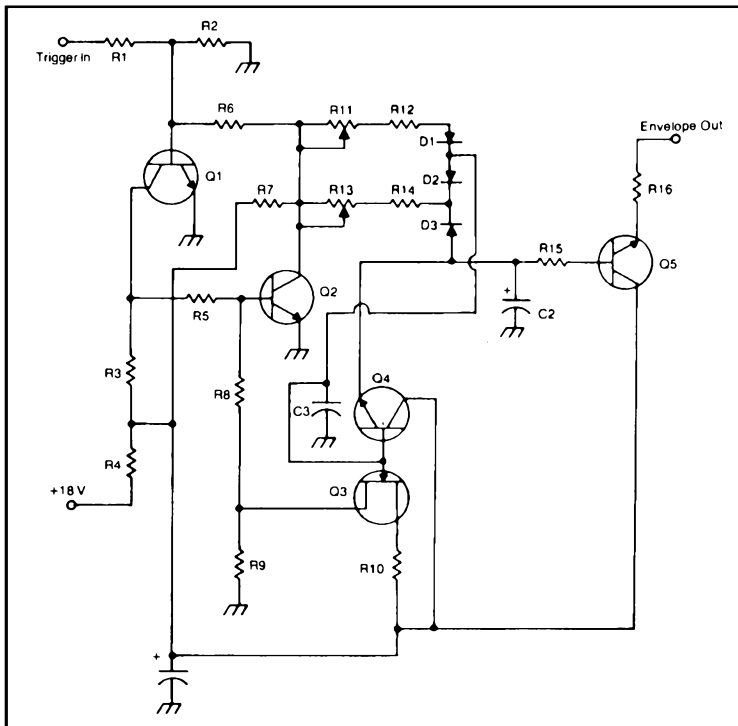


Fig. 8-6. Improved AR envelope generator.

signal must last at least as long as the attack portion of the envelope. If the trigger signal ends before the attack is completed, the envelope will immediately begin to decay.

Five Transistor AR Type. A more advanced AR envelope generator circuit is shown in Fig. 8-6. This circuit requires a 15 to 18 volt power supply. The parts list is given in Table 8-2. Potentiometer R11 controls the attack time, while R13 sets the release time.

Resistor R16 could be replaced by a potentiometer for a manually variable sustain level. This potentiometer should have a value between 1 k Ω , and 10 k Ω .

Either gate or trigger pulses may be used to drive this envelope generator.

555 Timer ADSR Type. Figure 8-7 shows the circuit for an Al)SR type envelope generator. The parts list is given in Table 8-3.

This envelope generator is a bit unusual, since the sustain time can be set, rather than depending on the length of the gate signal. The circuit is built around a pair of 555 timers (a single 556 dual timer IC may be substituted, of course). IC2 determines the attack time, while IC3 sets the length of the sustain portion of the envelope.

Potentiometers R4 and R7 interact in setting the attack time. The initial decay is set with potentiometer R3. The portion of the

Table 8-2. Parts List for Fig. 8-6.

R1	47 k Ω resistor
R2	330 k Ω resistor
R3, R7	6.8 k Ω resistor
R4	330 Ω resistor
R5, R6	100 k Ω resistor
R8	33 k Ω resistor
R9	47 Ω resistor
R10	6.2 k Ω resistor
R11, R13	500 k Ω potentiometer
R12, R15	10 k Ω resistor
R14, R16	1 k Ω resistor
C1	100 μ F electrolytic capacitor
C?	2.5 μ F electrolytic capacitor
C3	0.2 μ F capacitor
D1, D2, D3	silicon diode (1N4148 or similar)
Q1, Q2	NPN transistor (2N744, 2N4264, 2N5129, GE-20, RCA SK3020, Motorola HEP53 or similar)
Q3	UJT (Radio Shack RS2029 or similar)
Q4, Q5	NPN transistor (2N2712, Radio Shack RS2015, GE-20, RCA SK3020, or similar)

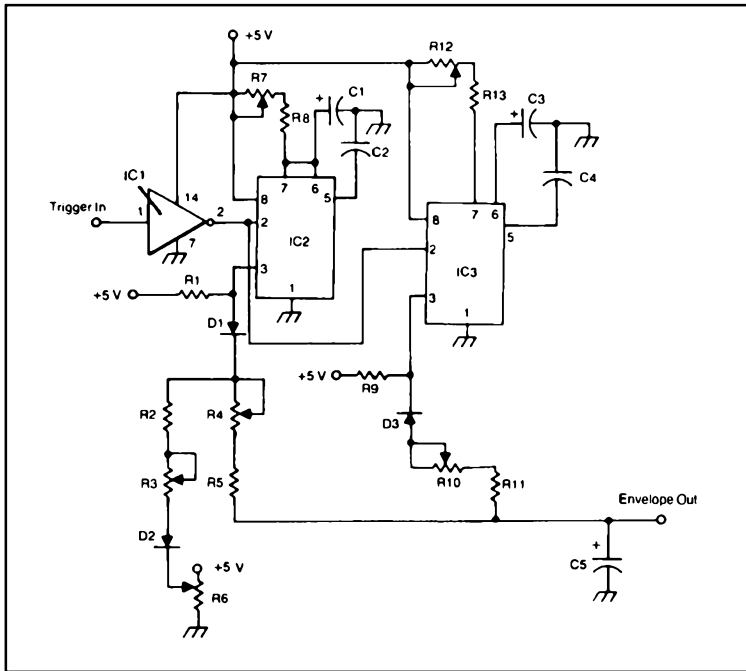


Fig. 8-7. 555 ADSR envelope generator.

circuit including R2 through R5, D1, and D2 is actually the same basic circuit that was presented in Fig. 8-5.

The sustain time is controlled by potentiometer R12, and the sustained level is determined by the setting of potentiometer R6. Finally R10 determines the length of the final release section of the envelope.

Table 8-3. Parts List for Fig. 8-7.

R1, R9	2 2 k Ω resistor
R2, R5	10 k Ω resistor
R3	50 k Ω potentiometer
R4, R7, R10, R12	1 M Ω potentiometer
R6	1 k Ω resistor
R8, R13	22 k Ω resistor
R11	47 k Ω resistor
C1, C3, C5	1 μ F electrolytic capacitor
C2, C4	0.01 μ F capacitor
D1, D2, D3	silicon diode (1N4148 or similar)
IC1	1/6 of a 7404 hex inverter IC
IC2, IC3	555 timer IC

IC1 is a digital inverter. It is actually one section of a six section hex inverter IC. It is not strictly essential, but it is helpful for interfacing this circuit with other synthesizer modules. If the inverter is used, a 5 volt (approximately) pulse will trigger the envelope. Without the inverter, you would have to feed a constant 5 volts into the trigger input and interrupt it (drop the voltage to zero) to initiate an envelope.

Combined Generator and vca. The circuit shown in Fig. 8-8 is a bit different from the other envelope generators discussed in this section. Ordinarily, the audio signal does not pass through an envelope generator. The envelope generator is triggered to produce a voltage that varies in a specific pattern. This voltage is generally used to control a separate vca module, although envelope generators may be freely used with any voltage-controlled module. The circuit of Fig. 8-8, on the other hand, is self contained. It is essentially an envelope generator and a vca in one. It can only be used to add an amplitude envelope over an existing signal. This circuit has both a trigger input and a signal input. There is only a signal output. The envelope pattern is not available as an external control voltage.

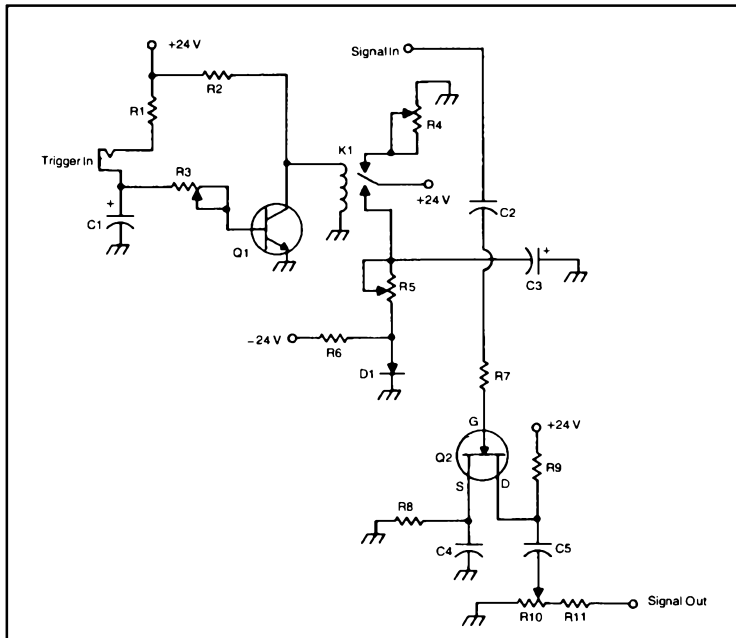


Fig. 8-8. Single module envelope circuit.

Table 8-4. Parts List for Fig. 8-8.

R1	1.5 k Ω resistor
R2	4.7 k Ω resistor
R3	100 k Ω potentiometer
R4, R5	500 k Ω potentiometer
R6	1 k Ω resistor
R7	1 M Ω resistor
R8	2.2 k Ω resistor
R9	10 k Ω resistor
R10	1 M Ω potentiometer
R11	100 k Ω resistor
C1, C4	22 μ F electrolytic capacitor
C2, C5	0.1 μ F capacitor
C3	10 μ F electrolytic capacitor
D1	zener diode (1N749 or equivalent)
Q1	NPN transistor (MPS2926, Motorola HEP-722, Radio Shack RS2016, or similar)
Q2	FET (2N5459, Radio Shack RS2036, or similar)
K1	4.5 volt SPDT relay

Potentiometer R4 controls the attack time, potentiometer R3 controls the sustain time, and potentiometer R5 sets the release time. There is no initial decay in the envelopes generated by this circuit. The amplitude of the sustain portion of the envelope (the envelope's maximum level) is determined by the setting of potentiometer R10.

The parts list for this circuit is given in Table 8-4.

Auto Pulser. A circuit that may supplement the envelope circuit of Fig. 8-8 is illustrated in Fig. 8-9. This circuit generates a string of trigger pulses for repeating envelopes as long as switch S1 is closed. Potentiometer R4 sets the repetition rate for the envelope cycles. You should be aware that for long envelopes, the automatic trigger pulses may occur before the previous envelope is completed. This could cause problems. On the other hand, it might create just the effect you need. Experiment.

The parts list for the auto-pulser circuit is given in Table 8-5.

CEM 3310 IC As with many of the other synthesis modules already discussed in this book, dedicated envelope generator ICs are available.

The pin-out diagram for Curtis Electromusic Specialties' CEM 3310 envelope generator is shown in Fig. 8-10. This is a full ADSR type envelope generator. A unique feature of this device is that each of the envelope parameters (attack time, initial decay time, sustain level, and final release time) may be voltage-controlled for dynamically changing envelopes.

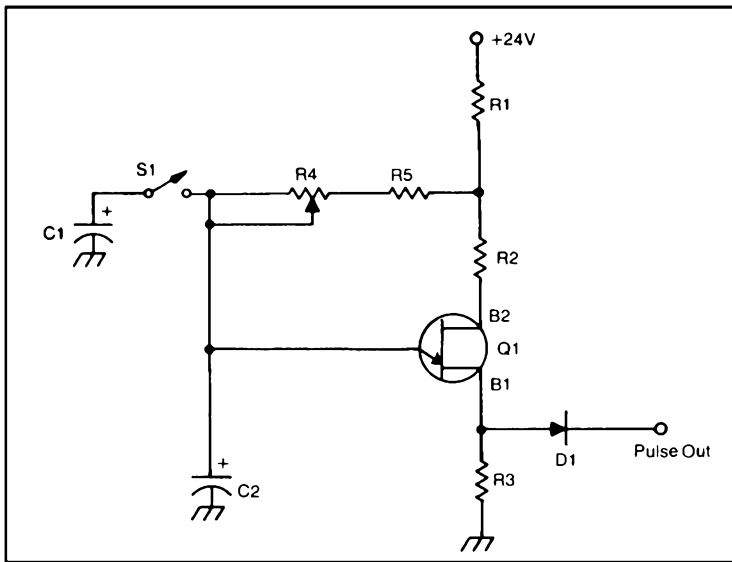


Fig. 8-9. Auto-pulsar.

SSM 2055 IC Solid State Micro Technology also manufactures several voltage-controlled envelope generator ICs. The pin-out diagram for the SSM2055 is shown in Fig. 7-11. The times may be set through a 1 to 50,000 range, which gives the device an amazing degree of versatility.

Combining Envelope Generators. Envelope generators are fairly simple devices, but they are extremely useful in electronic sound synthesis.

For greater versatility in generating envelope shapes, two or more envelope generators may be used together. Figure 8-12, shows one way two envelopes could be combined in parallel. The patch for this effect is illustrated in Fig. 8-13.

Table 8-5. Parts List for Fig. 8-9.

R1	2.2 k Ω resistor
R2	560 Ω resistor
R3	100 Ω resistor
R4	50 k Ω potentiometer
R5	3.3 k Ω resistor
C1	250 μ F electrolytic capacitor
C2	22 μ F electrolytic capacitor
D1	silicon diode (1N4148, or similar)
Q1	UJT (GEX10, Radio Shack RS2031, or similar)
S1	SPST switch

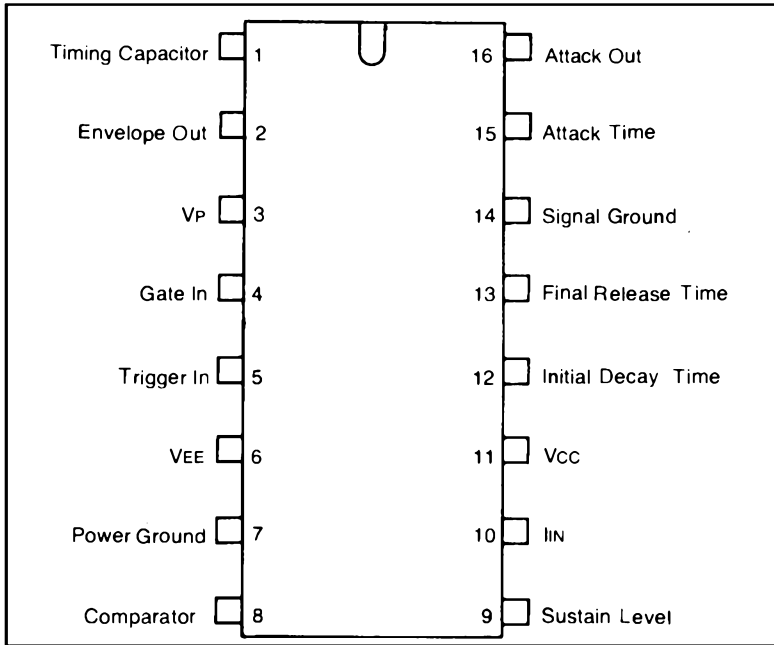


Fig. 8-10. Pin-out diagram for the CEM 3010 envelope generator IC.

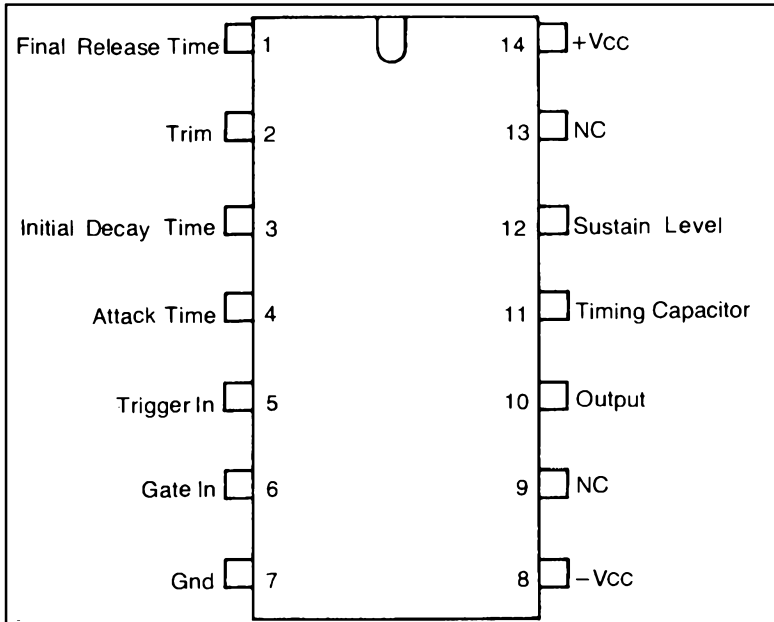


Fig. 8-11. Pin-out diagram for the SSM 2055 envelope generator IC.

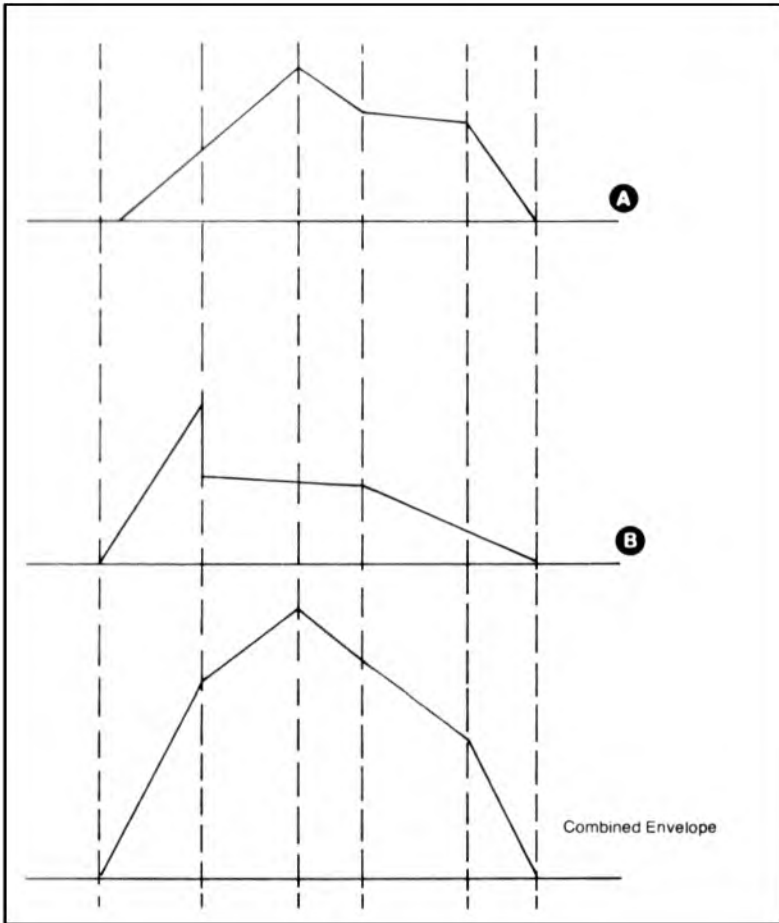


Fig. 8-12. Combining envelopes in parallel.

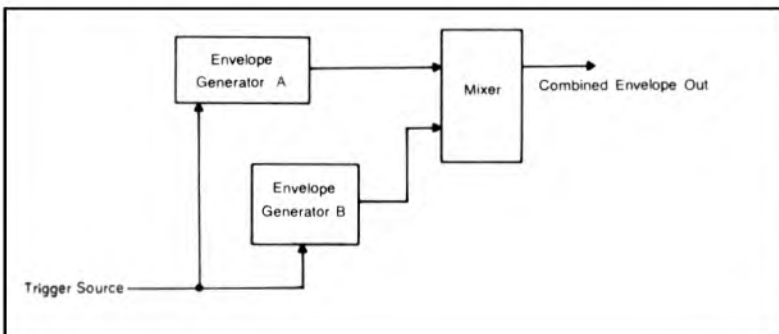


Fig. 8-13. Patch diagram for parallel envelopes.

Another way to combine envelopes is to let one envelope trigger the next when its level passes a specific point. A typical example of this is shown in Fig. 8-14. The patch for creating this effect is shown in Fig. 8-15.

Envelope Followers

In electronic music, envelopes are usually obtained by envelope generators of some sort. However, when you are combining a microphone, or another musical instrument with a synthesizer, you can take the natural envelope from the original sound, and use it with another, purely electronic sound. Many unique and fascinating

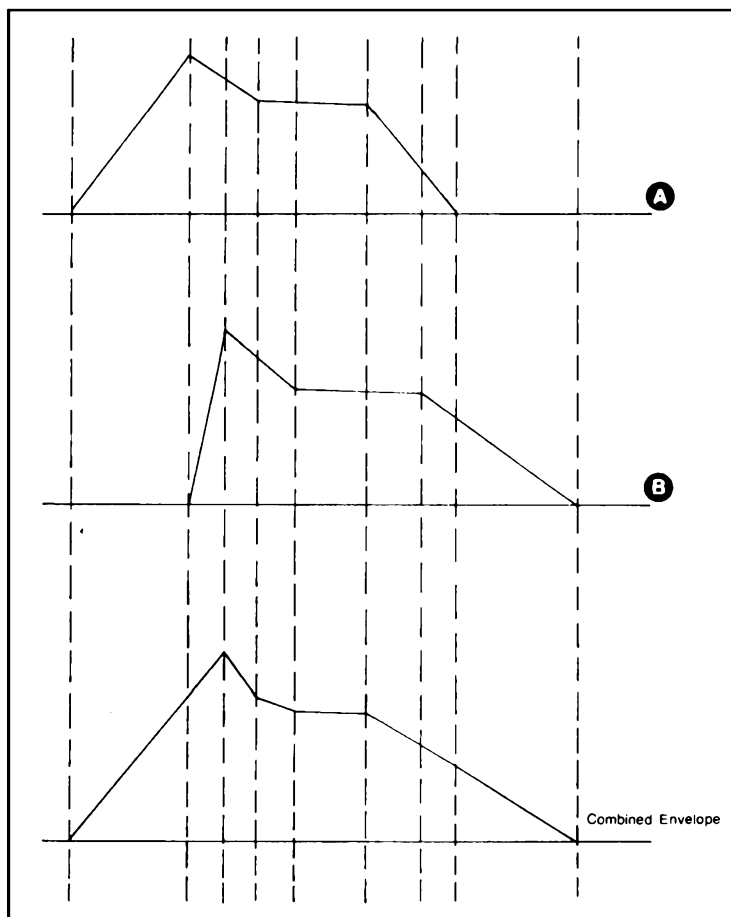


Fig. 8-14. Combining envelopes in series.

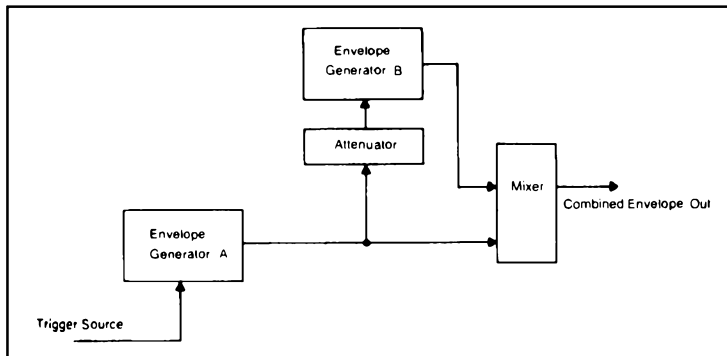


Fig. 8-15. Patch diagram for series envelopes.

effects can be created using this technique. For example, by speaking or singing into a microphone and using the envelopes from your voice on a vco's output, you can make the oscillator seem to talk. The clarity of the effect will depend on the exact equipment used.

The circuit that strips off the envelope of a signal is called an *envelope follower*. The patch diagram for the speaking oscillator effect is shown in Fig. 8-16.

A circuit for a basic envelope follower is shown in Fig. 8-17. The parts list is given in Table 8-6.

An envelope follower may be used in any of the ways you can use an envelope generator. While most commonly used to drive

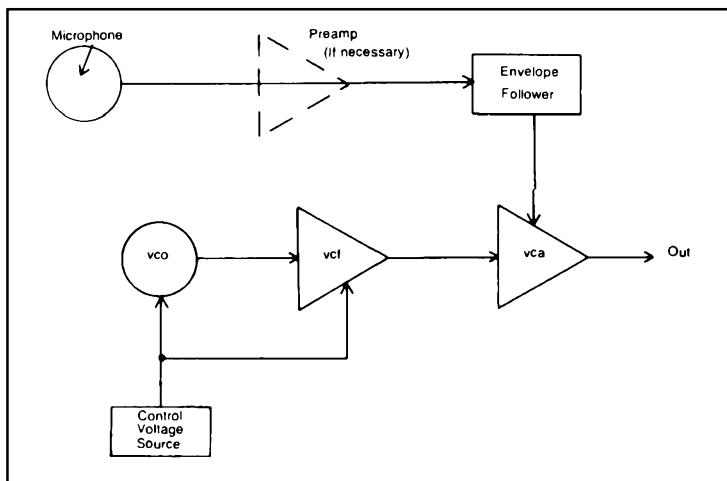


Fig 8-16. Vocal envelopes on electronic sounds.

vcas, there is no reason why the output of an envelope follower can't be used with a vco or vcf.

SEQUENCERS

Another popular automatic controller used in electronic music synthesizers is the sequencer. A *sequencer* is a device that generates a preset pattern of voltages. The output of a typical sequencer is shown in Fig. 8-18. The pattern can be repeated as often as desired. This makes sequencers ideal for rhythm lines.

Essentially a sequencer plays a synthesizer voice by itself, freeing the musician to play other, simultaneous lines.

Many sequencers also output gate and trigger pulses to initiate an envelope on each voltage step.

Four Step. A super simple four step sequencer circuit is shown in Fig. 8-19. The parts list is given in Table 8-7. This circuit is built around five 555 type timers, but a quad 558 unit and a single 555 are shown in the schematic. If you use all 555 single timers, or 556 dual timers, be sure to check the pin numbers. The circuit will

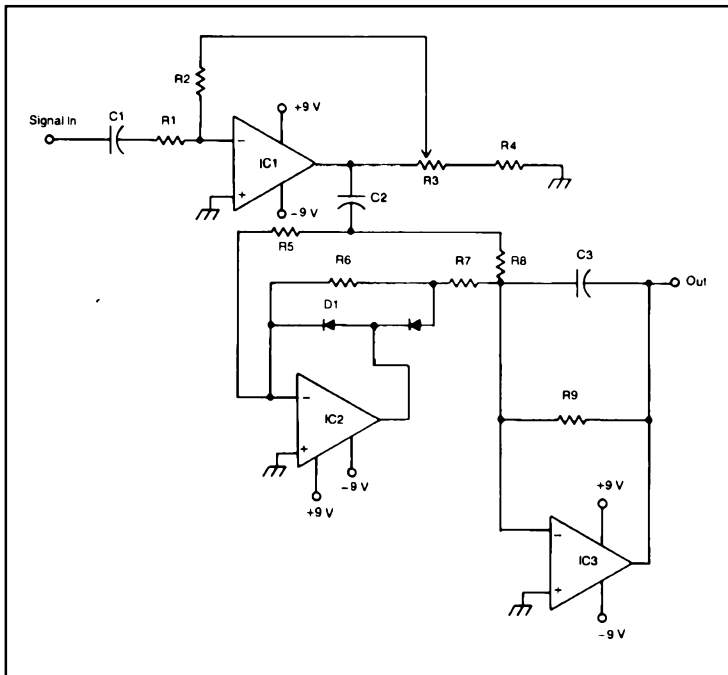


Fig. 8-17. Envelope follower.

Table 8-6. Parts List for Fig. 8-17.

R1	47 k Ω resistor
R2	270 k Ω resistor
R3	5 k Ω potentiometer
R4	100 Ω resistor
R5, R6, R8	39 k Ω resistor
R7	18 k Ω resistor
R9	220 k Ω resistor
C1, C2	0.1 μ F capacitor
C3	0.047 μ F capacitor
D1, D2	silicon diode (1N4148, or similar)
IC1, IC2, IC3	op amp IC

work with any of these versions, but the pins to be connected are different.

This circuit is not really a true sequencer in the sense usually meant by electronic musicians. It is an event sequencer that outputs a series of trigger pulses to initiate various events. No control voltages are outputted. First a trigger signal appears at output 1, then at output 2, output 3, output 4, output 1, output 2, and so forth.

Potentiometers R3, R8, R9, and R10 control the length of each trigger signal. Potentiometers R11 and R11 interact to set the overall sequencer rate.

This device is not as useful in an electronic music synthesizer system as more standard sequencers (several will be presented shortly), but it can be useful in some special cases. A typical patch using this module is shown in Fig. 8-18. Each envelope generator is set to create a different timbre, causing the texture of the sound to continuously change in a regular four step pattern.

Another patch using this four event sequencer circuit is illustrated in Fig. 8-21. Each sound source will be heard in turn.

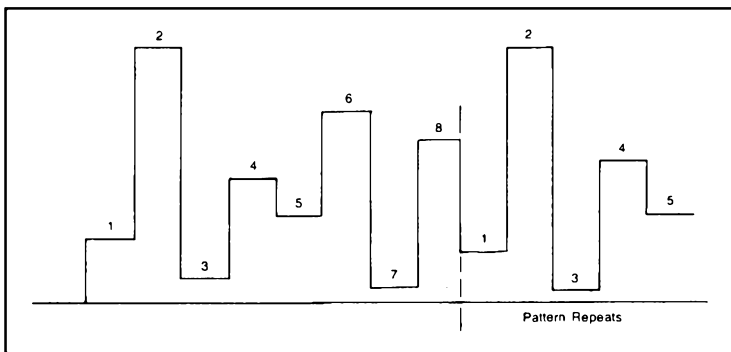


Fig. 8-18 Typical sequencer output.

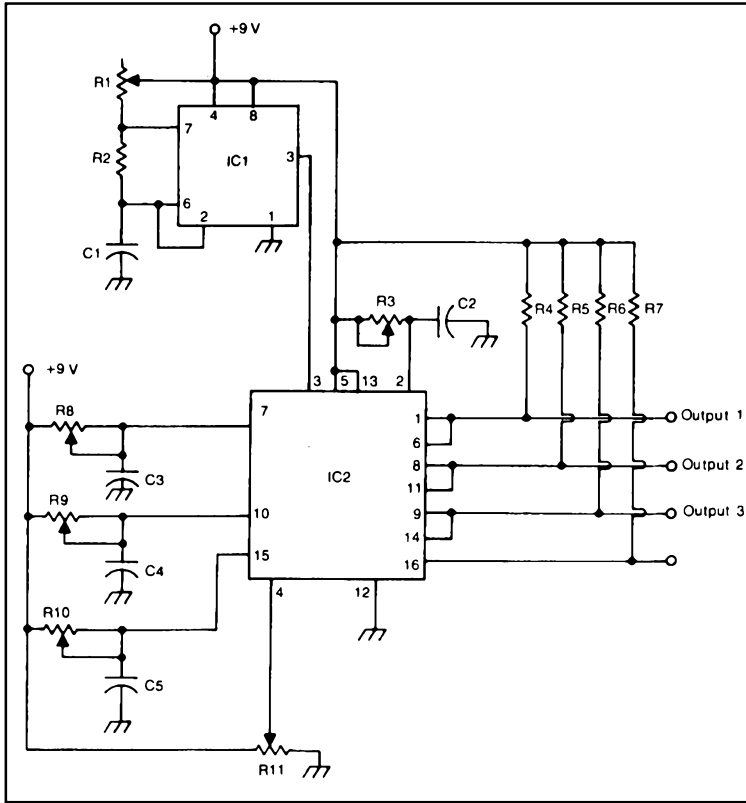


Fig. 8-19. Simple 4 event sequencer.

Eight Step. The circuit shown in Fig. 8-22 is an eight step control voltage sequencer. Each voltage is adjusted by one of the eight potentiometers (R3 through R10). The rate of the sequence will be set by the position of R1 and S1 (which selects one of six timing capacitors).

Notice that this circuit has two outputs. Pin 3 of IC1 provides the gate signal for triggering. The control voltage output is the common end of potentiometers R3 through R10.

You might want to consider adding LEDs in series with each of the voltage output level potentiometers, so you can see which step the sequencer is currently on. Another modification would be to insert a switch between pin 3 of IC1 and pin 5 of IC2. This will allow you to stop the sequencer on any step.

The parts list for the eight step sequencer is given in Table 8-7.

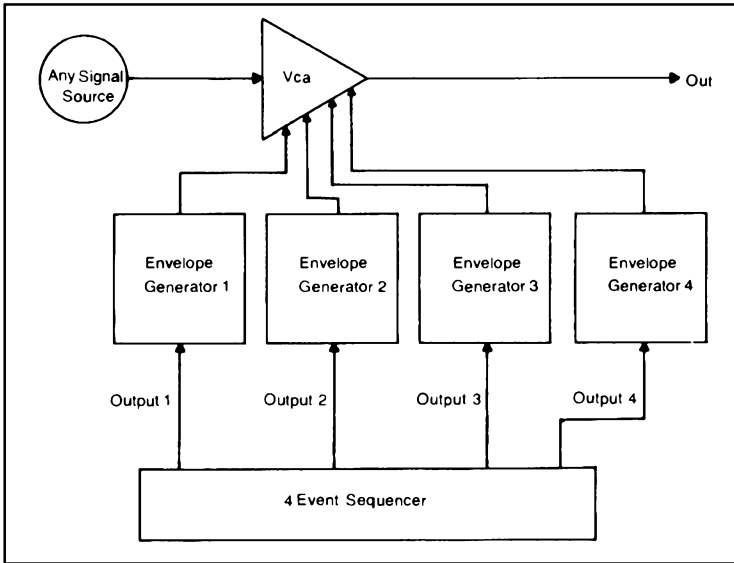


Fig. 8-20. Typical patch for the simple 4 event sequencer.

Ten Step. Another fairly simple sequencer circuit is illustrated in Fig. 8-23. It is quite similar to the one we just discussed except that this generates a ten step sequence. As the parts list in Table 8-9 indicates, somewhat different circuitry is used here to achieve basically the same ends.

Thirty-two Step. The final sequencer we will look at here is a little different. This unit steps through a 16 step sequence, then runs back through the entire sequence backwards, before starting over (forwards again). The circuit is shown in Fig. 8-24, and the parts list is given in Table 8-10.

Potentiometers R3 through R18 set the output voltage levels for each step. The sequence rate is determined by potentiometer R1.

Table 8-7. Parts List for Fig. 8-19.

R1	500 k Ω potentiometer
R2	2.2 k Ω resistor
R3, R8, R9, R10, R11	10 k Ω potentiometer
R4, R5, R6, R7	3.3 k Ω resistor
C1	0.047 μ F capacitor
C2, C3, C4, C5	0.1 μ F capacitor
IC1	555 timer IC
IC2	558 quad timer IC (see text)

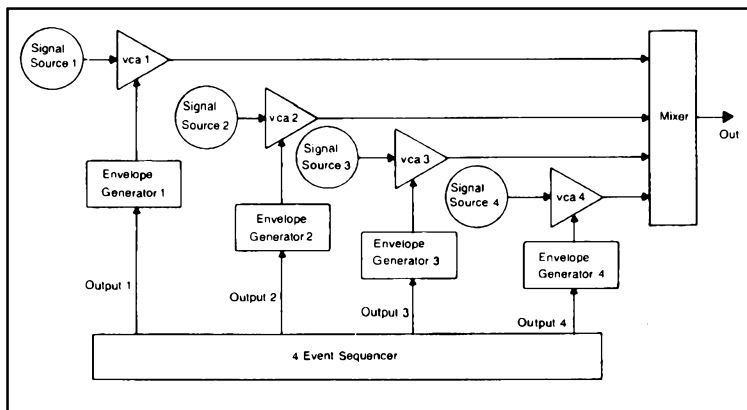


Fig. 8-21. A second typical patch for the simple 4 event sequencer.

This back and forth sequence can create some fascinating patterns, but it might not always be musically useful. Many patterns that you will want your sequencer to perform won't sound quite right. However, when you make use of it, this circuit effectively gives you a 32 step (16 forward and 16 backward) sequencer at a ridiculously low cost. You should be able to build this device for less than \$30. The most expensive parts will be the 17 potentiometers.

PSEUDORANDOM VOLTAGE SOURCES

Many people involved in electronic music like to experiment with chance music. That is, they want their synthesizer to randomly play itself. Figure 8-25 shows a circuit that can help them achieve this end. The parts list is given in Table 8-11.

Table 8-8. Parts List for Fig. 8-22.

R1	500 k Ω potentiometer
R2	1 k Ω resistor
R3-R10	1 k Ω potentiometer
C1	0.01 μ F capacitor
C2	0.1 μ F capacitor
C3	1 μ F electrolytic capacitor
C4	10 μ F electrolytic capacitor
C5	50 μ F electrolytic capacitor
C6	100 μ F electrolytic capacitor
S1	Single-pole/6 throw rotary switch
IC1	555 timer IC
IC2	74LS193 up/down counter IC
IC3	7404 hex inverter IC (only 1 section is used)
IC4	74LS138 decoder/demultiplexer IC

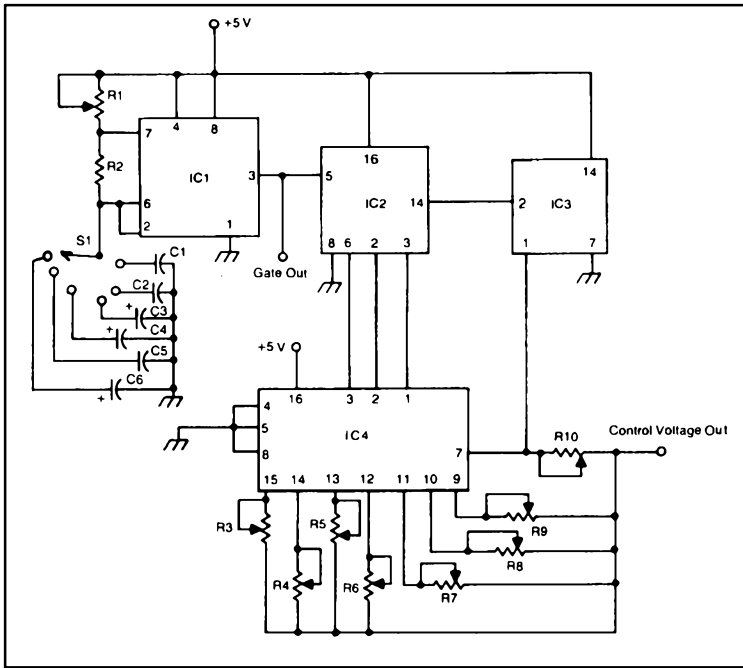


Fig. 8-22. 8 step sequencer.

This circuit generates a more or less random series of control voltages. It is not a truly random series. The pattern will be repeated. But the entire pattern is long enough to sound random.

The pseudorandom voltage source is operated by a single potentiometer and two rotary switches. Potentiometer R1 simply sets the rate of the clock pulses, and thus the speed at which the

Table 8-9. Parts List for Fig. 8-23.

R1	500 k Ω potentiometer
R2	1 k Ω resistor
R3-R12	1 k Ω potentiometer
C1	1 μ F electrolytic capacitor
C2	2.2 μ F electrolytic capacitor
C3	4.7 μ F electrolytic capacitor
C4	10 μ F electrolytic capacitor
C5	25 μ F electrolytic capacitor
C6	47 μ F electrolytic capacitor
S1	Single-pole/6 throw rotary switch
IC1	555 timer IC
IC2	7490 decade counter IC
IC3	7441 BCD decoder IC

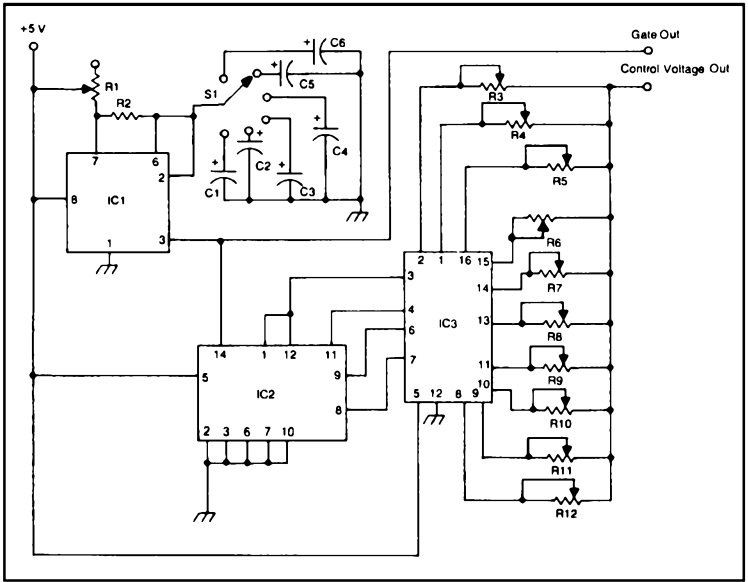


Fig. 8-23. 10 step sequencer.

output voltages will change. The capacitors selected by switch S2 determine the range of clock speeds. These six capacitors allow a very wide range of speeds to be selected.

The third control, switch S1 can be used to change the pattern. Each position of this switch should create a somewhat different sounding "random" pattern.

There are a number of ways you can adapt this circuit. Smaller capacitors will speed up the pattern, and larger capacitors will slow it down. The 1 kΩ resistors (except R2) may be made as large as 10 kΩ, but they should all have equal values, and the resistors listed as 2.2 kΩ in the parts list should have approximately twice the new value of the 1 kΩ resistors. Reducing these resistances is not recommended.

Table 8-10. Parts List for Fig. 8-24.

R1	100 kΩ potentiometer
R2	1 kΩ resistor
R3-R18	1 kΩ potentiometer
C1	0.68 μF capacitor
IC1	555 timer IC
IC2	7400 quad NAND IC
IC3	74LS193 up/down counter IC
IC4	74154 (4 to 16 decoder/demultiplexer) IC

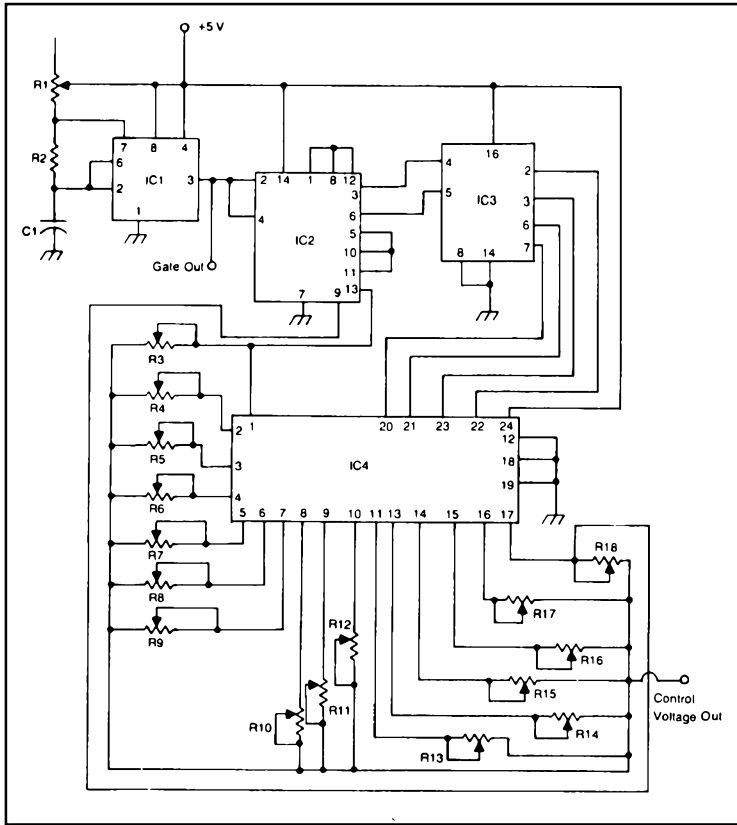


Fig. 8-24. Back & forth sequencer.

The pseudorandom voltage source is a unique module, not found on most synthesizers. It can produce some fascinating effects.

Using this device to control a vco will produce a series of random tones. These tones are not likely to sound very melodic, however. A gate signal could be taken off of pin 3 of IC3 to trigger an envelope generator. Using the pseudorandom voltage source to control a vca will give an interesting warbling effect, especially if the voltage generator is operating at a fairly high speed. A patch for this effect is shown in Fig. 8-26. Operating a vcf with the pseudorandom voltage source can also create some intriguing sounds.

SAMPLE AND HOLD

Many synthesizers achieve pseudorandom control voltage ef-

fects with special circuits called sample and hold, or S/H.

A *sample and hold* circuit has two inputs and a single output. One input is an ac signal source, and the other is a pulse or clock signal. Each time a clock pulse triggers the S/H circuit, the instantaneous level of the ac signal is captured and held constant at the output until the next clock signal is received. The effect is illustrated in Fig. 8-27. You can see from this diagram that the output of the circuit is a series of pseudorandom control voltages that change at a rate equal to the clock frequency. Obviously, the clock frequency should be fairly low with respect to the input signal. In most cases, the clock frequency will be below the audible range.

Input Requirements

For the pseudorandom effect, the clock frequency and the input frequency should not be harmonically related. If these frequencies are harmonically related, a short repeating pattern will be created. This may be desirable in some cases.

Any ac signal may be fed to the input of a sample and hold

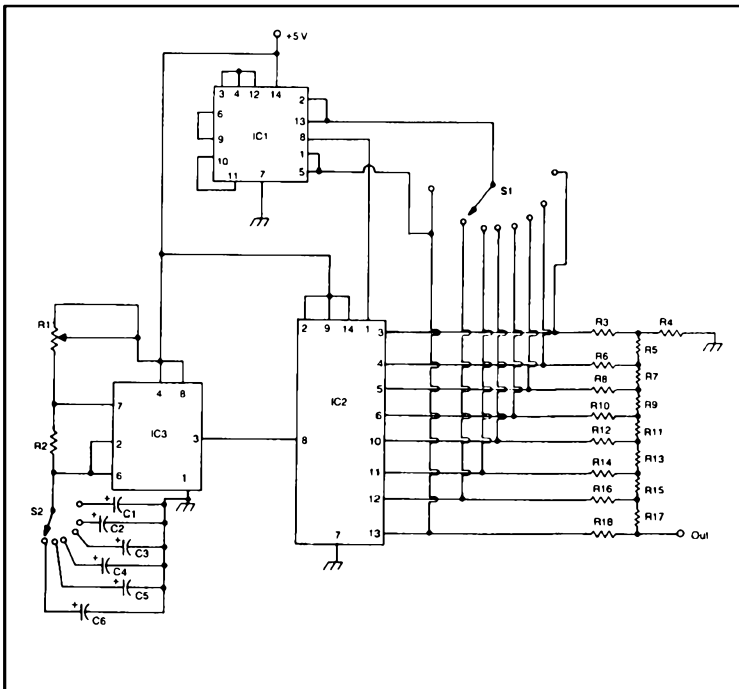


Fig. 8-25. Pseudorandom voltage source.

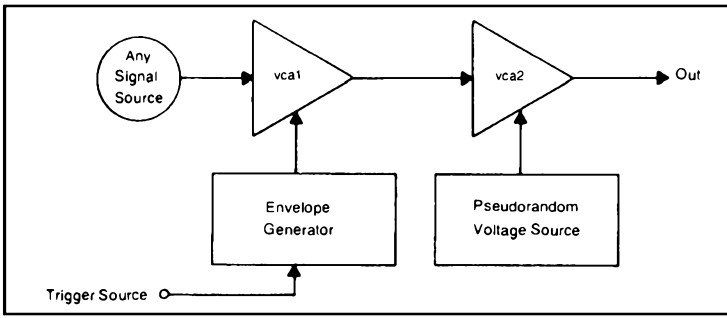


Fig. 8-26. Random warble effect patch.

circuit, although rectangle waves will produce rather uninteresting results. This point is illustrated in Fig. 8-28.

A sine wave was shown as the input in Fig. 8-27. Figure 8-29 demonstrates the effect if the input signal is an ascending sawtooth

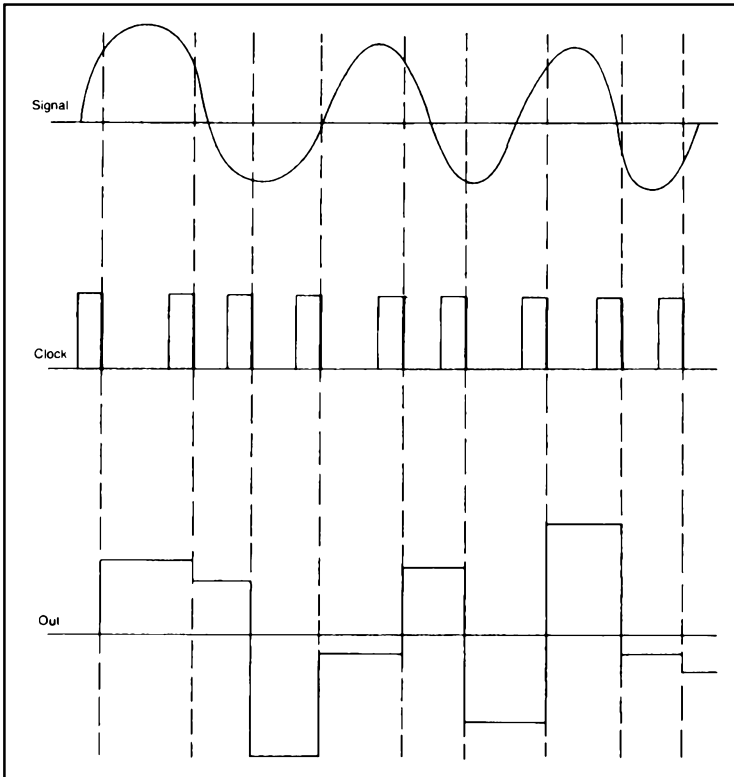


Fig. 8-27. Sample and hold effect with a sine wave.

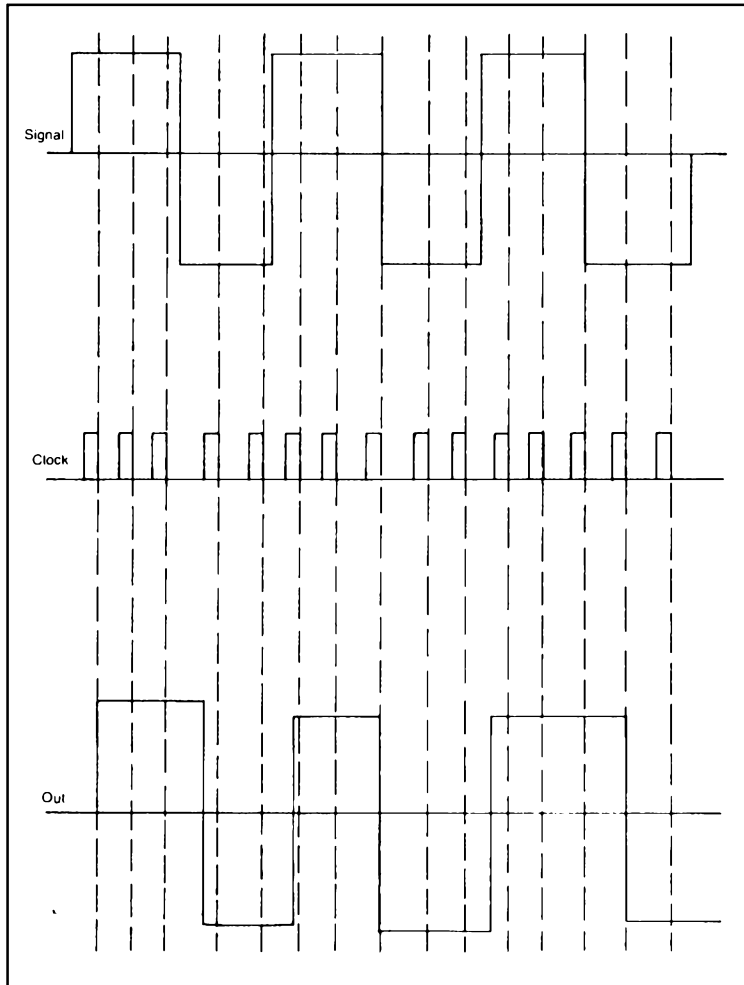


Fig. 8-28. Sample and hold effect with a rectangle wave.

wave, and Fig. 8-30 shows a triangle wave as the input signal. Of course, the relative frequencies of the input signal and the clock signal will also have a dramatic effect on the output pattern.

Not surprisingly, the most random effects can be achieved if random noise is used as the input signal. This is shown in Fig. 8-31.

Whenever any periodic waveform is used as the input to a sample and hold, the output pattern will eventually repeat. Of course, this is not true for a noise signal, which is aperiodic. The length of the pattern before it repeats will depend on the lowest

common multiple of the input signal frequency and the clock frequency. By avoiding harmonic relationships, very long patterns can be generated. A long enough pattern will tend to sound purely random, and the repetition will tend to go unnoticed.

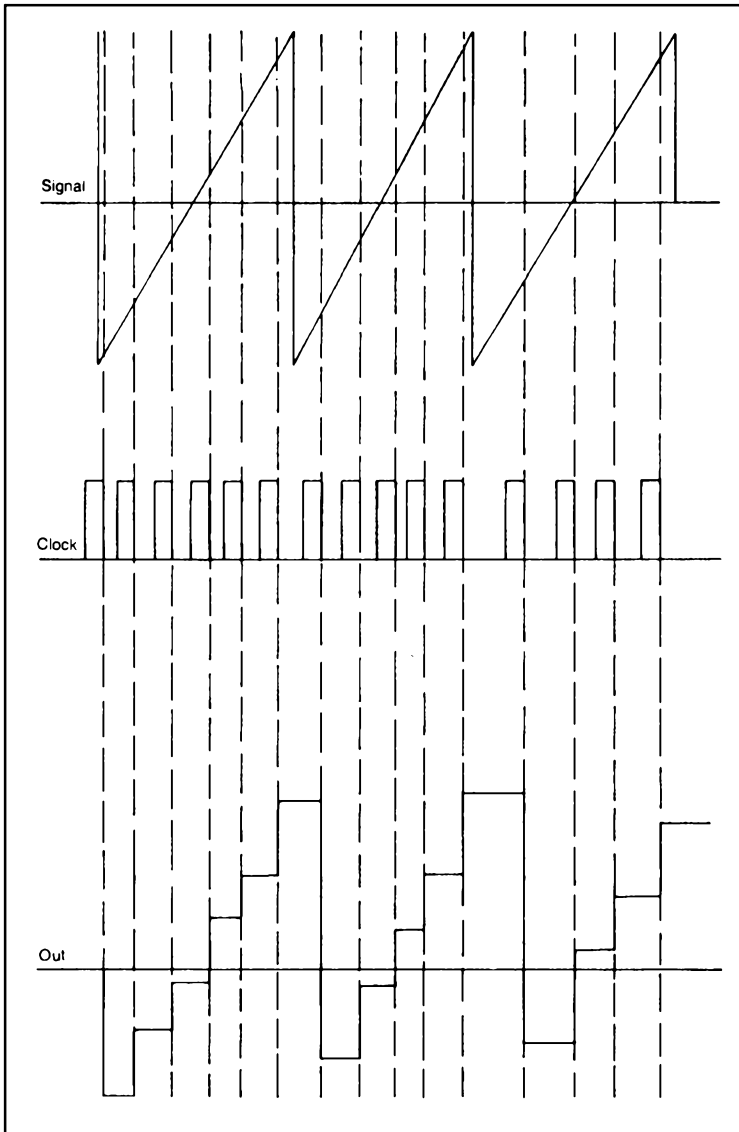


Fig. 8-29. Sample and hold effect with a sawtooth wave.

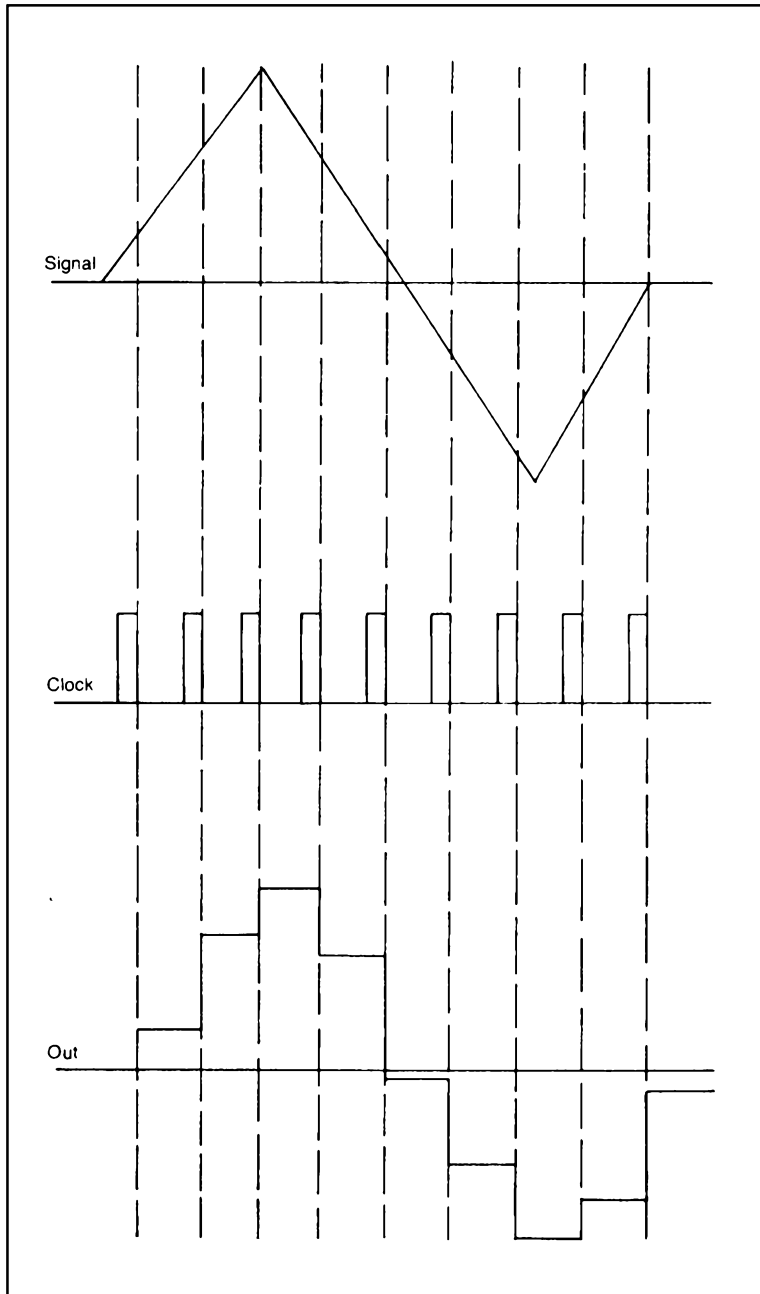


Fig. 8-30. Sample and hold effect with a triangle wave.

Circuits

In most sample and hold circuits, the held output value is stored by a relatively large capacitor. This capacitor must be capable of being charged up very quickly, while having a very low amount of leakage. No capacitor is ideal, so no practical sample and hold circuit can hold an output value indefinitely. Eventually the charge must leak off the capacitor. This effect is illustrated in Fig. 8-32.

Op Amp S/H. A simple sample and hold circuit built around a low noise op amp IC is illustrated in Fig. 8-33. The parts list is given in Table 8-12.

The op amp is actually a buffer amplifier with a gain of 1 (unity). It is used to help reduce excessive leakage into the circuit con-

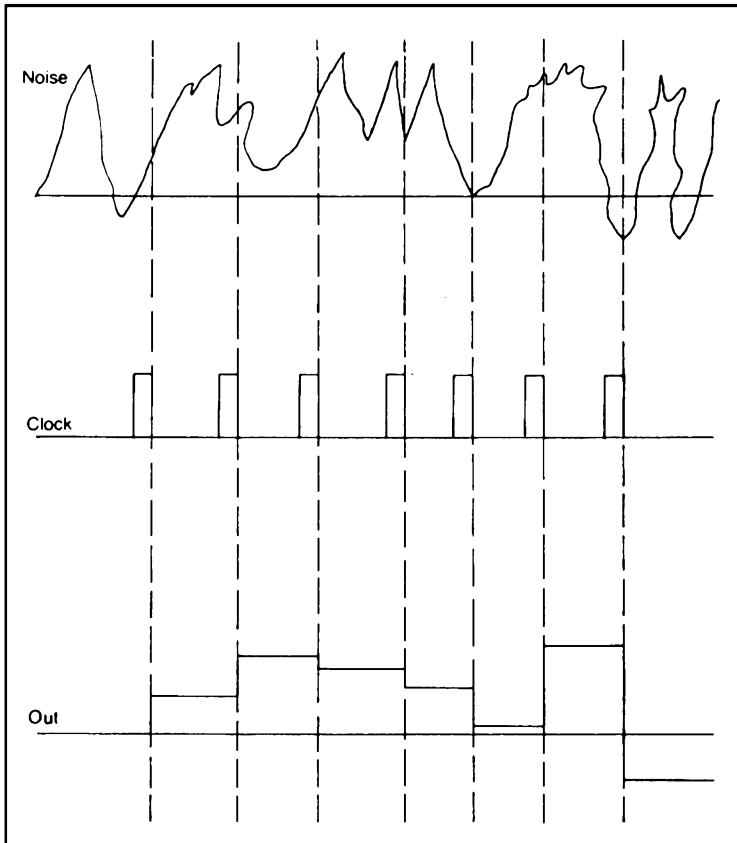


Fig. 8-31. Sample and hold effect with noise.

Table 8-11. Parts List for Fig. 8-25.

R1	500 k Ω potentiometer
R2, R5, R7, R9, R11, R13, R15, R17, R3, R4, R6, R8, R10, R12, R14, R16, R18	1 k Ω resistor
C1	2.2 k Ω resistor
C2	1 μ F electrolytic capacitor
C3	2.2 μ F electrolytic capacitor
C4	5 μ F electrolytic capacitor
C5	10 μ F electrolytic capacitor
C6	22 μ F electrolytic capacitor
S1	50 μ F electrolytic capacitor
S2	Single-pole/8 throw rotary switch
IC1	Single-pole/6 throw rotary switch
IC2	74LS00 quad NAND gate IC
IC3	74LS164 shift register IC
	555 timer IC

nected to the output. The real work of sampling and holding is done by the FET, Q1, and capacitor C3. This capacitor must be a tantalum type. Electrolytic capacitors have too much leakage for this application.

IC2, R1, R2, and C1 make up a simple clock pulse generator. If you'd prefer to use an external clock, you can eliminate these four components. The clock pulses should be fed into the circuit at D1. A sample is taken when the clock signal drops from about 5 V or 6 V to ground.

HA 2420 S/H IC A number of dedicated sample and hold IC devices are available from several manufacturers. These devices are not designed specifically for electronic music, but they can be readily used.

The pin-out diagram for the HA 2420 S/H IC is shown in Fig. 8-34. The input stage of this unit is a high performance operational amplifier. This op amp was specially designed to offer an extremely fast slew rate and to drive highly capacitive loads without instability. These features are essential in sample and hold circuits. Instability could result in oscillation when a large holding capacitor is used for long holding times.

As with all operational amplifiers, there are two inputs to this circuit—an inverting input (the output is 180° out of phase with the input), and a noninverting input (the output is in phase with the input). Either of these inputs may be used separately, or they may be used together for some unusual special effects.

The heart of any sample and hold circuit is the switching element, which is controlled by the clock's trigger pulses. In the HA

2420, this is a high efficiency bipolar switching transistor stage. The leakage through this stage is extremely low when it is in its OFF condition. When the clock signal (applied to pin 14, S/H CONTROL) is high, the circuit is switched into the holding mode. Whatever the input signal was at the instant of switching will be held at the output. Conversely, a low clock signal switches the circuit into the sampling mode—that is, the output will follow the level changes of the input signal.

(Note that if the original input signal is applied to the inverting input, the output signal will be 180° out of phase. In other words, when the input goes positive, the output will go negative by a like amount, and vice versa. A signal at the noninverting input will be duplicated at the output without a change of polarity.)

The output stage of the HA-2420 S/H IC is another op amp, which is connected as a simple voltage follower, similar to what we had in Fig. 8-33.

Drifting can be a problem with this unit. To prevent excessive drift, great care must be taken to minimize leakage paths, especially on the PC board itself. Remember that a capacitor is simply two conductors separated by an insulator, so two adjacent foil traces on a PC board can act as a phantom capacitor. This is less of a problem at the audio frequencies we'll be working with than it is at radio frequencies. However, it would still be a good idea to take some simple precautions against possible future headaches.

Since the output voltage (pin 7) is approximately equal to the voltage on the holding capacitor (pin 11), it can easily be used as a guard line surrounding the capacitor connection. To facilitate this, the two pins on either side of the holding capacitor connection (pins 10 and 12) are designated as guard pins and are not connected to the chip's internal circuitry. Figure 8-35 illustrates how these pins may be connected in a loop on the PC board at the same potential as the output line. Any stray capacitances that occur between the foil

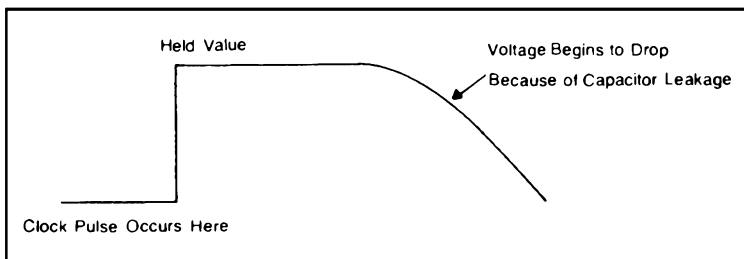


Fig. 8-32. Effect of capacitor leakage on a held voltage.

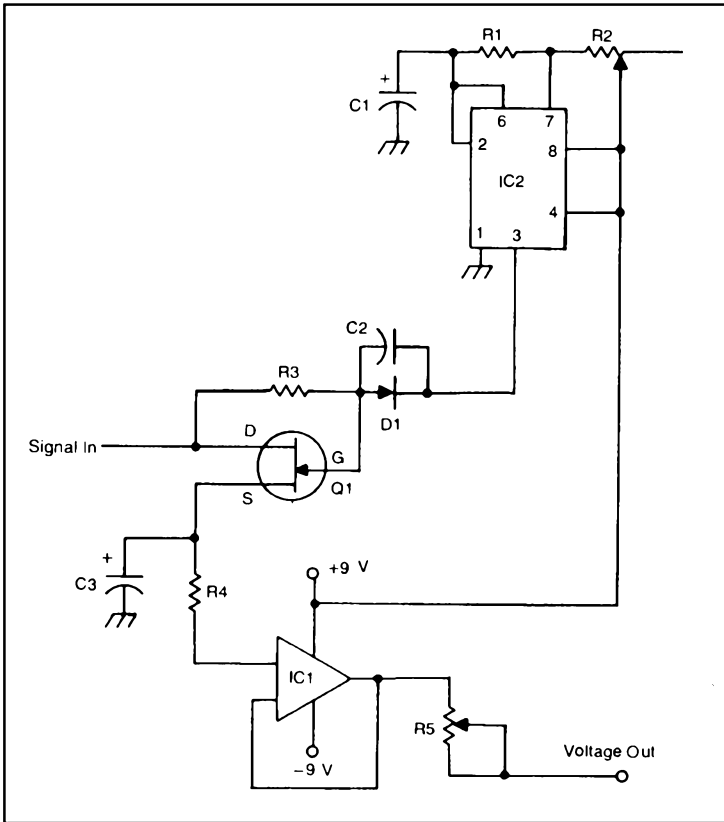


Fig. 8-33. An op amp sample and hold circuit.

traces will be of little importance since the guard loop is already at the same voltage as the holding capacitor.

Figure 8-36 shows a simple manually clocked circuit using the HA 2420 S/H IC. This circuit is for demonstration and/or experimental purposes only. It would be of little practical value in an electronic music system.

When the switch is in its grounded position, the circuit is in the sample mode. Moving the switch to the 3 volt position places the circuit in the hold mode.

Capacitor C, of course, is the holding capacitor. The larger this capacitor is, the longer the maximum hold time that will be possible without excessive drift. However, an important trade-off comes into play here. Increasing the capacitance of C increases its charging rate, reduces the slew rate of the circuit as a whole, and reduces

Table 8-12. Parts List for Fig. 8-33.

R1	1 k Ω resistor
R2	500 k Ω potentiometer
R3	120 k Ω resistor
R4	2.2 k Ω resistor
R5	10 k Ω potentiometer
C1	10 μ F electrolytic capacitor
C2	50 pF capacitor
C3	5 μ F tantalum capacitor (see text)
D1	silicon diode (1N4148 or similar)
Q1	FET (A5T3821, Radio Shack RS2028, or similar)
IC1	high quality op amp IC (308, 536, or similar)
IC2	555 timer IC

the frequency bandwidth during the sample period. All of this means that when the circuit is switched into the hold mode, it may not be able to latch precisely on the instantaneous value of the input signal if C is made too large.

For voltage control purposes in an electronic music synthesis system, the sampling bandwidth isn't particularly critical, although the effect on the charging rate and the slew rate may cause the

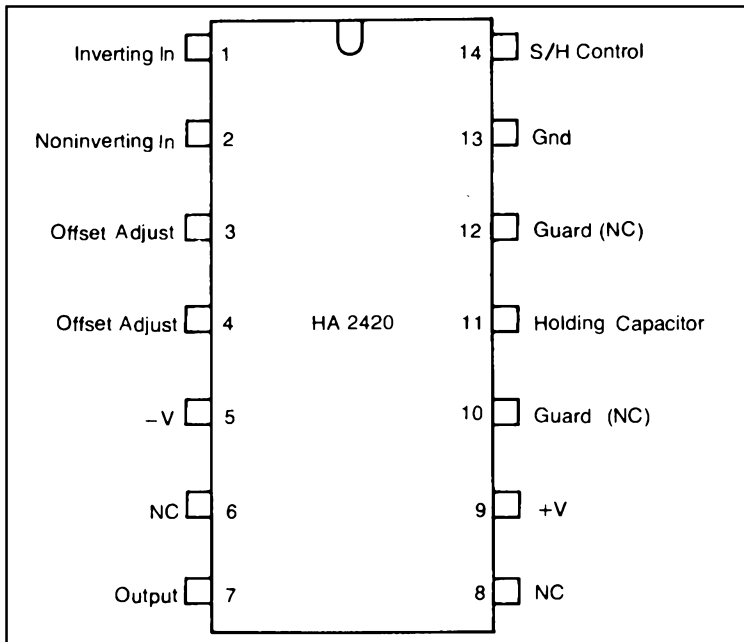


Fig. 8-34. Pin-out diagram of the HA 2420 sample and hold IC

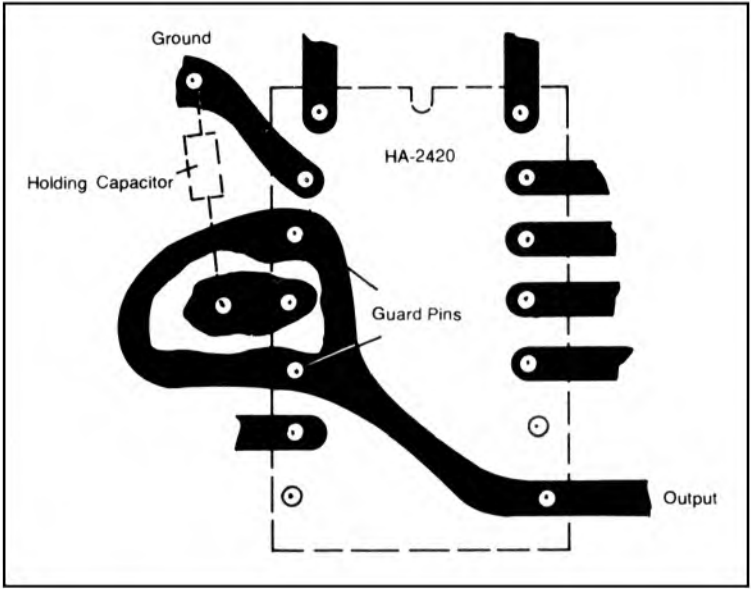


Fig. 8-35. Using the guard pins on the HA 2420 IC to prevent phantom capacitances.

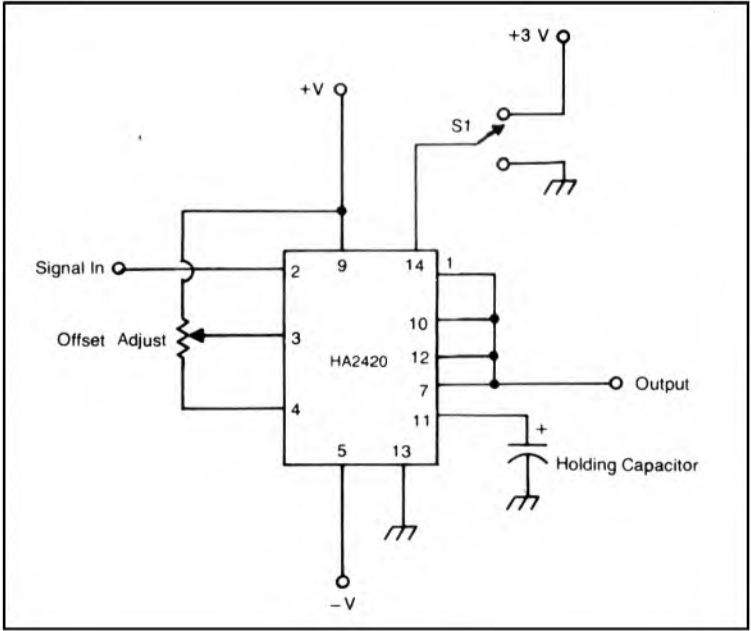


Fig. 8-36. Manually clocked demonstration of the HA 2420 S/H circuit.

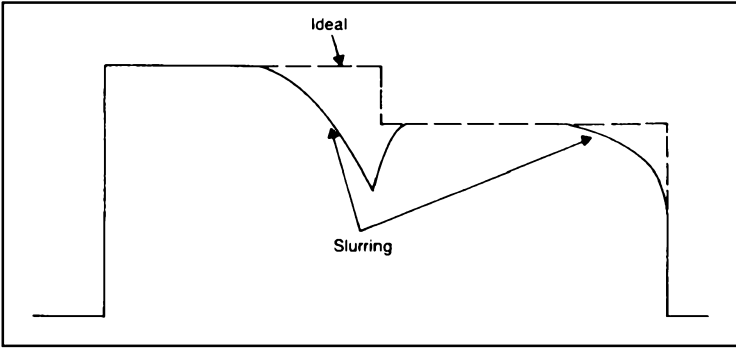


Fig. 8-37. Slurring due to long hold times.

holding levels to slur from value to value when long holding times are used. This problem is illustrated in Fig. 8-37.

A practical version of this sample and hold circuit is shown in Fig. 8-38. We have simply replaced the manual switch with a simple pulse generator. This circuit is a square wave generator, so the sample times and hold times are of equal length. The output signal will therefore look something like the one shown in Fig. 8-39. This could be annoying in some cases, but if you use the gate signal to trigger an envelope generator as illustrated in the patch diagram of

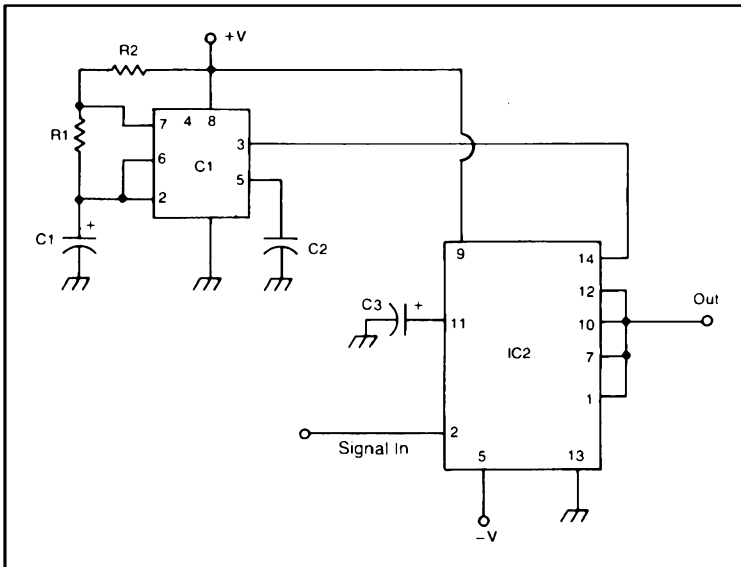


Fig. 8-38. A practical HA 2420 S/H circuit.

Fig. 8-40, the vca will be cut off during most of the sampling period anyway.

Alternatively, you could use an external clock signal. Ideally, the high portion of the waveform should be high for most of each cycle, as shown in Fig. 8-41. This would keep the hold times relatively long and the sample times relatively short, making for sharper steps in the output.

If the output voltages are too low for your desired application, you can build a similar S/H circuit around the HA 2420 IC with gain. This circuit is shown in Fig. 8-42. The amount of gain is determined by the combined value of resistors R3 and R4. The formula for the gain of this circuit is as follows:

$$G = \frac{R4}{R3+R4}$$

In any sample and hold circuit, if a complex signal or a noise source is being used as the sampled signal input, some low-pass filtering will probably be desirable. Figure 8-43 illustrates a simple S/H circuit incorporating a passive filter. If an active filter is used, the sampling time must be long enough for the filter to settle. This

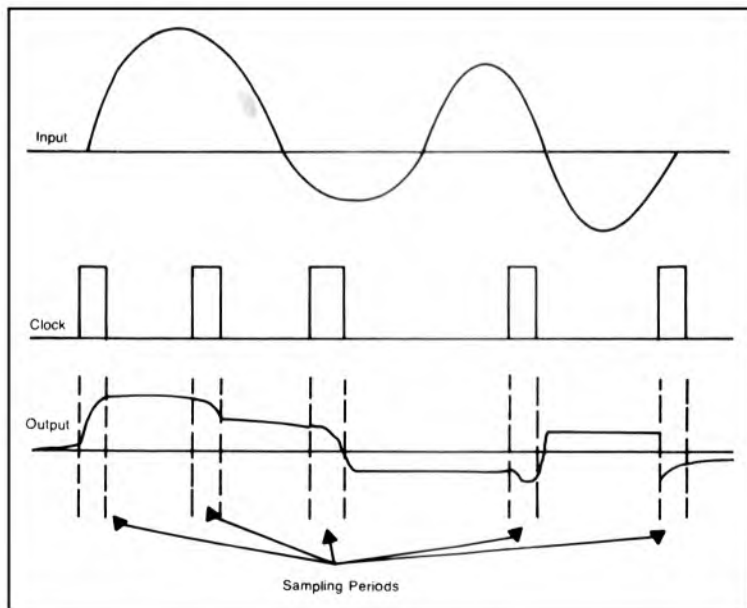


Fig. 8-39. Output signal of the circuit in Fig. 8-38.

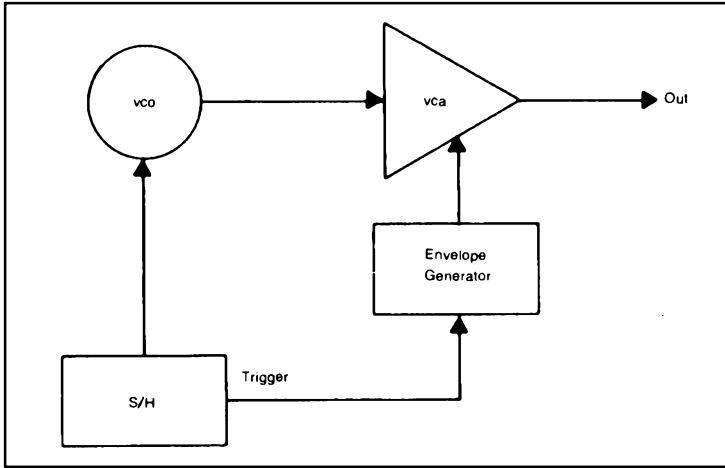


Fig. 8-40. Basic sample and hold patch.

is one case where a passive filter may be preferable to an active circuit.

Other ICs. The HA 2420 is by no means the only sample and hold IC on the market today. It would be outside the scope of this book to cover all of the devices now on the market.

However, to demonstrate the variety of ICs available, Fig. 8-44 shows the pin-out diagram for the LF198/LF298/LF398 series of sample and hold units from National Semiconductor. This eight pin device is somewhat simpler than the HA 2420, but certainly adequate for electronic music applications.

The holding capacitor should be connected between pins 6 and 7. No inverting input is available on this chip.

RHYTHM PATTERN GENERATOR

Similar to the sequencer is the rhythm pattern generator. In fact, these devices could be used together.

One major problem with most sequencers, including those



Fig 8-41 Ideal clock waveform.

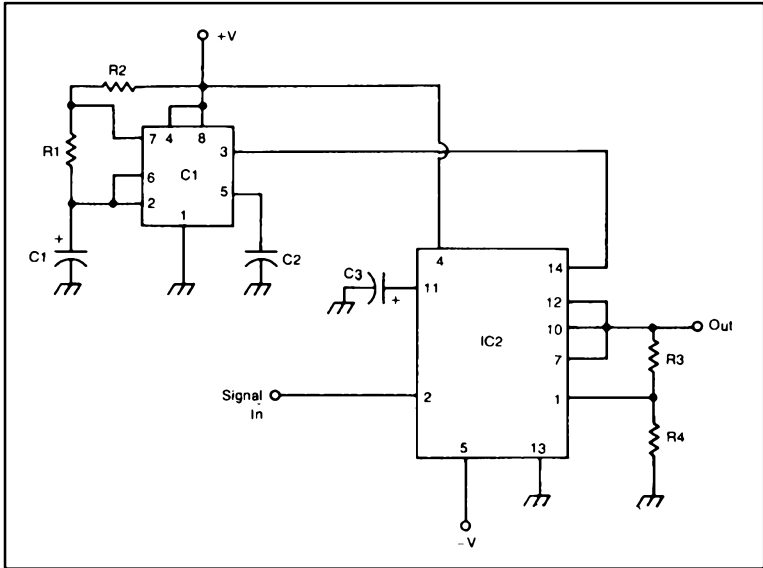


Fig. 8-42. HA 2420 S/H circuit with gain

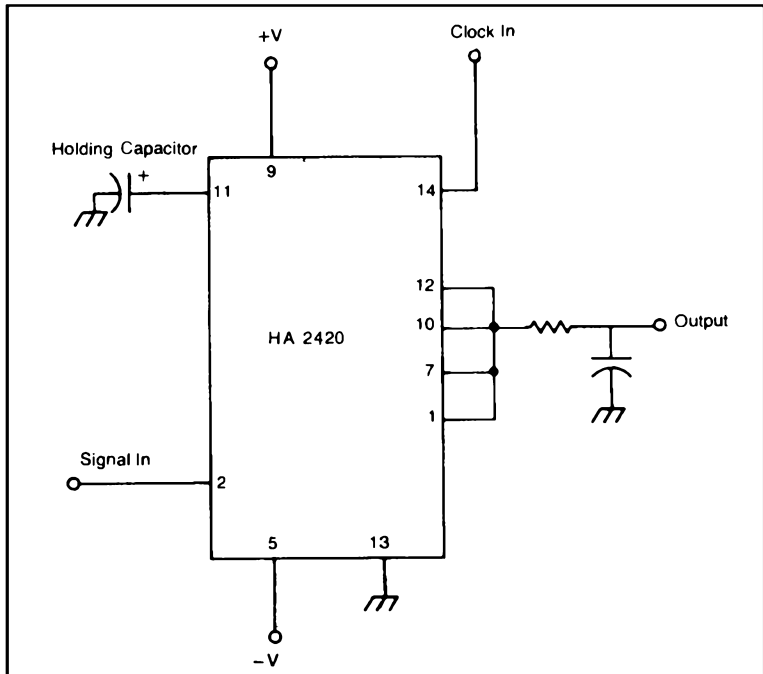


Fig. 8-43. S/H circuit with a passive filter.

presented earlier in this chapter is that each step is equally spaced. Rhythmically, this can be rather dull.

A rhythm pattern generator, on the other hand, feeds out a string of rhythmic trigger pulses, often to multiple outputs.

A rhythm pattern generator in IC form is the MM5871. The pin-out diagram for this chip is shown in Fig. 8-45. This device was designed for use in rhythm boxes or automatic drummers, although an imaginative experimenter could find a number of additional applications. It is preprogrammed with six standard rhythm patterns and has five trigger signal outputs.

Three of the six available rhythms are in 3/4 time. These are waltz, swing, and slow rock. The other three preprogrammed rhythms—samba, bossa, and rock—are in 4/4 time. For very unusual effects, you can select more than a single pattern at a time.

The five trigger signal outputs are intended to drive generators for synthesizing snare drum, brushes, bongo, woodblock and bass drum. The MM5835N (discussed in Chapter 3) is ideal for producing the snare and brush effects. The two chips were designed to be compatible. Of course, these trigger outputs could also be used to trigger envelope generators and similar modules instead of drum synthesizers.

The MM5871 rhythm pattern generator IC has an on chip tempo-generator clock. The pattern speed can be manually controlled with an external potentiometer, or voltage control may be applied to pin 2 for dynamically changing rhythm rates. Connecting a 0.005 μ -F capacitor from pin 2 to ground and a resistance of about 120 k Ω to 1.1 m Ω connected from pin 2 to V_{DD} , will give a tempo range extending from approximately 2.5 beats per second to more than 25 beats per second. This is a reasonable range for musical applications.

The 3/4 time patterns are based on six cycles of the tempo

Table 8-13. Parts List for Fig 8-38.

R1	1 k Ω resistor
R2	27 k Ω resistor
C1	10 μ F electrolytic capacitor
C2	0.01 μ F capacitor
C3	22 μ F tantalum capacitor
IC1	555 timer IC
IC2	HA-2420 S/H IC

Table 8-14. ¾ Rhythm Patterns for the MM 5871.

Rhythm pattern

Outputs	Counts—Swing	Counts—Slow Rock	Counts—Waltz
	0 1 2 3 4 5	0 1 2 3 4 5	1 2 3 4 4 5
pin 9 (bass drum)	X - - - -	X - - - -	X - - - -
pin 10 (wood block)	- - - X -	X X X X X	- - X - X -
pin 11 (bongo)	X - - - -	X - - - -	X - - - -
pin 12 (brush)	- - - X -	- - - X -	- - X - X -
pin 13 (snare)	X - - X - X	X X X X X	- - X - X -

clock. Table 8-13 illustrates which output pins are triggered on each clock pulse. Similarly, the 4/4 time patterns are derived from eight clock cycles. Table 8-15 shows the output triggers for this set of rhythm patterns.

The pulse width of the output trigger signals is variable which allows the synthesist to use the rhythm pattern generator with a number of other modules with a minimum of problems. An RC network connected to pin 3, as shown in Fig. 8-46, provides control of the output pulse width.

All five trigger outputs are open drain transistor circuits. This

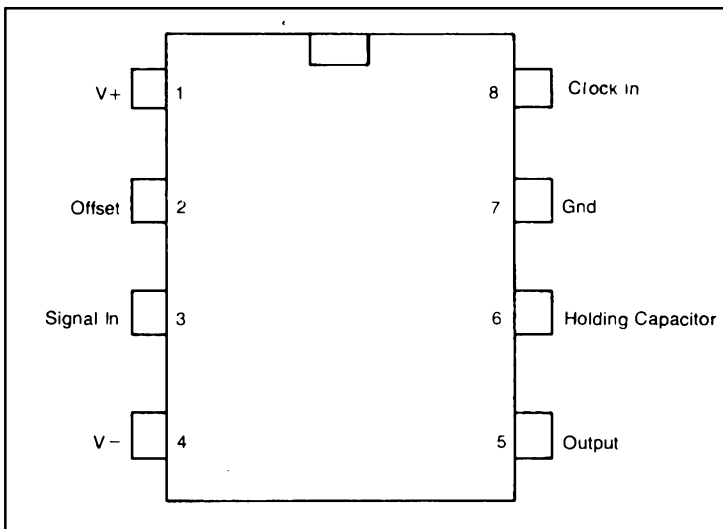


Fig. 8-44. Varying the output trigger pulse width on the MM 5871.

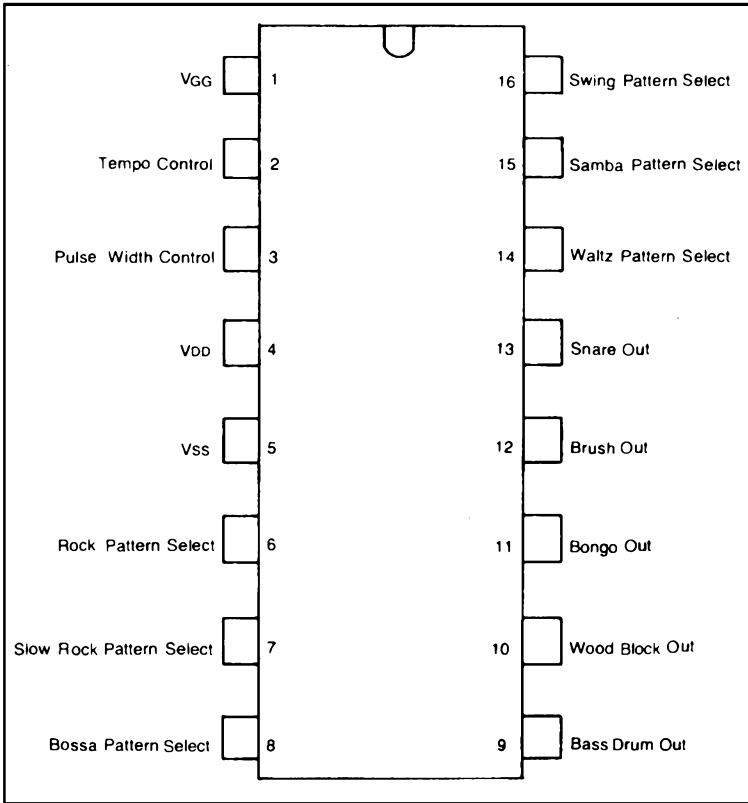


Fig. 8-45. Pin-out diagram for the MM 4871 rhythm pattern generator IC

means the outputs present a high impedance to the load device being triggered when off. When an output is activated, a logic high signal will be fed to the load device.

Figure 8-47 shows a typical automatic drummer circuit. Some of the resistance values in the drum synthesizer circuits may have to be altered somewhat for the best results. Experiment with various resistor values. The parts list for this circuit is given in Table 8-16.

The MM5871 rhythm pattern generator IC is fairly tolerant of a wide range of supply voltages. Two supply voltages are required for chip operation, and both are below ground level. Changing V_{oo} with respect to ground will affect the tempo control and the pulse width control, so voltage regulation is recommended. Typical supply voltage values are $V_{oo} = -14V \pm 0.5 V$, and $V_{ss} = -28 V$.

Of course, the sound generators triggered by the MM5871 do not necessarily have to be drum synthesizers. Virtually any rela-

Table 8-15. 4/4 Rhythm Patterns for the MM 5871.

Rhythm Pattern			
Outputs	Counts—Rock 0 1 2 3 4 5 6 7	Counts—Samba 0 1 2 3 4 5 6 7	Counts—Boss 0 1 2 3 4 5 6 7
pin 9 (bass drum)	X - - X - -	X - - X - -	X - - X - -
pin 10 (wood block)	- - X X - X -	X - - X - X - -	- - X - - X - -
pin 11 (bongo)	X - - X X - -	X - - X - -	- - - - - X X
pin 12 (brush)	- - X - - X -	X - - X - X - -	X - - X - - - -
pin 13 (snare)	X X X X X X X X	X X X X X X X X	X X X X X X X X

tively short sound you desire can be triggered in the rhythmic patterns generated by this chip. Some very unique effects can be achieved by using the MM5871 to trigger five SN74677 complex sound generator chips (see Chapter 3). Or, as mentioned earlier, the rhythm pattern generator can control a series of envelope generators.

For more complex (and rather odd) rhythm patterns, you might try experimenting with driving five rhythm pattern generators with

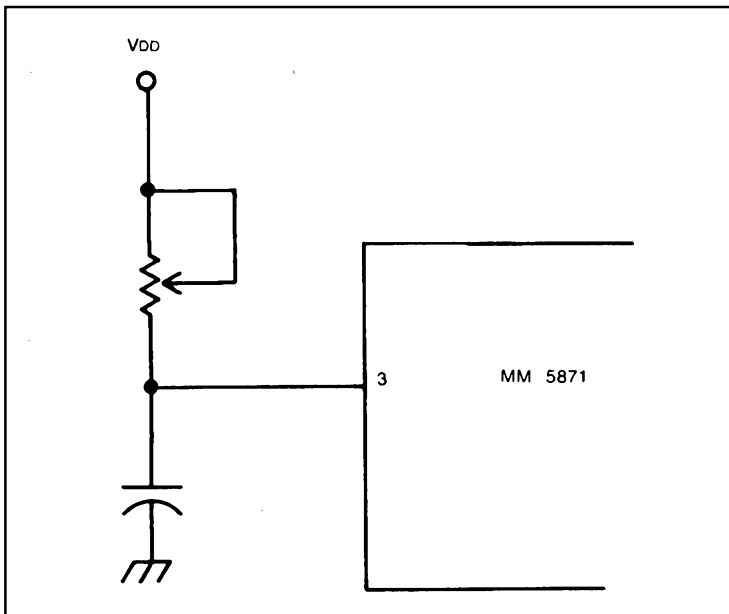


Fig. 8-46. Varying the output trigger pulse width on the MM 5871.

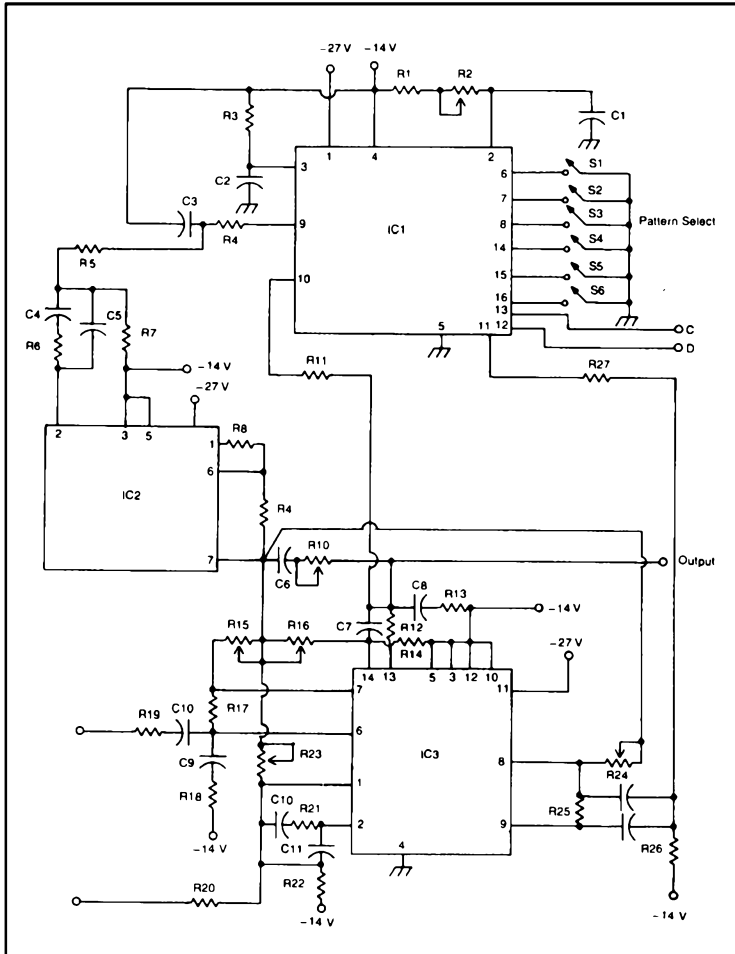
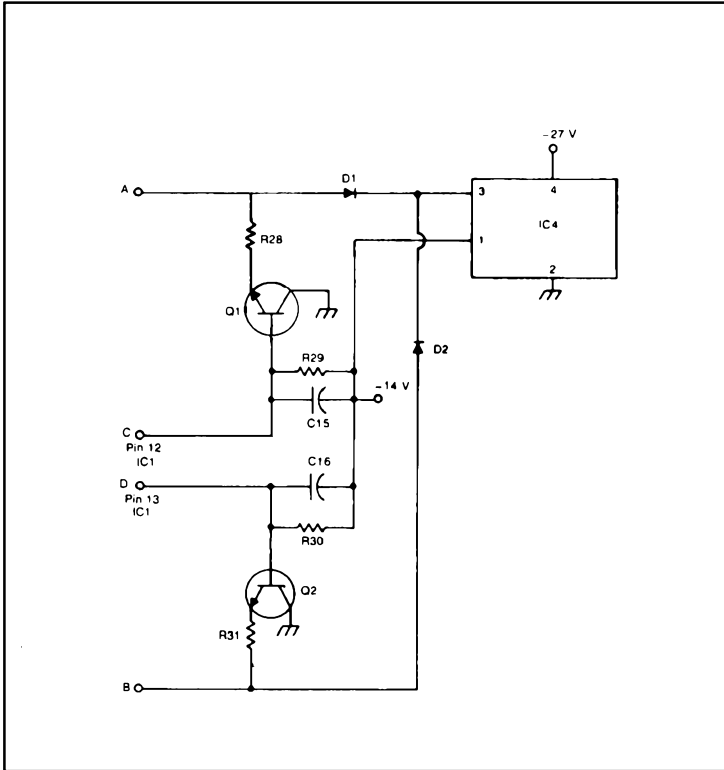


Fig. 8-47. Automatic drummer.

a master rhythm pattern generator. A block diagram for this is illustrated in Fig. 8-48.

Another possibility would be to combine the MM5871 rhythm pattern generator with a 50240 top octave synthesizer (see Chapter 7) and a few digital gates to create a rhythmic melody sequencer. A block diagram for this idea is shown in Fig. 8-49. Rotary switches could be used, so any note can be selected at each trigger point.

I haven't had a chance to breadboard either of these ideas yet, but I see no reason why they shouldn't work out just fine.



OTHER AUTOMATIC CONTROLLERS

In this chapter, we have covered most of the more commonly used automatic controllers in electronic music synthesizers. But there is still plenty of room for your own original experimentation. For example, how about recording trigger pulses on a tape? Obviously this would allow you to record any rhythmic pattern you choose. By using a stereo recorder, you could have two independent sets of trigger pulses to drive separate modules. If you have access to a quadrophonic tape recorder you could set up to four channels of trigger pulses.

Table 8-16. Parts List for Fig. 8-47.

R1, R3, R28, R31	100 k Ω resistor
R2	500 k Ω potentiometer
R4	10 k Ω resistor
R5	33 k Ω resistor
R6, R12, R14	1 M Ω resistor
R7	2.2 k Ω resistor
R8	27 k Ω resistor
R9	18 k Ω resistor
R10	250 k Ω potentiometer
R11	180 k Ω resistor
R13	680 Ω resistor
R15, R16	2.5 k Ω potentiometer
R17	8.2 k Ω resistor
R18	1 k Ω resistor
R19	3.9 k Ω resistor
R20	3.3 k Ω resistor
R21	6.2 k Ω resistor
R22	820 Ω resistor
R23	2.5 k Ω potentiometer
R24	100 k Ω potentiometer
R25	4.7 M Ω resistor
R26	1.5 k Ω resistor
R27, R29	220 k Ω resistor
R30	2.2 M Ω resistor
C1, C2, C4, C5, C7, C8	0.0047 μ F capacitor
C3, C6, C9, C10	0.022 μ F capacitor
C11, C12	0.01 μ F capacitor
C13, C14	0.0068 μ F capacitor
C15, C16	0.1 μ F capacitor
D1, D2	silicon diode (1N4148, or similar)
Q1, Q2	NPN transistor
IC1	MM5871 rhythm pattern generator IC
IC2	LF353 preamp IC
IC3	324 quad op amp IC
IC4	MM5837 noise generator IC

Many experimenters have been controlling their synthesis equipment with computers. This exciting avenue of research will be touched upon in Chapter 11.

What additional ideas can you come up with? Just because I haven't mentioned something here, don't assume you can't use it. Remember the whole idea is to create your own customized electronic music synthesizer!

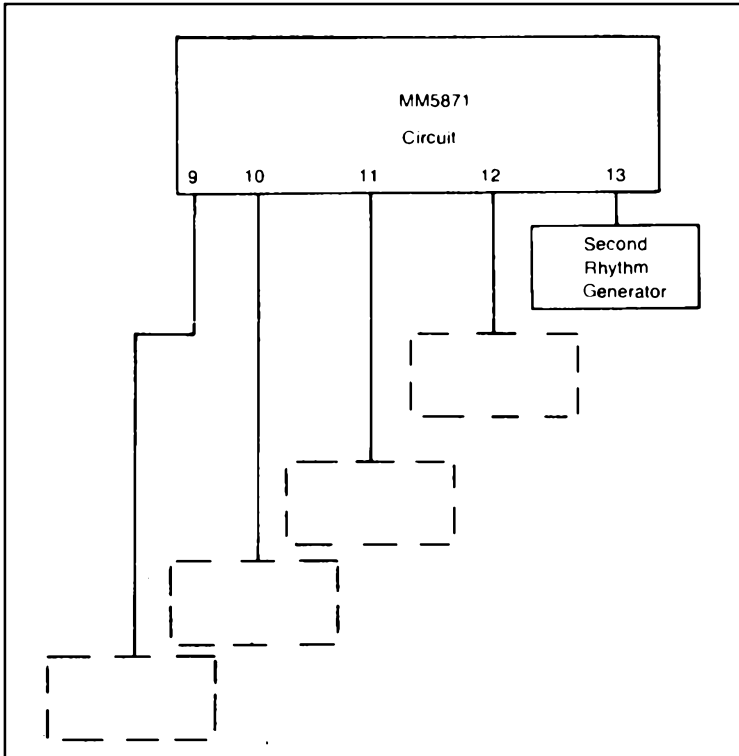


Fig. 8-48. Cascaded rhythm generators.

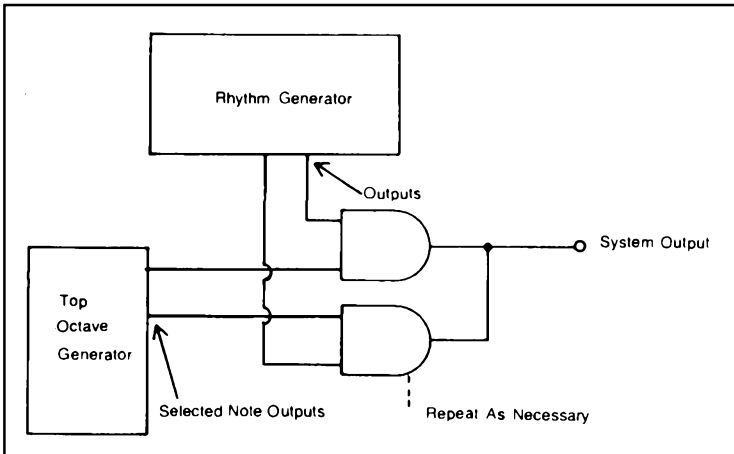
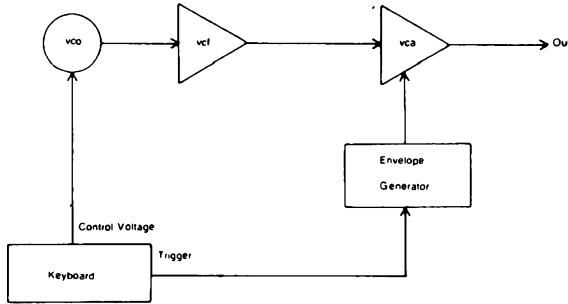


Fig. 8-49. Rhythmic melody sequencer.

Chapter 9



Other Useful Circuits

We have covered most of the standard types of synthesizer modules. However, there are many additional circuits that don't fit into any of the basic categories of the previous chapter, but could be very useful to the electronic musician. In this chapter, we will explore a few of these special purpose devices.

You should keep up with the various electronics magazines and go through as many circuit books as you can get ahold of for additional ideas. You will be surprised at how many devices can be applied to electronic music synthesis. Don't just concentrate on projects that are specifically designed for sound synthesis. Almost any circuit that generates or in some way manipulates a signal in the audible range can be put to work in your synthesizer. Also be on the look out for circuits that generate changing dc voltage of various patterns for control voltage purposes.

One of the nice things about building a modular electronic music synthesizer, is that it is never 100 percent complete. You can continuously add more modules as you discover (and/or can afford) them.

INVERTER

Sometimes you will need a mirror image version of some signal. This effect can be achieved by using a circuit known as an inverter. Figure 9-1 shows some typical waveforms and their inversions. Inverting a signal is the same thing as changing its phase by

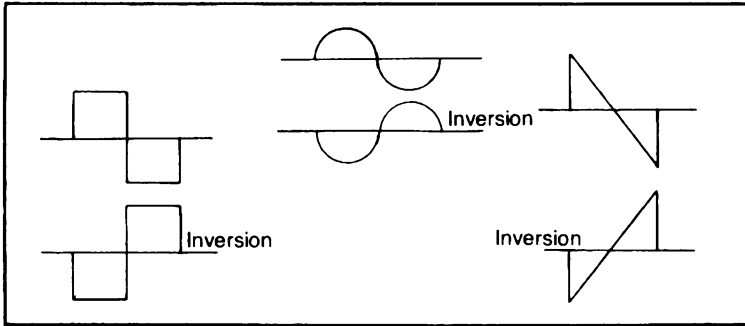


Fig. 9-1. Inverting typical waveforms.

180°. Dc signals, such as envelopes, can also be inverted. The polarity, as well as the direction of voltage change, will be reversed. Figure 9-2 shows the effect of inverting a typical envelope.

You should recall that an op amp has two inputs: an inverting input and a noninverting input. We can take advantage of the inverting input of an op amp to build a practical inverter circuit. The schematic is shown in Fig. 9-3, and the parts list is given in Table 9-1. Only three components are required. The two resistors should have equal values so the signal level will be neither amplified, nor attenuated.

DIFFERENCE AMPLIFIER

Another circuit that takes advantage of the inverting and

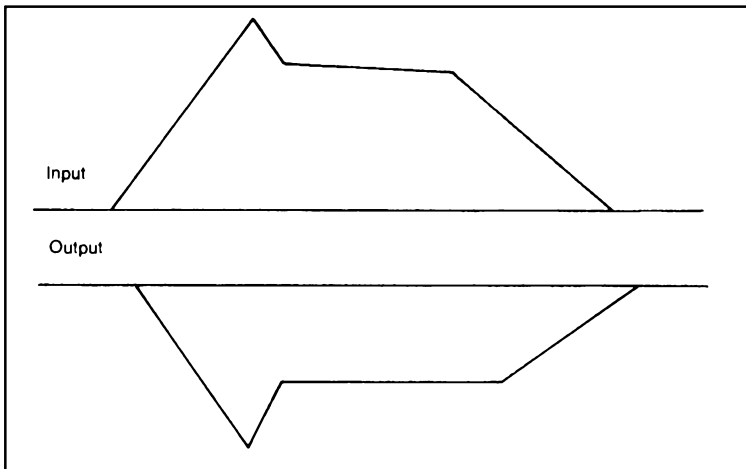


Fig. 9-2. Inverting a typical envelope.

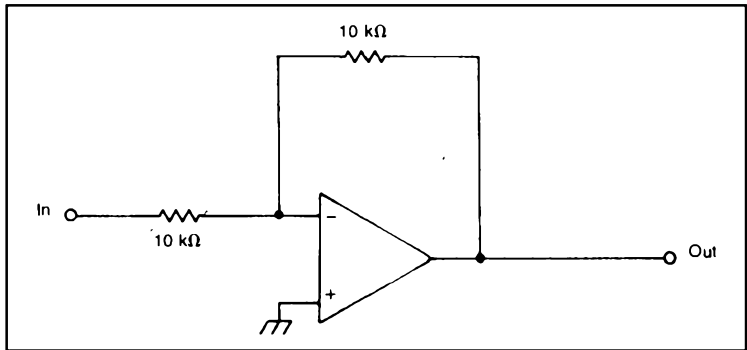


Fig. 9-3. Inverter.

noninverting inputs of the op amp is shown in Fig. 9-4. The parts list is Table 9-2. This circuit is a difference amplifier. The output will be the difference between the two inputs. This is illustrated in Fig. 9-5.

Potentiometers R1 and R4 allow you to set the input levels of

Table 9-1. Parts List for Fig. 9-3.

R1, R2	10 kΩ resistor
IC1	741, or any op amp IC

each of the signals before the difference is determined by the op amp. This allows you to create a wide range of unique effects.

You should be aware that if signal 1 is greater than signal 2, the output will be 180° out of phase, as in the inverter circuit discussed above.

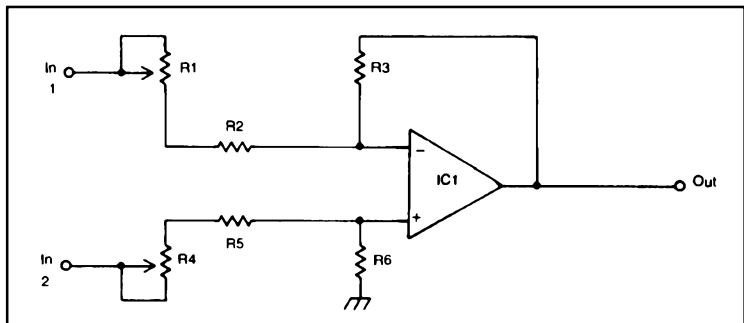


Fig. 9-4. Difference amplifier.

Table 9-2. Parts List for Fig. 9-4.

R1, R4	100 k Ω potentiometer
R2, R5	10 k Ω resistor
R3	120 k Ω resistor
R6	100 k Ω resistor
IC1	741 or any op amp IC

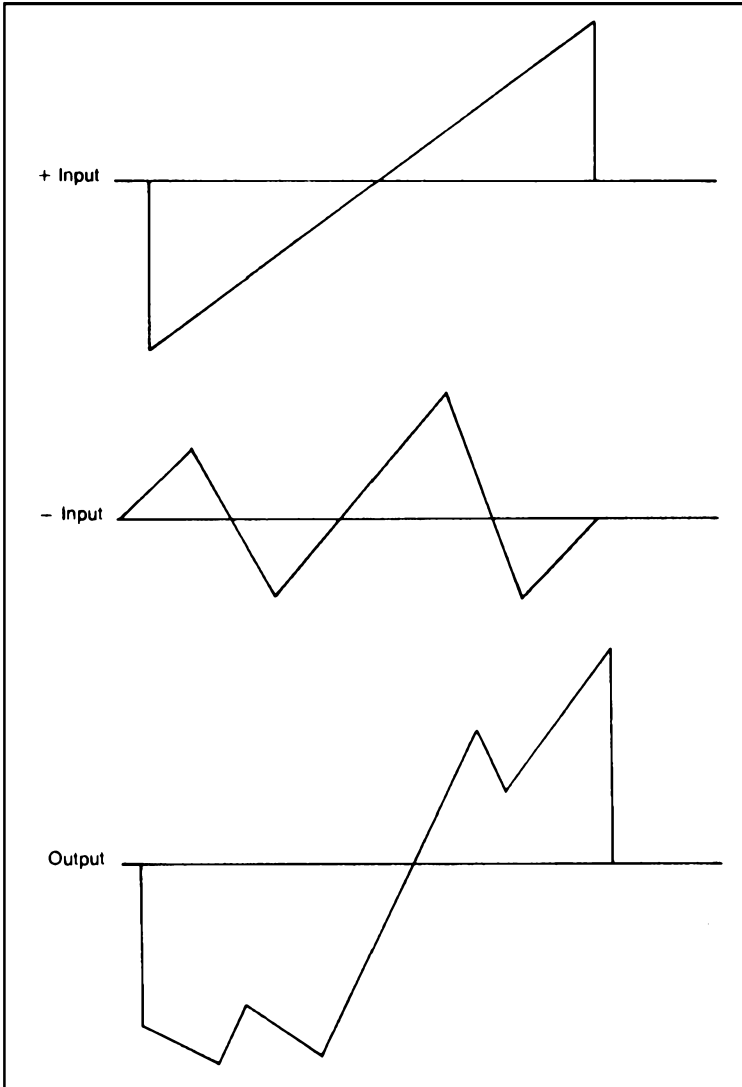


Fig. 9-5. Typical signals for the difference amplifier.

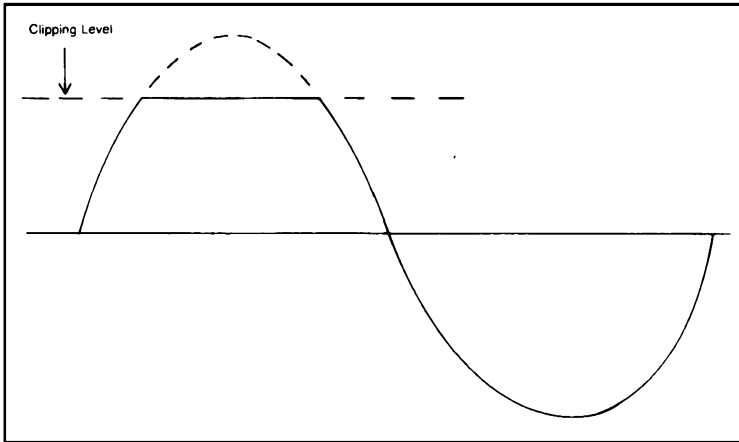


Fig. 9-6. Clipping.

SIGNAL CLIPPERS

A number of different sounds can be created by clipping a high level signal. A clipping circuit is one that leaves low level signals alone, but will not allow the output signal to exceed a specific value. This effect is illustrated in Fig. 9-6.

A very simple passive clipping circuit is shown in Fig. 9-7. The parts list is given in Table 9-3. Only three components (two diodes and a resistor) are required. The diodes must be germanium types. Do not use silicon diodes.

If you want to experiment, you could use zener diodes in this circuit. The zeners' breakdown voltage will determine the clipping point.

An active clipping circuit is shown in Fig. 9-8. This device is known as a clipping amplifier. The amount of gain is set by potentiometer, R2. Changing this potentiometer's setting changes the

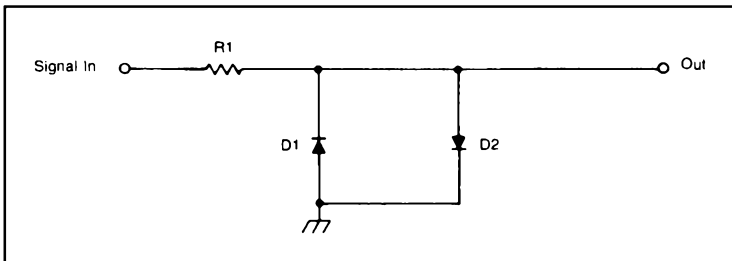


Fig. 9-7. Passive clipper.

Table 9-3. Parts List for Fig. 9-7.

R1	1 k Ω resistor
D1, D2	almost any germanium diode

level of the signal, and thus, the amount of clipping. The clipping point can also be altered by substituting zener diodes with other breakdown voltages for the one specified in the parts list (see Table 9-4).

RING MODULATOR

In the early days of electronic music, ring modulation was an extremely popular effect. It was so popular, in fact, that it has become something of a cliché, and has fallen into disfavor. If used sparingly and tastefully, however, ring modulation can still produce some effective sounds.

Ring modulation is a variation on amplitude modulation (see Chapter 6). Amplitude modulation, you should recall, produces phantom signals known as sidebands. These sidebands have frequencies equal to the sum and difference of the two input signals. For example, if we amplitude modulate a 500 Hz sine wave with a 750 Hz sine wave, the output will consist of four frequencies:

250 Hz	difference
500 Hz	carrier
750 Hz	program
1250 Hz	sum

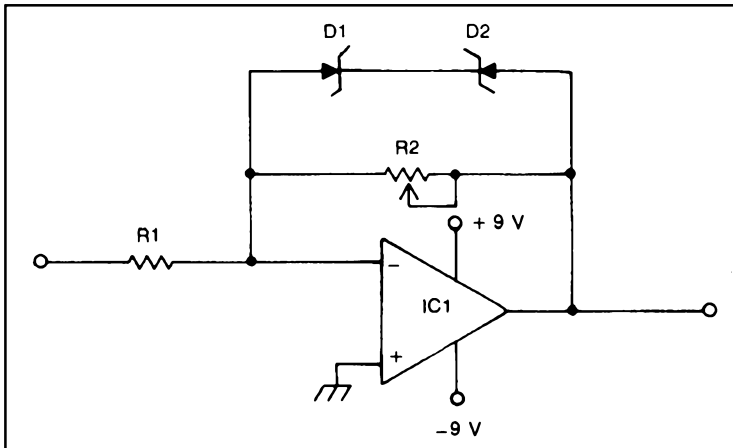


Fig. 9-8. Clipping amplifier.

Table 9-4. Parts List for Fig. 9-8.

R1	100 k Ω resistor
R2	500 k Ω potentiometer
D1, D2	3 volt zener diode (see text)
IC1	741, or other op amp IC

Sum and difference sidebands are also produced by ring modulation. The original input frequencies, however, are suppressed in the output. This means that if we ring modulate a 500 Hz sine wave with a 750 Hz sine wave, the output will consist of two new frequencies:

250 Hz	difference
1250 Hz	sum

Ring modulation creates an odd, nonharmonic, clanging sound quality. The effect tends to sound very unnatural and electronic. These observations are especially true when complex waveshapes are ring modulated. Table 9-5 describes a typical example.

A very simple ring modulation circuit is illustrated in Fig. 9-9. Table 9-6 gives the parts list. Only half a dozen components are required for this passive circuit.

Because this is such a simple, passive circuit, the input frequencies will not be completely blocked, but they will be greatly attenuated, and the output signal will sound very much like traditional ring modulation.

Let me repeat—use ring modulation sparingly. It has been grossly overused in electronic music in the past.

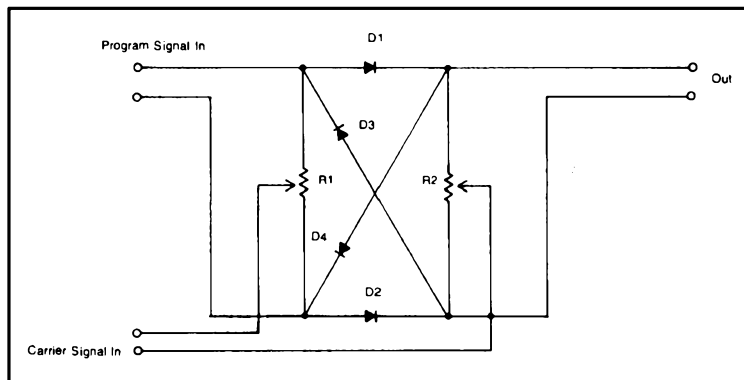


Fig. 9-9. Simple ring modulator.

Table 9-5. Ring modulation example.

Input	Output
300 Hz (program) 500 Hz (carrier)	200 Hz difference—fundamental
1000 Hz (2nd)	700 Hz difference—2nd 800 Hz sum—fundamental
1500 Hz (3rd)	1200 Hz difference—3rd 1300 Hz sum—2nd
2000 Hz (4th)	1700 Hz difference—4th 1800 Hz sum—3rd
2500 Hz (5th)	2200 Hz difference—5th 2300 Hz sum—4th
3000 Hz (6th)	2700 Hz difference—6th 2800 Hz sum—5th
3500 Hz (7th)	3200 Hz difference—7th 3300 Hz sum—6th
and so forth . . .	

Table 9-6. Parts List for Fig. 9-9.

R1, R2	500 k Ω potentiometer
D1, D2, D3, D4	silicon diodes (1N4148, or similar)

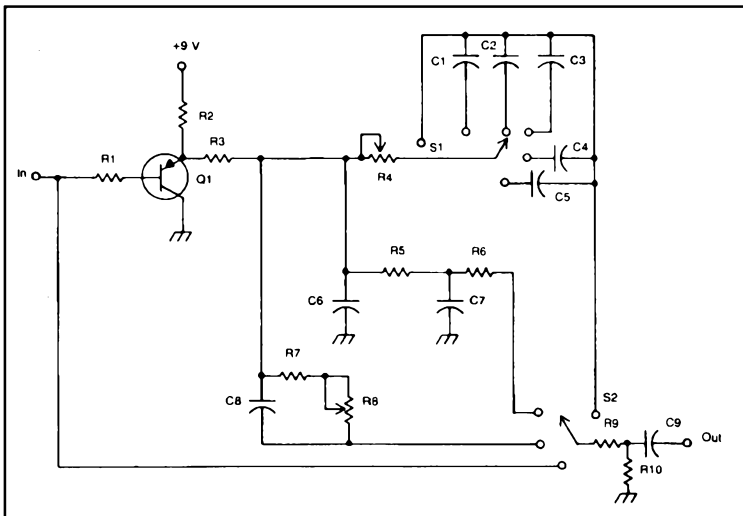


Fig. 9-10. Voicing circuit 1.

VOICING BOX

Figure 9-10 shows the schematic for a simple, but versatile voicing circuit. The parts list is given in Table 9-7.

This circuit can alter a signal in a number of ways. Switch S2 selects which of four possible paths the signal will take. One position simply bypasses the entire circuit. The other three positions allow the signal to pass through a simple preamplifier circuit (Q1, R1, R2) and then through one of three resistor/capacitor networks. R3 is common to all three networks.

Network one lets the signal flow through R3, potentiometer R4, and a capacitor selected by Switch S1. One position of S1 selects no capacitor, so the two resistors act as a simple attenuator. The other five positions will produce various effects depending on the capacitor selected and the setting of potentiometer R4.

The next possible signal path is a simple T type passive filter made up of R3, R5, R6, C6, and C7.

The final possible signal path is through R3, R7, R8, and C8. This is basically a passive clipping/fuzz circuit (discussed shortly). The degree of the effect is determined by the setting of potentiometer R8. You could try experimenting with different values for capacitor C8.

R9, R10, and C9 prevent the next stage from loading down these passive circuits. Despite the preamplifier stage and its power supply, these voicing circuits are purely passive devices. The preamplifier will make up for their losses, but the effect of each path will vary with the signal's frequency. This circuit is best used with signals that are kept within a fairly limited range of frequencies.

Table 9-7. Parts List for Fig. 9-10.

R1	100 k Ω resistor
R2	1.2 k Ω resistor
R3, R5, R6, R7, R10	1 k Ω resistor
R4, R8	10 k Ω potentiometer
R9	3.3 k Ω resistor
C1	0.01 μ F capacitor
C2	0.022 μ F capacitor
C3	0.033 μ F capacitor
C4, C6, C7, C8, C9	0.047 μ F capacitor
C5	0.068 μ F capacitor
S1	single pole/six throw rotary switch
S2	single pole/four throw rotary switch
Q1	PNP transistor (almost any type— 2N3251, 2N3906, GE-21, Motorola HEP-715, Radio Shack RS2034, or similar)

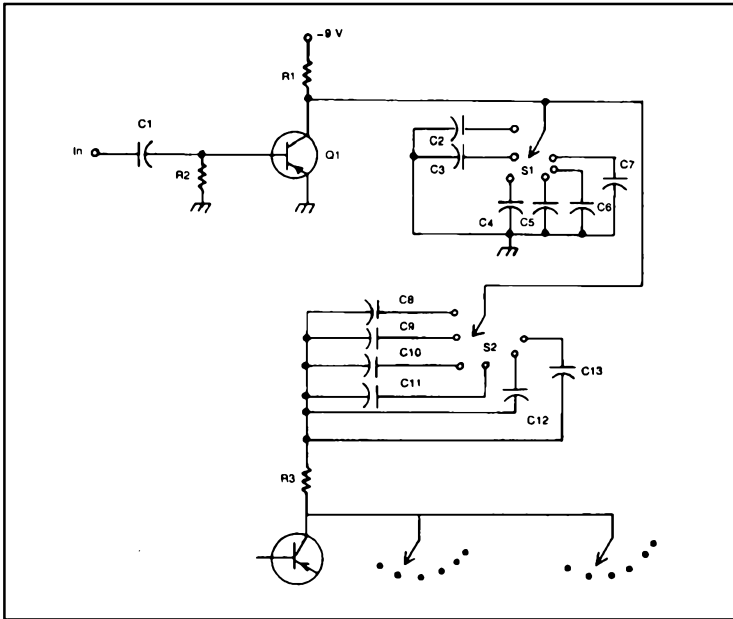


Fig. 9-11. Voicing circuit 2.

Another voicing circuit is illustrated in Fig. 9-11, with the parts list given in Table 9-8. This module will add harmonics to a signal. If the input is a square wave, the output will resemble a sawtooth wave. Other waveshapes at the input will create various oddball waveforms at the output.

Table 9-8. Parts List for Fig. 9-11.

R1	1 k Ω resistor
R2	10 k Ω resistor
R3	50 k Ω potentiometer
C1, C5, C9	0.01 μ F capacitor
C2, C8	0.0047 μ F capacitor
C3	0.0068 μ F capacitor
C4	0.0082 μ F capacitor
C6	0.015 μ F capacitor
C7, C10	0.022 μ F capacitor
C11	0.047 μ F capacitor
C12	0.1 μ F capacitor
C13	0.47 μ F capacitor
S1, S2	single pole/six throw rotary switch
Q1	PNP transistor (2N3906, or similar)

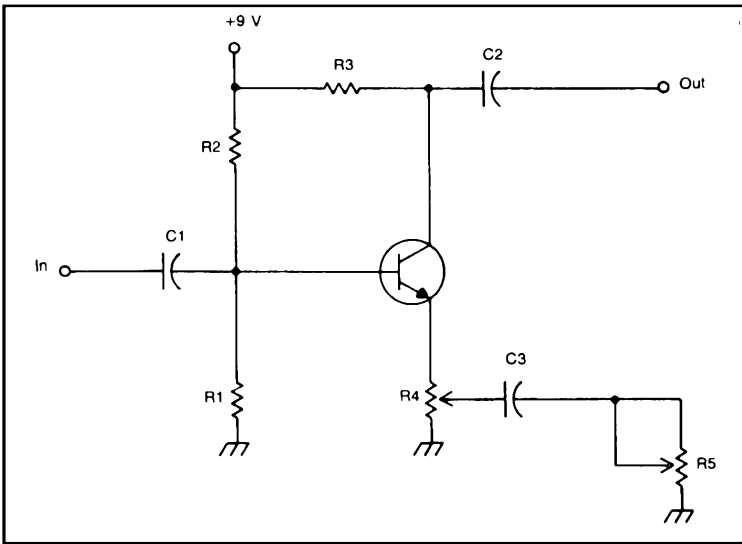


Fig. 9-12. High frequency brightener.

SIGNAL BRIGHTENERS

Some times you may be working with a signal that sounds a little dull. In this section, two circuits for brightening up a flat signal will be presented.

The schematic for the first is shown in Fig. 9-12. The parts list is given in Table 9-9. This circuit will work especially well on high frequencies, brightening the upper harmonics of a complex signal. Potentiometer R4 determines the intensity of the effect, and potentiometer R5 sets the output amplitude of the signal. A fixed resistor could be substituted for R5 if you will not need variable output levels.

Another signal brightener circuit is illustrated in Fig. 9-13. Table 9-10 is the parts list for this circuit.

Table 9-9. Parts List for Fig. 9-12.

R1	62 k Ω resistor
R2	120 k Ω resistor
R3	6.8 k Ω resistor
R4	5 k Ω potentiometer
R5	1 k Ω potentiometer
C1	0.22 μ F capacitor
C2	0.47 μ F capacitor
C3	0.15 μ F capacitor
Q1	NPN transistor (2N3904, or similar)

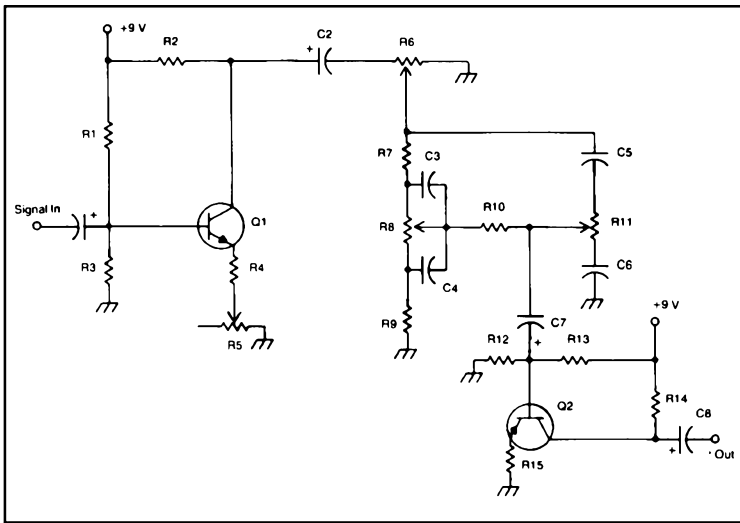


Fig. 9-13. All signal brightener.

This second circuit works pretty much across the entire audible frequency band. Potentiometer R8 controls the amount of brightening for the lower frequencies, while potentiometer R11 determines the effect on high frequencies. Potentiometer R6 serves as an overall level control.

The trimpot (R5) is a set and forget type adjustment. Adjust this control until the circuit breaks into oscillation, then back up the setting to the point where the oscillation just stops. This will set the

Table 9-10. Parts List for Fig. 9-13.

R1, R13	1 M Ω resistor
R2, R14	47 k Ω resistor
R3, R12	100 k Ω resistor
R4	1.8 k Ω resistor
R5	10 k Ω trimpot
R6, R8, R11	100 k Ω potentiometer
R7	22 k Ω resistor
R9	2.2 k Ω resistor
R10	15 k Ω resistor
R15	4.7 k Ω resistor
C1, C2, C7, C8	1 μ F capacitor (see text)
C3	0.01 μ f capacitor
C4	0.1 μ F capacitor
C5	0.0047 μ F capacitor
C6	0.047 μ F capacitor
Q1, Q2	NPN transistor (2N3904, 2N3391, or similar)

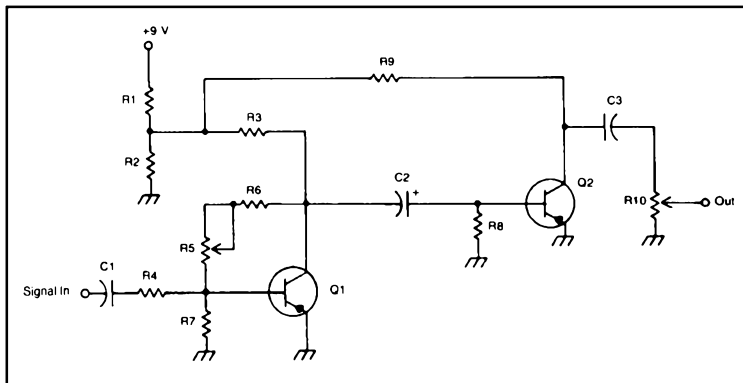


Fig. 9-14. Distortion booster 1.

circuit for optimum performance. To prevent this setting from drifting, you might want to put a drop of paint on the trimpot to hold it in position.

For best results, capacitors C1, C2, C7, and C8 should be tantalum types. Electrolytics may or may not work (depending on the leakage of the specific capacitors used), and could make the circuit extremely temperamental and difficult to use.

DISTORTION BOOSTERS

In most audio equipment, distortion is something to be carefully avoided. Distortion implies that the output signal is not as pure as the input signal. Since a distorted signal can have a distinctive sound quality of its own, controllable distortion can be highly desirable to the electronic musician. For that reason, several circuits specifically designed to create distortion will be presented here.

Table 9-11. Parts List for Fig. 9-14.

R1	1.5 k Ω resistor
R2	3.3 k Ω resistor
R3, R9	10 k Ω resistor
R4, R8	22 k Ω resistor
R5	1 M Ω potentiometer
R6	100 k Ω resistor
R7	18 k Ω resistor
R10	50 k Ω potentiometer
C1	0.1 μ F capacitor
C2	4.7 μ F electrolytic capacitor
Q1, Q2	PNP transistor (Radio Shack RS2004, Motorola HEP-632, or similar)

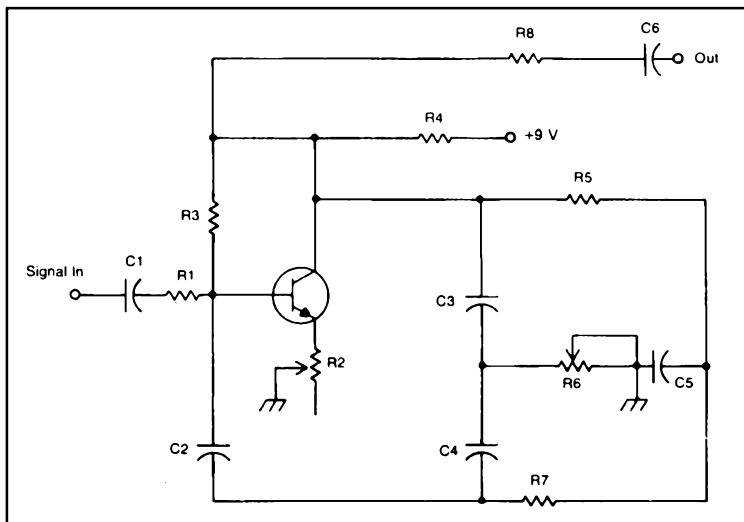


Fig. 9-15. Distortion booster 2.

The first of these distortion booster circuits is shown in Fig. 9-14. The parts list is given in Table 9-11.

This circuit will give most signals a raspy, dirty quality. The degree of distortion is adjustable by potentiometer R5. At fairly low settings, the effect will be rather reminiscent of tube amps. This circuit is quite similar to the fuzz boxes that are so popular with the electric guitarist. The fuzz effect is somewhat cliched on the guitar, but it can be very effective when used with other (synthesized) sounds. Potentiometer R10 allows you to set the output level.

Figure 9-15 shows another distortion booster circuit. The parts list is given in Table 9-12. Potentiometer R6 determines the amount of distortion in the output signal.

Table 9-12. Parts List for Fig. 9-15.

R1, R8	22 k Ω resistor
R2	2.5 k Ω trimpot
R3	470 k Ω resistor
R4, R5, R7	56 k Ω resistor
R6	10 k Ω potentiometer
C1, C6	0.1 μ F capacitor
C2, C5	0.01 μ F capacitor
C3	0.47 μ F capacitor
C4	0.022 μ F capacitor
Q1	NPN transistor (Motorola HEP-722, Radio Shack RS2016, or similar)

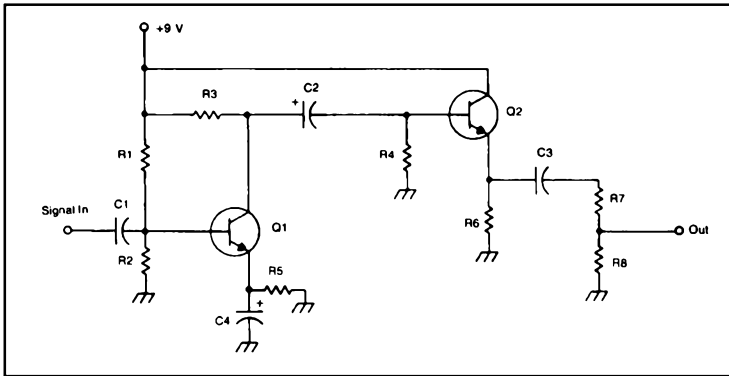


Fig. 9-16. Distortion booster 3.

Trimpot R2 is included to compensate for slight differences between transistors used for Q1. This is a set and forget type adjustment. The trimpot should be set so that the circuit begins to oscillate, then back off from this setting until the oscillation just stops. Apply a dab of paint or nail polish to hold the trimpot in place.

Yet another distortion booster circuit is illustrated in Fig. 9-16. Table 9-13 is the parts list for this circuit.

No manually adjustable controls are included in this circuit as it is shown here, but you should feel free to experiment with different resistor and capacitor values on a breadboard. Try changing R1, R3, R4, C1, and/or C2. Do not change the value of R5, however, or transistor Q1 might be damaged.

ROTATING SPEAKER SIMULATOR

You may have heard of the Doppler effect. This effect is best

Table 9-13. Parts List for Fig. 9-16.

R1, R4	1 M Ω resistor
R2	22 k Ω resistor
R3	10 k Ω resistor
R5	18 Ω resistor
R6	4.7 k Ω resistor
R7	2.2 k Ω resistor
R8	2.7 k Ω resistor
C1	0.001 μ F capacitor
C2	1 μ F electrolytic capacitor
C3	0.0047 μ F capacitor
C4	2.2 μ F electrolytic capacitor
Q1, Q2	NPN transistor (Motorola HEP-722, Radio Shack RS2016, or similar)

described by an example. If you are standing by a railroad track, the pitch of the engine's whistle will seem to get higher as the train approaches you, and lower as it moves away from you. This is because the speed of the train makes each sound wave shorter as the source approaches the listener, and longer as the distance between the source and the observer increases.

Now imagine that the source repeatedly draws nearer to the listener then withdraws, then comes close again. Assuming the speed of the source is fairly high, an intriguing warbling effect will be added to the sound. One way to achieve this effect would be to continuously rotate the speaker recreating the sound. This effect was made popular on old Leslie organs, and is therefore sometimes referred to as the Leslie effect.

Physically rotating speakers introduces a number of mechanical problems, and for the experimenter, it could prove to be far more trouble than it's worth. Fortunately, the effect is not difficult to simulate electronically. A circuit for electronically simulating the

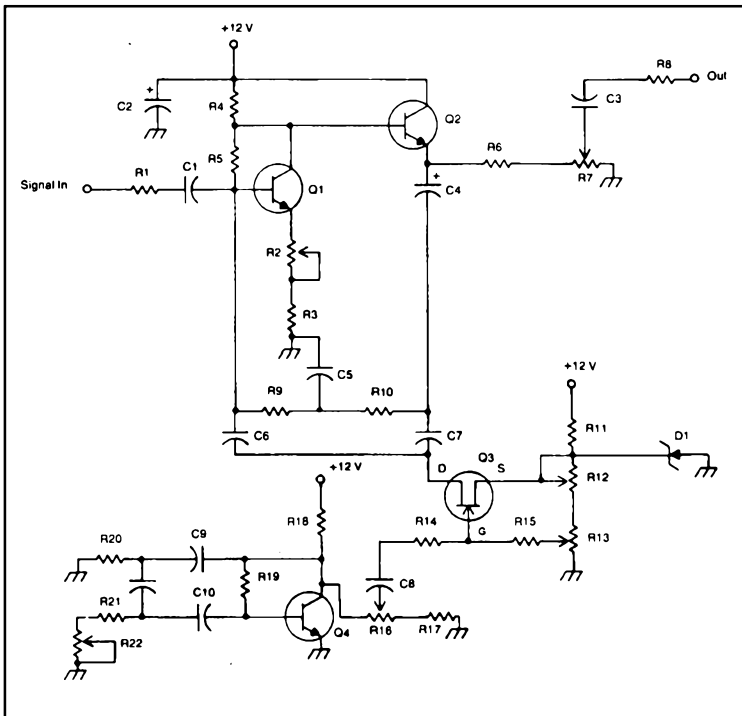


Fig. 9-17. Rotating speaker effect simulator.

Table 9-14. Parts List for Fig. 9-17.

R1, R8, R9, R10	33 k Ω resistor
R2	5 k Ω potentiometer
R3	820 Ω resistor
R4	22 k Ω resistor
R5	1 M Ω resistor
R6	1 k Ω resistor
R7, R13	1 k Ω potentiometer
R11	470 Ω resistor
R12	1 k Ω trimpot
R14, R15	470 k Ω resistor
R16	500 k Ω potentiometer
R17	100 k Ω resistor
R18	82 k Ω resistor
R19	3.9 M Ω resistor
R20	68 k Ω resistor
R21	10 k Ω resistor
R22	100 k Ω potentiometer
C1, C3, C5	0.1 μ F capacitor
C2	100 μ F electrolytic capacitor
C4	4.7 μ F electrolytic capacitor
C6, C7	0.01 μ F capacitor
C8, C9, C10	0.22 μ F capacitor
D1	5.6 volt zener diode
Q1, Q2, Q4	NPN transistor (2N2712, Radio Shack RS2015, or similar)
Q3	FET (MPF102, Motorola HEP-802, Radio Shack RS2036, or similar)

rotating speaker effect is shown in Fig. 9-17. The parts list is given in Table 9-14.

Potentiometer R22 sets the rate of the apparent rotation of the speaker. Other aspects of the effect are determined by the settings of potentiometers R2, R13, and R16. The effects of these controls are rather difficult to describe in print. Suffice it to say that these three controls allow you to create a multitude of variations on the basic effect. Potentiometer R7 simply sets the output level of the signal. This circuit can do far more than an actual physically rotating speaker without any of the mechanical hassles.

SYNTHI-WAA

Another popular effect that was used on electric guitars until it became hopelessly cliched is the waa-waa effect. This effect is named after the sound it creates. Electric guitarists generally used waa-waa circuits that were controlled by foot pedals. By moving his foot, the guitarist could make the sound go "Waaa—Waaa". Some performers could even produce a talking effect with a waa-waa pedal.

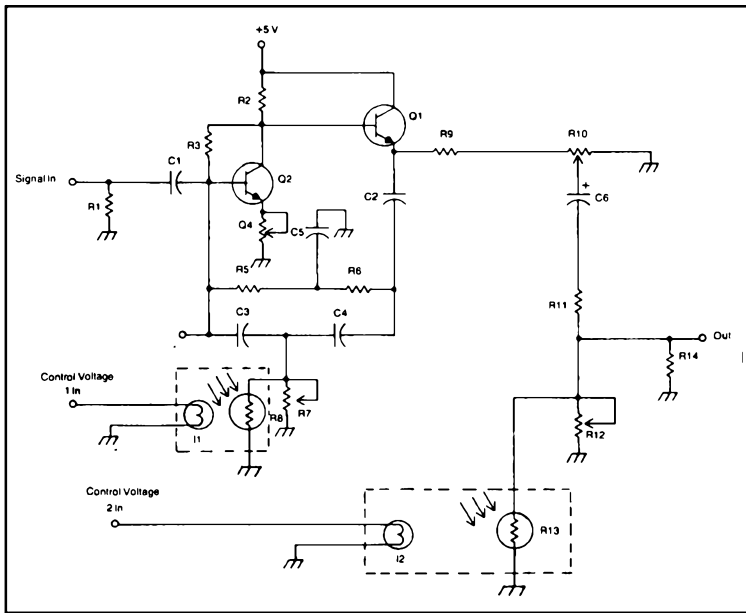


Fig. 9-18. Synthi-waa.

Once again, while the waa-waa effect has been overused on electric guitars, it can still seem fresh and interesting when used on a number of synthesized voices.

A waa-waa circuit that has been adapted for use with an electronic music synthesizer is illustrated in Fig. 9-18. The parts list is given in Table 9-15.

Table 9-15. Parts List for Fig. 9-18.

R1	12 k Ω resistor
R2	22 k Ω resistor
R3	1 M Ω resistor
R4	1 k Ω trimpot
R5, R6	33 k Ω resistor
R7	500 k Ω potentiometer
R8, R13	photoresistor
R9	1 k Ω resistor
R10	1 k Ω potentiometer
R11, R14	15 k Ω resistor
R12	100 k Ω potentiometer
C1, C2, C5	0.1 μ F capacitor
C3, C4	0.0047 μ F capacitor
C6	4.7 μ F electrolytic capacitor
I1, I2	6 volt flashlight bulb (see text)
Q1, Q2	NPN transistor (2N5129, Radio Shack RS2011, or similar)

This module may be controlled either with a manual device or with voltage control. Lamp 11 and R8, or potentiometer R7 control one filter section. The lamp is lit to varying brightnesses by the control voltage fed to it. The lamp and the photoresistor should be enclosed in a light-tight housing.

Similarly, the second filter section is controlled by lamp 12 and photoresistor R13, or by potentiometer R12. (Again the lamp and the photoresistor must be shielded from all extraneous light.) The waa-waa effect is achieved by the variation between these two filter stages. The depth of the effect is controlled by potentiometer R10.

R7 and R12 could be housed in foot pedals. To directly duplicate the traditional electric guitar waa-waa effect, the potentiometers should be mounted in a single pedal that is constructed so that as R7 is increased, R12 is decreased, and vice versa.

Trimpot R4 should be adjusted just below the point where the circuit breaks into oscillation, then it should be left alone.

This circuit can simulate the traditional waa-waa effect of the sixties, or it can be operated so that entirely new effects are achieved. Some fascinating results can be achieved by taking advantage of the voltage control option.

Lamps 11 and 12 could be replaced with LEDs, but control voltages must then be of the correct polarity.

DELAY CIRCUIT

Many fascinating effects can be achieved by combining a signal

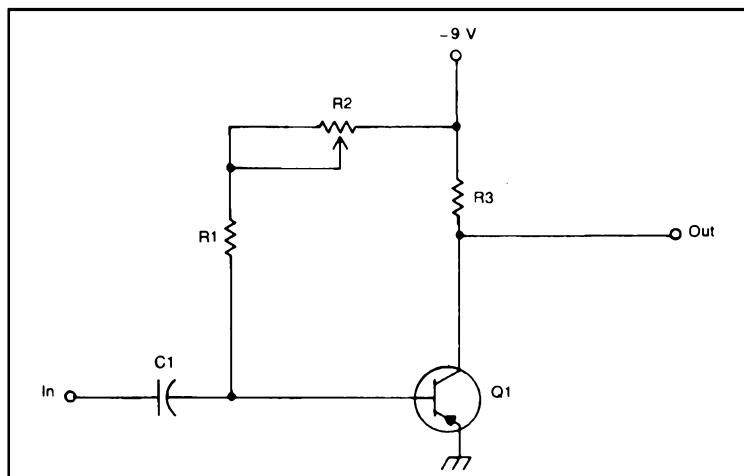


Fig. 9-19. Delay circuit.

Table 9-16. Parts List for Fig. 9-19.

R1	100 Ω resistor
R2	2.5 k Ω potentiometer
R3	10 k Ω resistor
C1	220 pF capacitor
Q1	PNP transistor (Radio Shack RS2005, or similar)

with an out of phase duplicate of itself. One way to achieve this effect is to delay part of the signal.

A very simple delay circuit is shown in Fig. 9-19. The parts list is given in Table 9-16. This circuit will delay the input signal by a fixed amount, set by potentiometer R2. It is not frequency sensitive. If the input frequency is changed, the amount of phase shift will also be changed. At some settings and frequencies, there will be no apparent phase shift. This is not a true phase shifter circuit, but it can be extremely useful in a synthesis system, especially considering its simplicity; only five components are required.

The basic patch for the pseudo phase shift effect is shown in Fig. 9-20. More complex effects could be achieved by using several of these delay circuits, each with a different setting. A patch illustrating this idea is shown in Fig. 9-21.

SCHMITT TRIGGERS

An electronic musician might often find a Schmitt trigger a handy item to have. A *Schmitt trigger* is basically an electronic switch controlled by an input signal. When the input signal exceeds a specific voltage, the schmitt trigger switches on (high output voltage). The circuit remains on until the input signal drops below another specific voltage, at which time the Schmitt trigger is switched off (low output voltage). This is illustrated in Fig. 9-22.

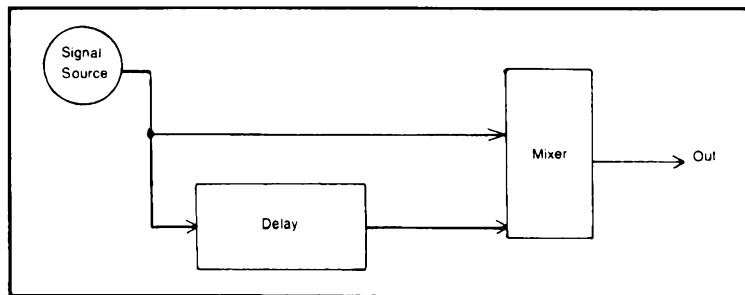


Fig. 9-20. Simple pseudo phase shift patch.

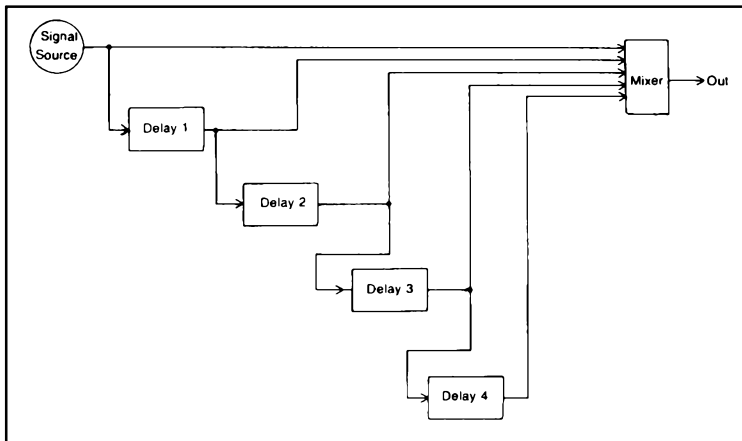


Fig. 9-21. Complex pseudo phase shift patch.

Note that the switch-off voltage is usually somewhat lower than the switch-on voltage. In most practical circuits, there is a dead zone between these two switching levels.

A practical Schmitt trigger circuit is shown in Fig. 9-23. The parts list is given in Table 9-17.

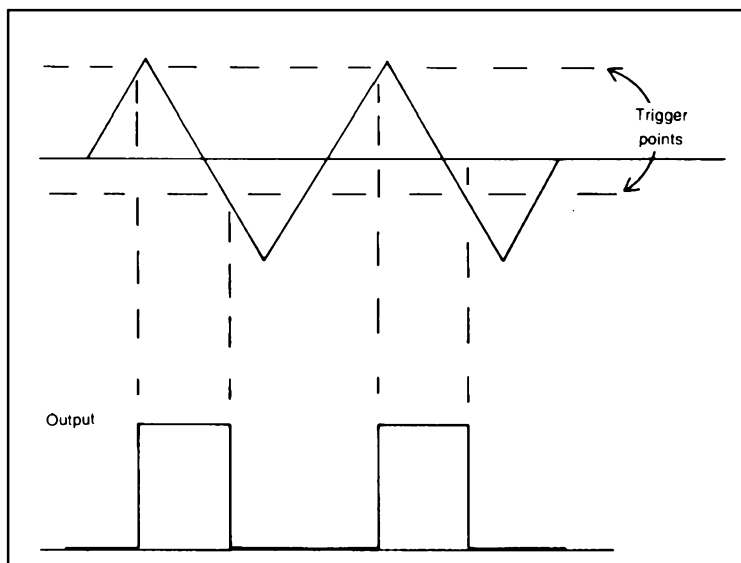


Fig. 9-22. Effect of a Schmitt trigger.

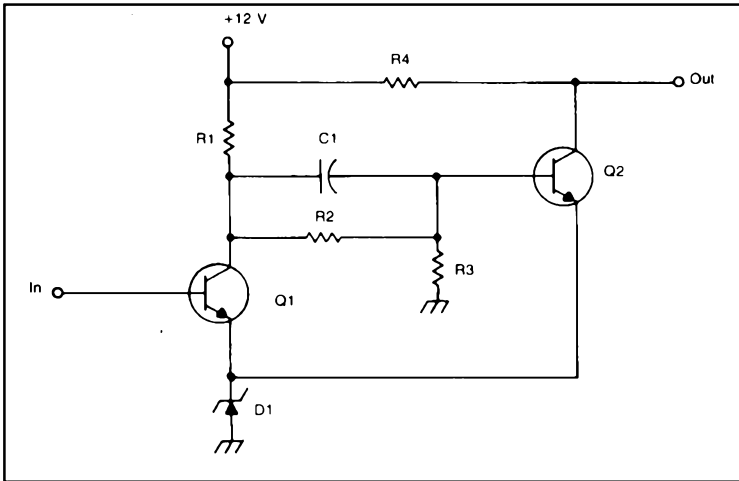


Fig. 9-23. Schmitt trigger.

FREQUENCY DIVIDERS

It is often desirable to duplicate a signal at a lower frequency. The circuit used to accomplish this is called, not surprisingly, a frequency divider.

The easiest type of electronic frequency division is by two. When you divide a frequency by two, you reduce it by exactly one octave. By combining a signal with an octave lower duplicate of itself (as illustrated in Fig. 9-24), a very full, rich tonal quality can be achieved.

A circuit for reducing a frequency by two (one octave) is shown in Fig. 9-25. The parts list is given in Table 9-18.

If you are familiar with digital circuitry, you may recognize this circuit as a simple flip-flop which reverses state each time it is triggered. Each cycle of the input waveform triggers the flip-flop once, so two input cycles are required to generate one complete output cycle.

Table 9-17. Parts List for Fig. 9-23.

R1, R2, R4	2.7 k Ω resistor
R3	18 k Ω resistor
C1	390 pF capacitor
D1	zener diode (1N747, or similar)
Q1, Q2	NPN transistor (2N1306, Radio Shack RS2001, or similar)

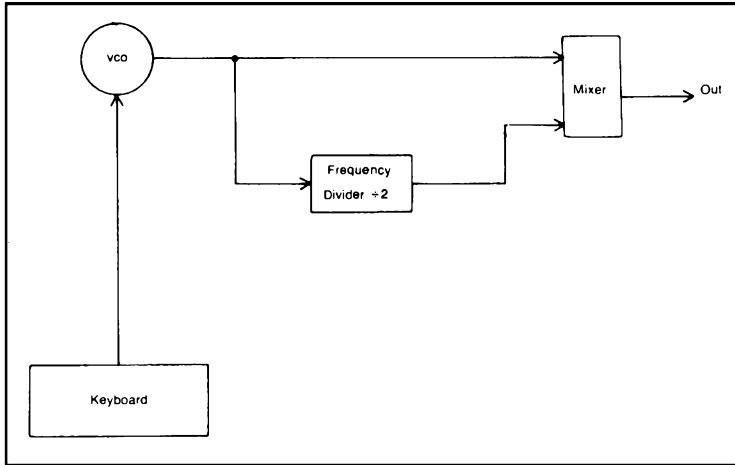


Fig. 9-24. Octaving a signal.

This type of circuit can output only square waves. It works best if the input is a square wave or rectangle wave. If you are working with other signals and find this module is working unreliably, try inserting a Schmitt trigger between the signal source and the frequency divider. A properly adjusted Schmitt trigger can convert any periodic waveform into a rectangle wave.

A more versatile frequency divider circuit is shown in Fig. 9-26. The parts list is given in Table 9-19. This circuit can divide the

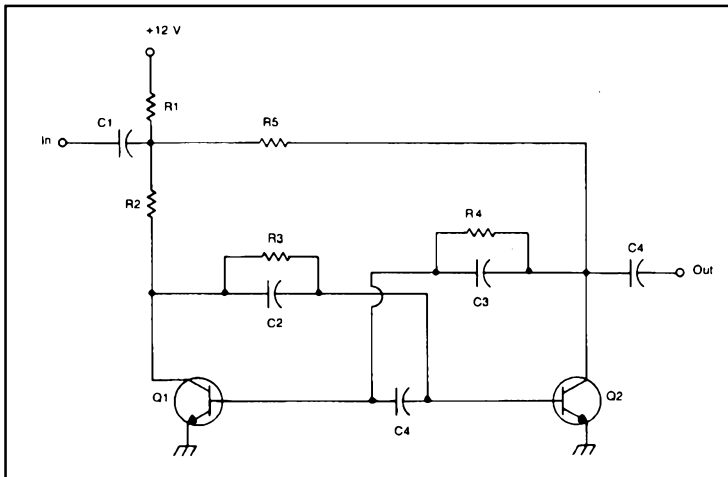


Fig. 9-25. Frequency divider.

Table 9-18. Parts List for Fig. 9-25.

R1, R2, R5	2.2 k Ω resistor
R3, R4	18 k Ω resistor
C1, C2, C3	4700 pF capacitor
C4	1000 pF capacitor
C5	0.1 μ F capacitor
Q1, Q2	NPN transistor (2N2926, Radio Shack RS2031, or similar)

input frequency by any whole number from one through ten by closing the corresponding switch. (S4 controls divide by four, S7 controls divide by seven, and so forth.)

This circuit also outputs only square waves, and may require a Schmitt trigger at the input.

FREQUENCY DOUBLER

You can also multiply an input frequency electronically. A frequency doubler (multiply times two) circuit is illustrated in Fig. 9-27. As the parts list in Table 9-20 shows, only two components are required.

The output of this circuit will always be a square wave. A Schmitt trigger may be required at the input for some waveshapes.

DIGITAL GATES

Various digital gates can be used in an electronic music syn-

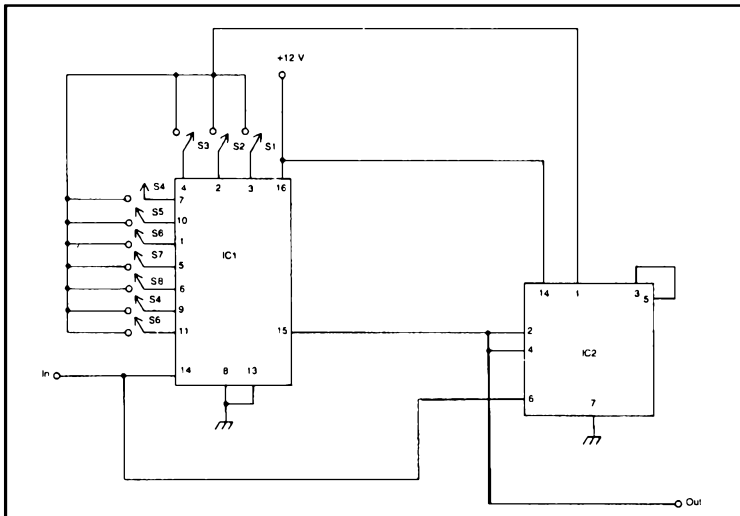


Fig. 9-26. Deluxe frequency divider.

Table 9-19. Parts List for Fig. 9-26.

IC1	CD4017 Decade Counter IC (CMOS)
IC2	CD4001 quad NOR gate IC (CMOS)
S1-S10	SPST switch

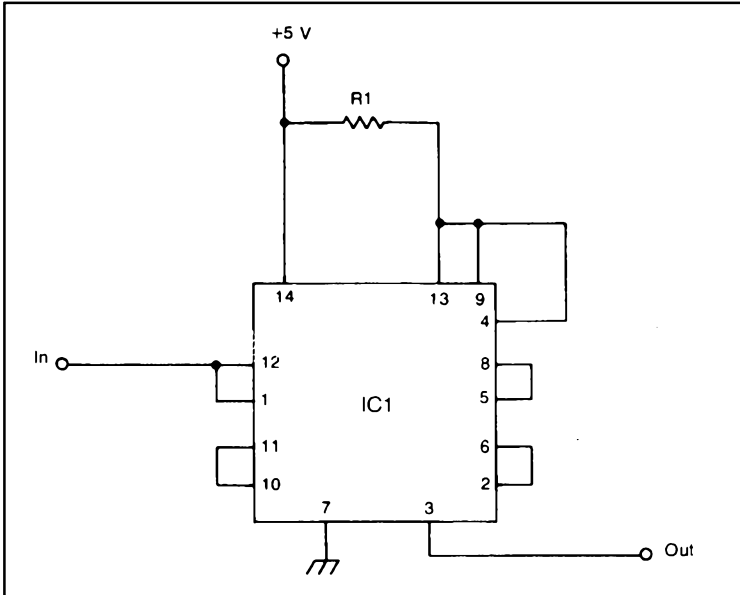


Fig. 9-27. Frequency doubler.

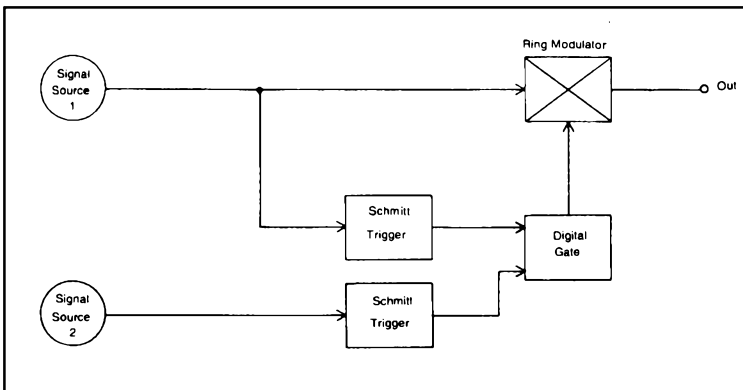


Fig. 9-28. Ring modulation with digital gates.

Table 9-20. Parts List for Fig. 9-27.

R1	1 k Ω resistor
IC1	7486 quad Exclusic OR gate IC (TTL)

thesizer system to create some offbeat special effects.

Digital gates have only two states: on or off. Most have an output determined by the states of two or more inputs.

For most input signals a Schmitt trigger will be necessary to ensure reliable operation. Many digital gates are available in IC form (typically four gates per package). There are several families of digital gate ICs. TTL devices are inexpensive, but are very sensitive to precise power supply and input voltages. A small overvoltage could damage the IC CMOS units are more tolerant of voltage variations, but are susceptible to damage from bursts of static electricity.

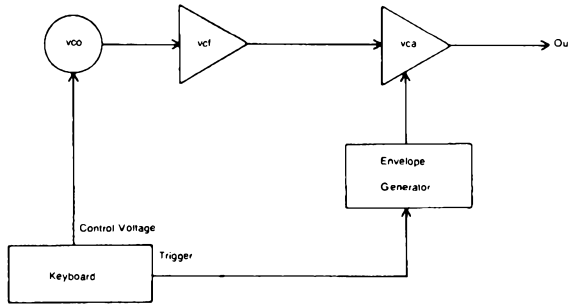
There are a great many ways digital gates may be used in an electronic music synthesis system. For example, several lfos can be combined to produce a single nonperiodic clock signal. One reason why sample and hold and sequencer patches tend to sound a bit mechanical is that each step is of equal length. By combining two or more clock signals with digital gates, a number of unusual and irregular rhythm patterns can be generated. Some may have a definite rhythm feel about them too.

Digital gates can also be used on regular audio range signals, but bear in mind that you will lose almost all sense of pitch in the majority of cases. In some instances, the output will tend to sound like filtered noise. For best results, use harmonically related inputs, or inputs that are very close in frequency, and mix the output with the main input signal to reaffirm the sense of pitch.

An unusual ring modulator patch is shown in Fig. 9-28. This effect can be quite striking, although I'd recommend using it in only small quantities.

There is no reason why you should limit yourself to just the basic gates. A handful of various gates can be combined to create almost any complex gate.

Chapter 10



Putting It All Together

So far in this book, a number of circuits have been presented. You may want to build any or all of these, along with additional projects you find in other sources. In building a customized electronic music synthesizer, you get to choose what will be included.

A collection of mismatched circuit boards and cases of various shapes and sizes would not make a very efficient system. In this chapter, we will explore some of the ways you can package your synthesizer modules. We will also examine some of the ways these circuits can be hooked up together to create various sounds and effects.

HARDWIRED INSTRUMENTS

More and more commercial synthesizers are using the hard-wired format. Connections between the modules are permanent, avoiding the hassle of patch cords or other methods of temporary hookups. Modules for each patch are selected with switches.

Hard-wired synthesizer systems are generally easier for the beginner to use than manually patched instruments. Most settings of the switches and other controls will probably produce some kind of sound. It is difficult to make drastic mistakes.

Another advantage is that you don't have to worry over having enough patch cords of the required type and length. Patch cords can get tangled, and sometimes the cable will break inside the insulation, making the cord unreliable.

Changing voices is generally faster on a hard-wired synthesizer than on a manually patched instrument.

Hard-wired synthesizers tend to be more attractive than a patch panel with several dozen cords hanging out of it. The instrument will also tend to be somewhat smaller (it won't need all that space taken up by jacks) and sturdier. Certainly, a hard-wired instrument is more appropriate for live performance situations, especially where extensive travelling is required.

But a hard-wired synthesizer is not the perfect solution to interconnecting your synthesis modules. There are disadvantages too.

Once a hard-wired system is built, it is difficult to expand it by adding new modules as you discover or can afford them.

By definition, a hard-wired system puts some limitations on the number of ways you can interconnect your modules. Most of the time this will not much matter, but there are exceptional cases. It is hard (if not impossible) to account for every possible patch when designing the switch panel for a hard-wired instrument. You are bound to come up with new ideas at a later date.

As with so many other things in life, you have to explore the various possibilities and decide on the compromise that works best for you. We will discuss hard-wired systems in more detail in the next few pages, then we will take a look at more flexible, but more awkward manually patched systems.

The Standard Basic Patch

While the hard-wired approach sounds extremely limited, and is to a certain extent, it is not nearly as bad as it may sound. Nine out of ten synthesizer patches are likely to be some variation on the basic patch shown in Fig. 10-1.

You should notice that a great deal of information is not presented in this patch diagram. This is, of course, where the possibilities for variations come in. What waveshape is the vco generating? What type of VCF is used? (Low-pass, high-pass, band-pass, or band-reject?) Is the keyboard's trigger signal a gate or a trigger pulse? What are the control settings on the envelope generator? Is it an AR or ADSR type?

There are also many common variations on this standard basic patch. For example, the filter could be controlled by the envelope generator along with the vca as shown in Fig. 10-2. Many acoustic instruments change their harmonic structure over time, and this effect can be synthesized with a filter envelope.

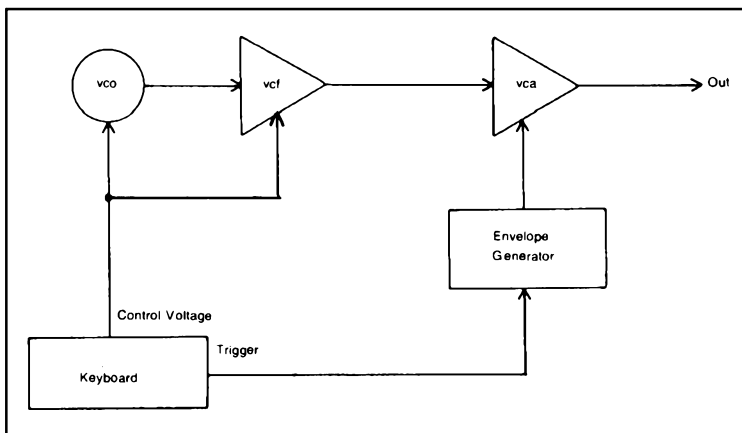


Fig. 10-1. The basic patch.

In many cases, the filter envelope may need to be different from the amplitude envelope. Obviously two envelope generators would be required here. This is illustrated in Fig. 10-3.

Some voices might require more than one filter type, each controlled in a different way. A typical patch of this nature is shown in Fig. 10-4.

Despite their differences, all of these variations are essentially the same basic patch. The wide-spread usefulness of this basic patch format is what makes hard-wired synthesizers practical. Even though some patching combinations will not be possible, the commonly used ones will be available.

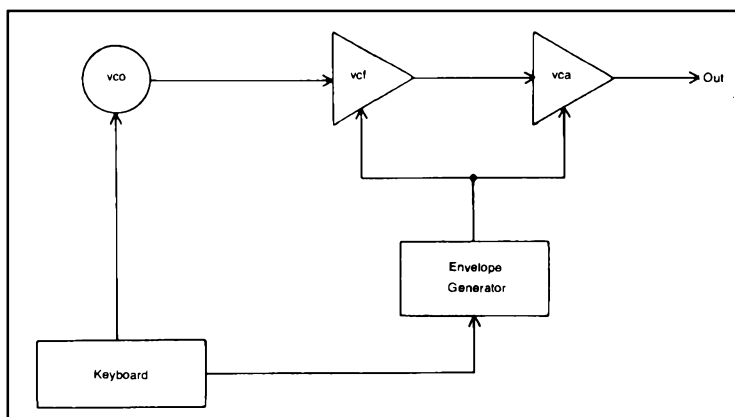


Fig. 10-2. First variation on the basic patch.

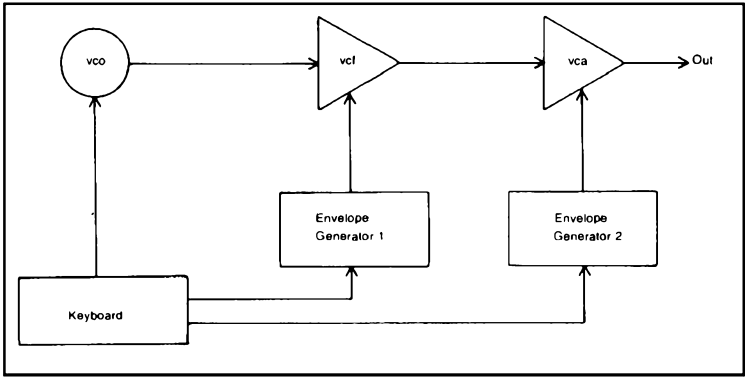


Fig. 10-3. Second variation on the basic patch.

Most hard-wired systems include switches to allow the musician to vary the patching in some way. A switchable basic patch suitable for a hard-wired electronic music synthesizer is illustrated in Fig. 10-5. This setup will allow you to create all of the patches shown in this section, plus many others. The effects of the switches are outlined in Table 10-1.

Other hard-wired patches are possible, especially if you want to include more modules, but this basic example demonstrates the format and the essential concepts involved. Notice how the switches in Fig. 10-5 are arranged to allow for maximum practical versatility. Some possible combinations are ignored; there would be little need to control the vco with an envelope generator, or the vca with the keyboard. These odd patches may come in handy once

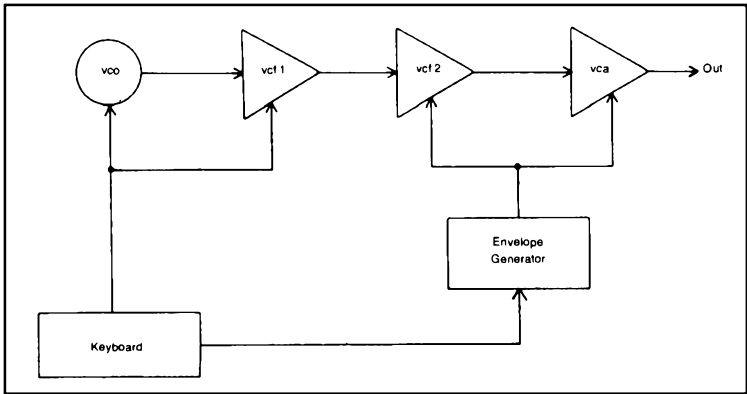


Fig. 10-4. Third variation on the basic patch.

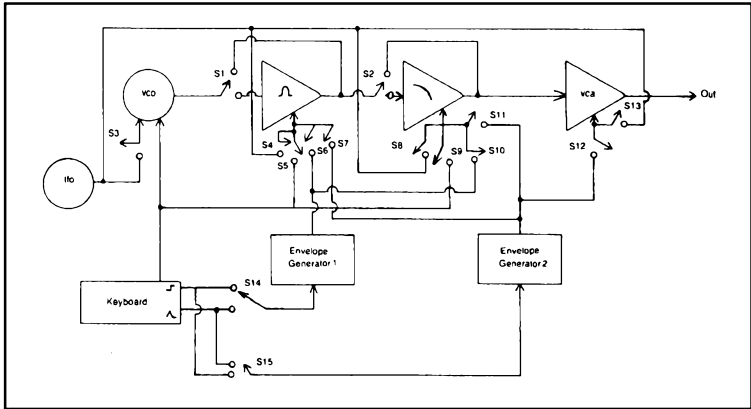


Fig. 10-5. Switchable basic patch suggestion.

Table 10-1. Effects of the Switches in Fig. 10-5.

Switch	Function	On	Off
S1	band-pass filter used?	YES	NO
S2	low-pass filter used?	YES	NO
S3	lfo control VCO?	YES	NO
S4	lfo control band-pass filter?	YES	NO
S5	keyboard control band-pass filter?	YES	NO
S6	envelope generator 1 control band-pass filter?	YES	NO
S7	envelope generator 2 control band-pass filter?	YES	NO
S8	lfo control low-pass filter?	YES	NO
S9	keyboard control low-pass filter?	YES	NO
S10	envelope generator 1 control low-pass filter?	YES	NO
S11	envelope generator 2 control low-pass filter?	YES	NO
S12	envelope generator 2 control vca?	YES	NO
S13	lfo control vca?	YES	NO
S14	trigger envelope generator 1	gate	pulse
S15	trigger envelope generator 2	gate	pulse

Table 10-2. More Effects of the Switches in Fig. 10-5.

Module	Controlled By			
	ffo	Keyboard	Envelope Generator 1	Envelope Generator 2
VCO	X	X	-	-
vcf, band-pass	X	X	X	X
vcf, low-pass	X	X	X	X
vca	X	-	-	X
Envelope Generator 1		gate	trigger pulse	
Envelope Generator 2		gate	trigger pulse	

in awhile, but as a general rule the need is not likely to arise. Remember that the more switches you add, the larger and more confusing your control panel will be. In addition to this, good switches can be fairly expensive. The hard-wired approach implies some level of compromise for the sake of simplicity and ease of operation.

Some Additional Tips On Switching

In setting up your hard-wired patch, there are a number of things to be kept in mind.

Many modules (if not all) will not be required in every patch. Often their effects will be undesirable. Switches to bypass each module should therefore be provided. The wiring is not complex. A SPDT (Single Pole/Double Throw) switch can be used as shown in Fig. 10-6. A simpler SPST (Single Pole/Single Throw) switch can be used in some cases (see Fig. 10-7), but this will often cause some degradation of the sound being synthesized, so this approach should be avoided. Spend a little extra for SPDT switches.

In a few instances, the use of some modules may be mutually exclusive (i.e., these modules will never be used together). In this kind of case, you can use multiposition switches, as illustrated in Fig. 10-8. Double pole switches to control both the inputs and the outputs of the modules, as indicated in the diagram, are often a good idea, although not 100 percent essential in most instances.

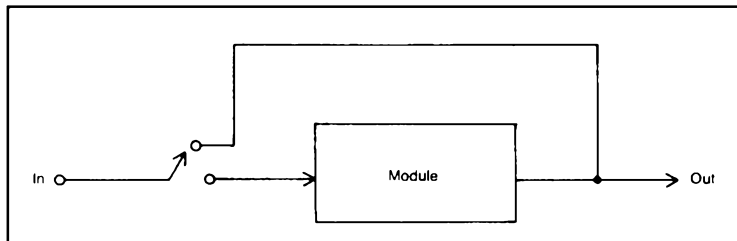


Fig. 10-6. Bypassing modules with SPDT switches.

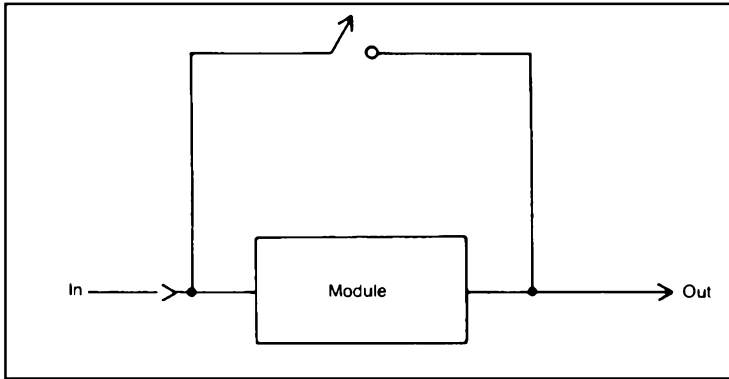


Fig. 10-7. Bypassing modules with SPST switches.

You should give some thought to the kinds of switches you use. Slide switches are OK, but they are difficult to mount, because of the need for a rectangular hole. Also, they can be a bit troublesome to use especially when you have to change a number of settings rapidly.

Rocker switches are convenient to use, although with some it is not always easy to see which position they are in. Again, a rectangular hole is required in the control panel to mount the switch.

Pushbutton switched (spring loaded, or PUSH ON/PUSH OFF) can be mounted in easily drilled round holes and can be placed

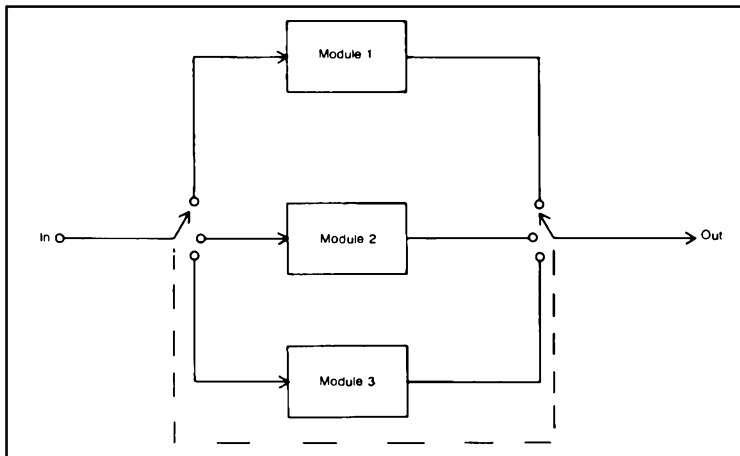


Fig. 10-8. Mutually exclusive module switching.

fairly closely together without creating problems. However, their current position (on or off) is not obvious at a glance. Lighted pushbutton switches are available, or you could mount LEDs, or small bulbs in parallel with each switch, but that can run up the expense of the project very rapidly. Lighted pushbutton switches are pretty bulky too, and of course, adding your own lights (or LEDs) also increases the size of your control panel. Pushbutton switches can have some reliability problems too—especially inexpensive units.

Toggle switches tend to be relatively expensive, but if you can get a good deal on them, they are a good choice. They are reliable, and the position of their shafts is clear, especially if bat-handle type toggle switches are used. Bat-handles are also the easiest to operate in relatively cramped quarters, although standard toggle switch handles don't introduce much of a problem along that line. Finally, toggle switches can be mounted in round holes.

When you need multiposition switches, rotary switches are the most practical choice. A knob is placed on the switch shaft, making the spacing requirements the same as for a potentiometer. The biggest disadvantage of rotary switches is that you can't always find exactly the switch you need. You may end up with extra sections on your switch to get the number you need. If you have a choice, try to let the excess be in the number of poles, rather than the number of positions. For example, if you need a single pole/four position switch, and the closest units available are a single pole/six position switch and a double pole/four position switch, I would strongly recommend that you opt for the double pole/four position unit. Extra positions can cause confusion. A switch might be placed in a dead (unused) position by mistake, and you could waste some time trying to figure out why you're not getting any sound. In some cases, dead switch positions are unavoidable, but when you can avoid them, by all means do so.

Super-Modules

In designing a hard-wired electronic music synthesizer, it is often useful to think in terms of super-modules. A *super-module* is a collection of related modules that can be grouped into a conceptual single unit.

For example, all the original signal sources may be grouped into a single super-module called SOURCE. The exact nature of your SOURCE super-module may vary from anybody else's, but it serves the same function, in this case, it originates a signal. A

SOURCE super-module may include two or three (or more) voltage-controlled oscillators, a noise source, and perhaps an input stage to accept external signals (from a microphone or a traditional musical instrument).

Other typical super-modules might be CONTROL (control voltage sources—keyboard, ribbon controller, joystick, etc.), FILTER (voltage-controlled filters), ENVELOPE (envelope generators, envelope followers, voltage-controlled amplifiers), and MODULATION (low-frequency oscillators for vibrato and tremolo type effects). You might also include a MISCELLANEOUS super-module for effects that don't quite fit into any of the standard categories.

A basic super-module patch is illustrated in Fig. 10-9. This is a simple, easy to follow diagram. You should have no trouble tracing the path of the signal. Moreover, it is applicable to virtually any patch. If individual modules were shown, it would probably be difficult to understand the setup of the synthesizer as a whole.

Super-modules can be diagrammed separately. Figures 10-10 through 10-15 shows possible super-module patches for each of the categories mentioned previously. You may come up with different super-modules that are more directly suitable to your individual needs. The internal structure of your super-modules will probably differ from the samples shown here, but the principle can be applied to almost any desired arrangement.

By numbering, or naming each switch, you can easily make notes on the switch positions required for each patch you want to keep a record of. A typical notation format is used in Table 10-3. You will also need to keep notes on any and all relevant potentiometer settings too. You could have calibrated scales over each knob, as

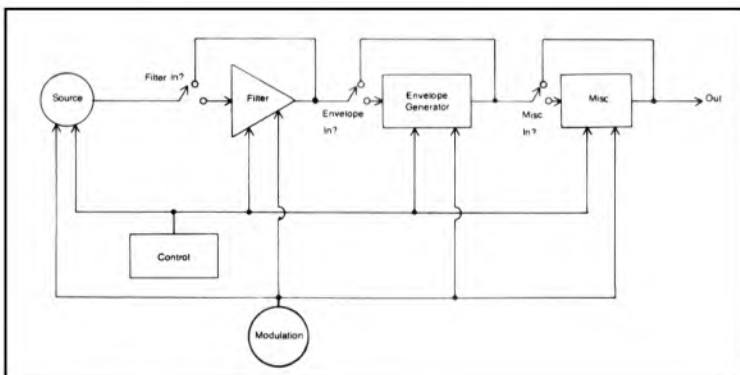


Fig. 10-9. Basic super-module patch.

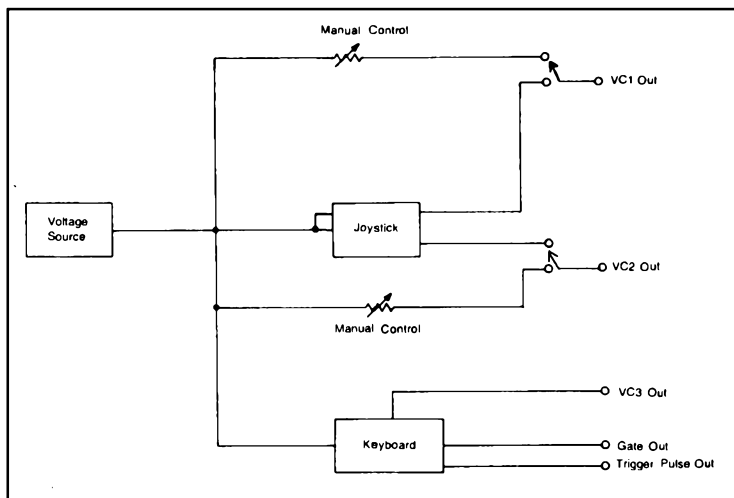


Fig. 10-10. CONTROL super-module.

shown in Fig. 10-16. Alternatively, you could specify each knob position by percentage, or by the clock face method. Both of these methods are illustrated in Fig. 10-17.

The use of super-modules is only to make the system easier to understand and notate. It does not really affect operation in any way. However, the super-module approach can help you arrange your instrument's control panel in a convenient way. Simply keep all of the controls grouped according to their super-modules. Later in this chapter, we will explore control panel arrangement in more detail.

MODULAR SYNTHESIZERS

The early synthesizers were all of the modular type. Many studio synthesizers are still of this type. There are certain advantages.

A modular synthesizer is, by definition, easily expandable. You can always add new modules as the need arises (or as you can afford them).

Modular synthesizers also offer full versatility. Since they can be interconnected in any combination, all patches are possible (providing, of course, that you have the necessary modules). In a hard-wired synthesizer, some combinations are ruled out for the sake of convenience. Admittedly, oddball patches come up rarely, but when they do, it's nice to be able to get the effects you want.

It is generally somewhat easier to troubleshoot and service

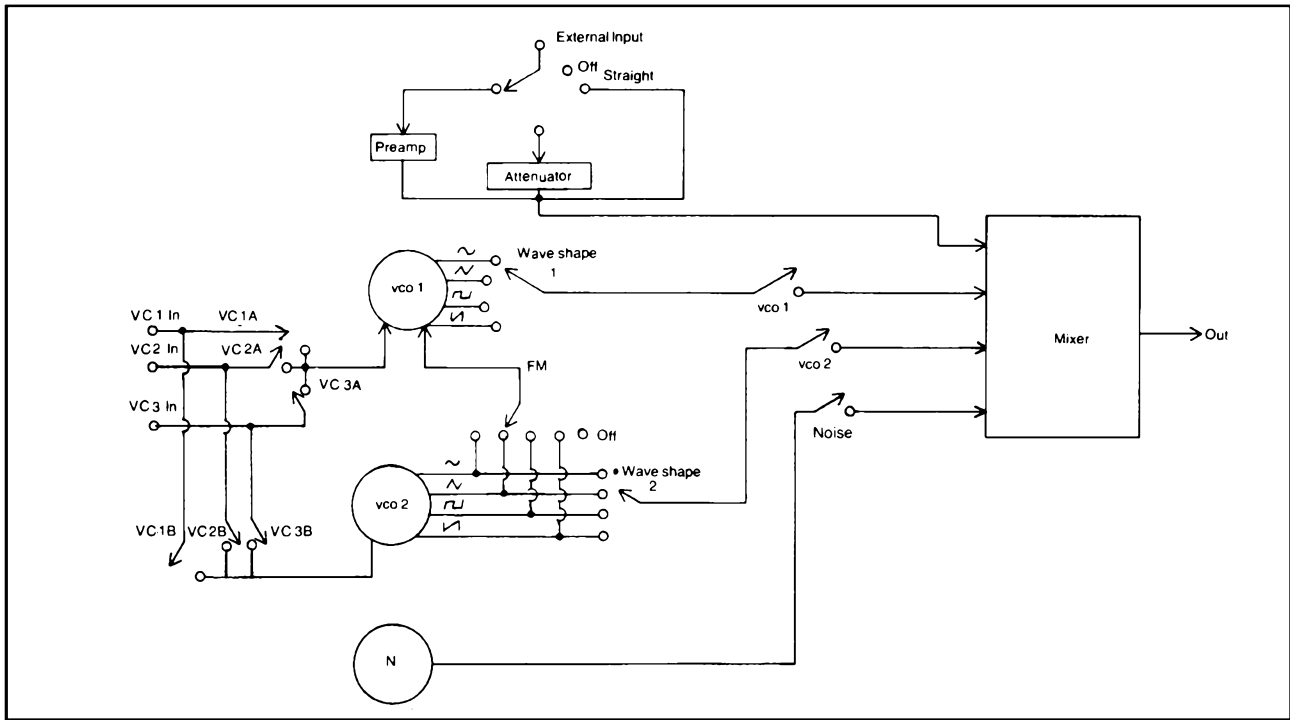


Fig. 10-11. SOURCE super-module.

Table 10-3. Typical Patch Notation for a Hard-Wired Synthesizer.

Super Module	Function	Position		
Control	VC1 out	bias or joystick		
	VC2 out	bias or joystick		
Source	vcc1 in	VC1 on or off		
	vcc1 in	VC2 on or off		
	vcc1 in	VC3 on or off		
	FM	out or waveshape		
	vcc1 out	waveshape select		
	vcc1 out	on or off		
	vcc2 in	VC1 on or off		
	vcc2 in	VC2 on or off		
	vcc2 in	VC3 on or off		
	vcc2 out	waveshape		
	vcc2 out	on or off		
	Noise	on or off		
External In	preamp/attenuate /straight /off			
Filter	Band-pass In	signal 2 or low-pass out		
	High-pass In	signal 1, low-pass out or band-pass out		
Envelope	VC1	in or out		
	VC2	in or out		
	Env 1	in or out		
	Env 2	in or out		
Env 2 trigger	external or Env 1			
Modulation	ifo2 VC	control voltage in /out		
	ifo2 manual	manual control in/out		
	Depth VC	control voltage VCA2 in/out		
	Depth manual	manual control VCA2 in/out		
Misc	S/H Clock	internal /external		
	Synthi-waa In	VC1 on/off — input A		
	Synthi-waa In	VC2 on/off — input A		
	Synthi-waa In	VC3 on/off — input A		
	Synthi-waa In	VC1 on/off — input B		
	Synthi-waa In	VC2 on/off — input B		
	Synthi-waa In	VC3 on/off — input B		
	Rotate In	VC1 on/off		
	Rotate In	VC2 on/off		
	Rotate In	VC3 on/off		
Control	VC1	VC2		
	X	X		
Source	vcc 1 In	VC2	VC3	FM
	VC1	X	X	X
	X	X	X	X
	VCO1 out			
	wave	on /off		
	X	X		
	vcc2 In			
	VC1	VC2	VC3	
	X	X	X	
	Noise			
	X			
	External Input			
X				
Filter	Band-pass In	High-pass In		
	X	X		
Envelope	VC1	VC2	Env 1	Env 2
	X	X	X	X
	Env 2	Trigger		
	X			
Modulation	ifo2		Depth 2	
	VC	Manual	VC	Manual
	X	X	X	X
Miscellaneous	SHClock			
	X			
	Synthi-waa In A			
	VC1	VC2	VC3	
	X	X	X	
	Synthi-waa In B			
	VC1	VC2	VC3	
	X	X	X	
	Rotate In			
	VC1	VC2	VC3	
X	X	X		

Each X should be replaced with the appropriate switch position

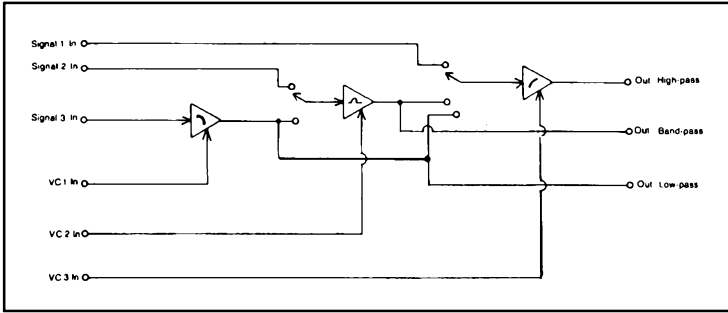


Fig. 10-12. FILTER super-module.

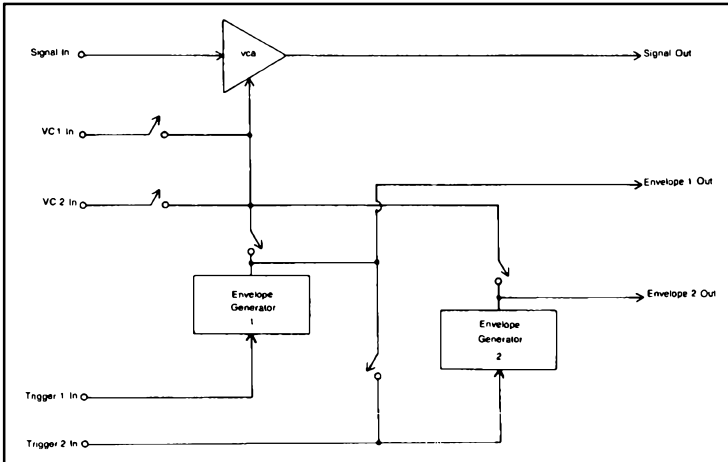


Fig. 10-13. ENVELOPE super-module.

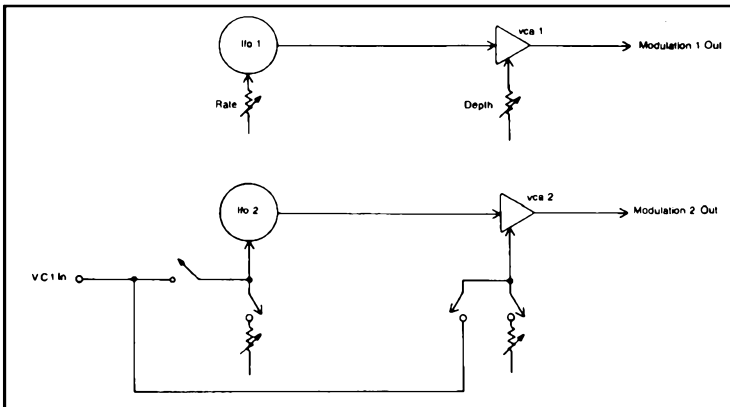


Fig. 10-14. MODULATION super-module.

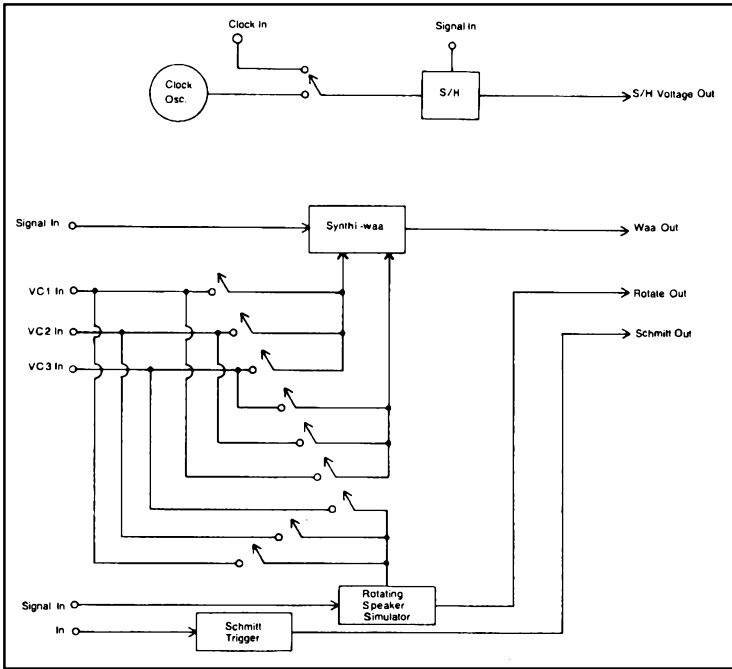


Fig. 10-15. MISC super-module.

modular synthesizers than their hard-wired counterparts. You only have to be concerned with the defective module. If a module is out of commission for awhile for whatever reason, you still have the rest of your synthesizer to work with. In a hard-wired system, it is often an all or nothing proposition.

In designing a hard-wired synthesizer, you have to decide on a permanent compromise for the arrangement of the controls. You

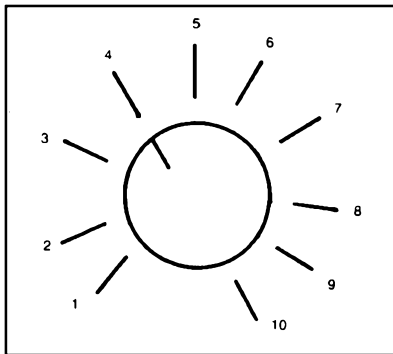


Fig. 10-16. Calibrated knob.

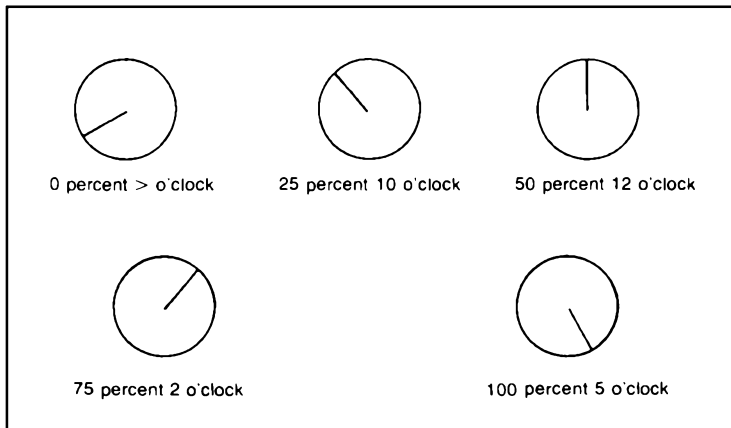


Fig. 10-17. Specifying knob positions by the percentage or o'clock methods.

have to determine the best position for each module for maximum convenience in most cases. Often, it won't be possible to find a satisfactory arrangement that is adequate for every patch and playing situation. Many modular synthesizers allow you to freely reposition modules as the need arises. If you travel, you only have to take the specific modules you need. This saves on bulk and weight, not to mention the chances for damaging a circuit.

Admittedly, many modular systems are awkward in live performance situations. But in a studio, I believe the advantages of the modular approach far outweigh the disadvantages. Usually a little bit of careful thought and planning can overcome most of the problems of playing a modular synthesizer live too.

Patch Cords

It could be fairly said that the most important part of a modular electronic music synthesizer is the patch cords. If you can't interconnect the modules, the damn thing is just going to sit there and collect dust.

Wire. Patch cords must be made of strong, flexible wire. They will inevitably be frequently twisted and bent. Stiff cables will soon develop internal breaks which can cause the electronic musician all sorts of grief—especially when intermittent breaks are involved.

Some synthesists insist on shielded cable. Shielding is certainly a good idea, if you can afford it, but I don't think it is 100 percent essential.

Under some circumstances, a patch cord (especially a fairly long one) can act as an antenna, picking up ac hum and other

unwanted signals. If you should run into this problem, you should consider using shielded cable for your patch cords.

Connectors. You should definitely standardize the connectors on your patch cords and modules. It is often hard enough to come up with the right patch cord in a well standardized system. If some modules require phone plugs, and others take phono plugs, and still others call for mini-plugs, you are just complicating your task.

Many experimenters prefer quarter inch phone jacks and plugs. These relatively large connectors are easy to work with and are widely available. They are generally used on most commercial instruments (electric guitars, organs, etc.). In addition, two or three conductor plugs and jacks are available.

The main disadvantage of quarter inch phone plugs is that they are rather large and take up quite a bit of control panel space. Modular synthesizers tend to be fairly bulky anyway. Mounting a number of large jacks will only make the instrument even more bulky.

So-called mini-plugs are a good choice. These units have a diameter of $\frac{1}{8}$ inch, half the size of the standard $\frac{1}{4}$ inch phone plug. Except for their smaller size, they are identical to their big brothers. Most mini-plugs are two conductor types, but three conductor mini-plugs have begun to appear on the market.

If you use a common ground for all of your modules, you will only need one conductor anyway. Two conductor quarter inch and mini-plugs are readily available, so they make a good choice even if one conductor is left unused.

On the other hand, if your modules do not share a common ground line, you will need two conductor cables to make the interconnections: one for the signal line, and one for the ground.

It is a good idea to use different types of plugs and jacks for signal lines and control voltage lines. This makes it far easier to keep track of what is going where. Using different colored cables can also make things easier to follow. Some synthesizers use mini-plugs for signal lines, and sub-mini, or pin plugs for the control voltage lines.

Remember, that there is really no fundamental difference between a signal voltage and a control voltage. There are times when you will want to use a signal source as a control voltage (for example, using a vco for FM effects). Occasionally you will also want to process a control voltage as if it was a signal voltage. An example of this is shown in the patch diagram of Fig. 10-18. To

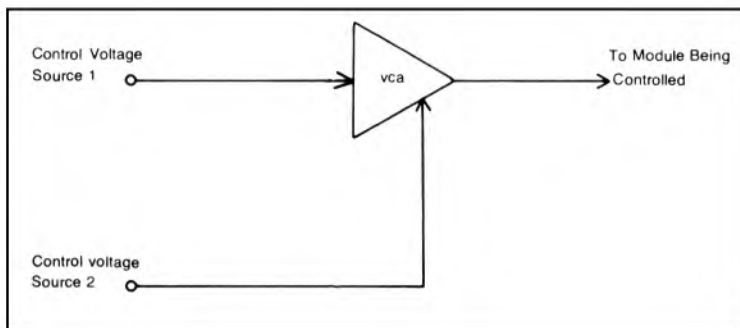


Fig. 10-18. Processing a control voltage.

facilitate these functional cross-overs, you should make up a few patch cords with a signal plug on one end, and a control voltage plug on the other.

In making up your patch cords, make several different lengths. Obviously, you will need a few that can reach between the furthest points of your synthesizer, but it is not a good idea to make all of your patch cords this long. Besides wasting wire, it increases the probability of major cable snarls, and confusion. A longer cable is also somewhat more likely to develop an internal break.

Pin Patching

A few commercial synthesizers employ an alternative to the standard patch cord approach. This system is known as *pin patching*.

A special patch panel with rows and columns of holes accounts for all of the inputs and outputs of the various modules. A simple example is shown in Fig. 10-19. The internal construction of the patch panel is illustrated in Fig. 10-20. By inserting a conductive pin in a hole, the appropriate input and output are shorted together.

A number of pins can be placed in a single row or column. For example, if the patch panel is pinned to look like Fig. 10-21, the synthesizer will be patched as shown in Fig. 10-22.

The pin patching system is certainly convenient. All of the potential connection points are collected in a single area. The control panel can be relatively small and compact. Best of all, it is a snap to tell at a glance what is connected to what.

Like everything else in the world, there are disadvantages to pin patching. It is a little difficult to build a patch panel, since three boards have to have their holes perfectly aligned (panel, inputs, and outputs).

In addition, the pins are usually fairly small and easily lost, so a source for replacements is essential.

The biggest potential problem with a pin patching system is that the patch panel can not be easily expanded to accommodate new modules.

Rack Mounting

A hard-wired synthesizer is usually housed in a single case. In a modular system, however, each module is more a discrete, individual unit. If you frequently change or expand your synthesizer system, housing your instrument can be a bit of a problem.

Inputs \ Outputs	vco 1	vco 2	Low-pass Filter	Low-pass Filter Control	Band-pass Filter	Band-pass Filter Control	vca	vca Control	System Out
Keyboard	00	00		00		00		00	
Joystick A	00	00		00		00		00	
Joystick B	00	00		00		00		00	
Envelope Generator 1	00	00		00		00		00	
Envelope Generator 2	00	00		00		00		00	
vco 1 Triangle		00	00	00	00	00	00	00	00
vco 1 Rectangle		00	00	00	00	00	00	00	00
vco 1 Sawtooth		00	00	00	00	00	00	00	00
vco 2 Triangle	00		00	00	00	00	00	00	00
vco 2 Square	00		00	00	00	00	00	00	00
Low-pass Filter	00	00			00	00	00	00	00
Band-pass Filter	00	00	00	00		00	00	00	00
vca	00	00	00	00	00	00			00
ifo	00	00	00	00	00	00	00	00	

Fig. 10-19. Pin patching panel.

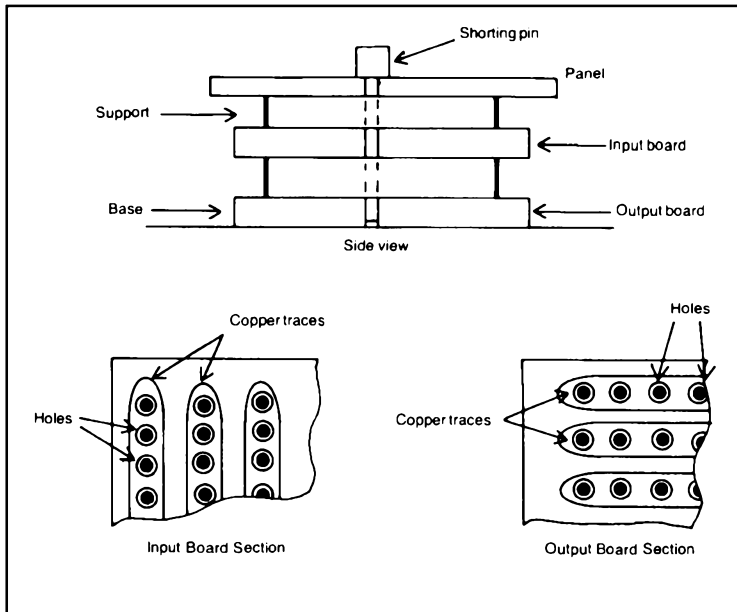


Fig. 10-20. Internal construction of a pin patching module.

One common solution is rack mounting. This is simply a relatively large housing that can hold a number of individual modules, each with their own front panel. This idea is roughly illustrated in Fig. 10-23. Module panels should all be of the same height, but they may be of different widths, so excess space is not wasted on modules with few controls.

Your initial rack should hold more modules than you start out with. This leaves room for expansion. Unused spaces can be left open, or, preferably, covered with blank front panels. Blank front panels will keep dust and what-not out of the circuitry. Of course, when you go to build a new module, you can drill and use one of the blank front panels.

Inside your rack mount housing you will need several bus lines—one for each power supply voltage and one for ground. These buses can be conductive metal rods, or stiff, uninsulated wire.

It is a good idea to include a bypass capacitor across each relevant power line and ground on each individual module. This will help cut down spurious transients that could develop along the lengthy bus lines. These bypass capacitors should be electrolytic capacitors with a value in the 100 μF to 500 μF range. Larger values

may be used, but aren't really necessary in this application. The use of these bypass capacitors is illustrated in Fig. 10-24.

Rack mounted synthesis equipment is not particularly attractive, and probably won't be as conveniently portable as, say, a hard-wired instrument. But in the constantly changing environment of an experimenter's studio, rack mounting is probably the most practical approach to housing modules that anyone has devised yet.

THE GREAT COMPROMISE

Both hard-wired and modular electronic music synthesizers have their advantages and disadvantages. Hard-wired units are

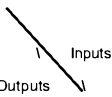
	vco 1	vco 2	Low-pass Filter	Low-pass Filter Control	Band-pass Filter	Band-pass Filter Control	vca	vca Control	System Out
Keyboard	•	•		•		00		00	
Joystick A	00	00		00		00		00	
Joystick B	00	00		00		00		00	
Envelope Generator 1	00	00		•		00		00	
Envelope Generator 2	00	00		00		00		•	
vco 1 Triangle		00	00	00	00	00	00	00	00
vco 1 Rectangle		00	00	00	00	00	00	00	00
vco 1 Sawtooth		00	00	00	•	00	00	00	00
vco 2 Triangle	00		00	00	00	00	00	00	00
vco 2 Square	00		•	00	00	00	00	00	00
Low-pass Filter	00	00			00	00		•	00
Band-pass Filter	00	00	00	00			00	•	00
vca	00	00	00	00	00	00			•
lfo	00	00	00	00	00	•	00	00	

Fig. 10-21. Sample pin patch.

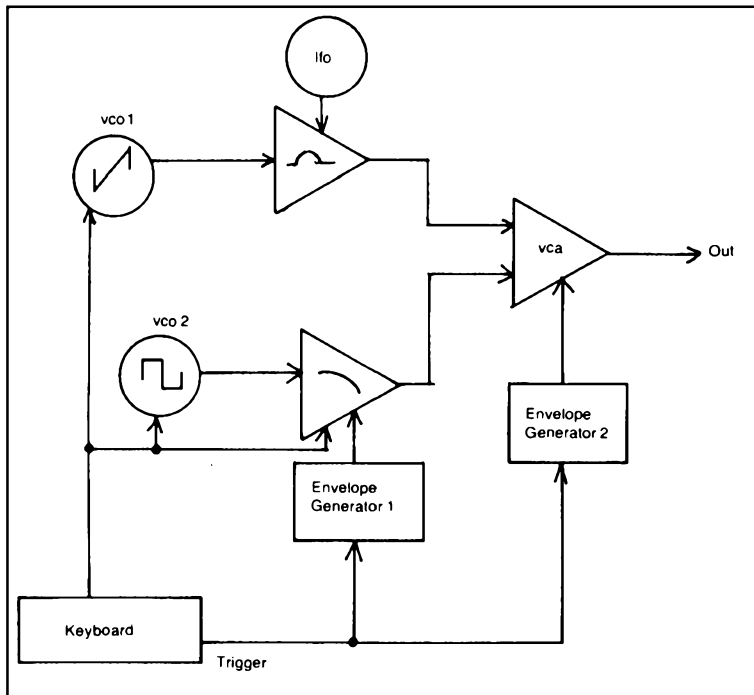


Fig. 10-22. Patch diagram for the sample pin patch of Fig. 10-21.

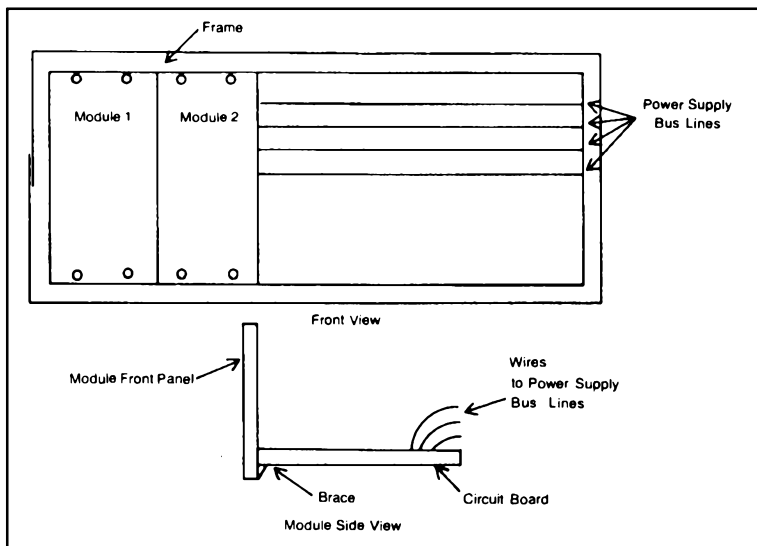


Fig. 10-23. Rack mounting modules.

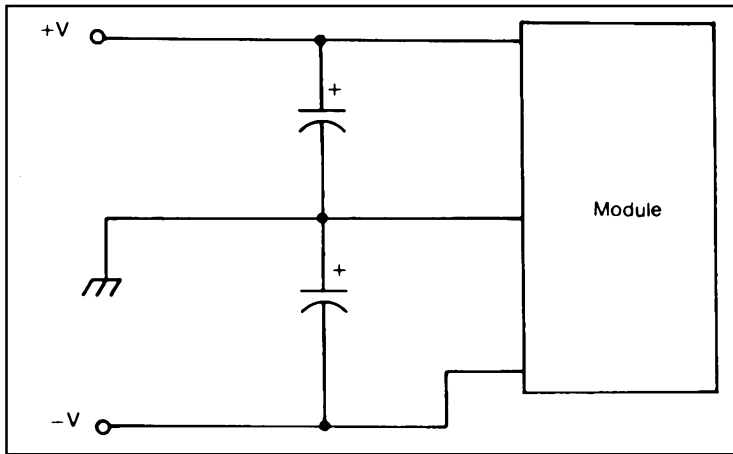


Fig. 10-24. Using bypass capacitors in power supply lines.

somewhat more convenient for live, concert performances and traveling, while the modular approach is more at home in a permanent studio.

Unfortunately, the decision is by no means as simple as the previous paragraph might suggest. Many musicians will want to perform both live and in a studio. Other individual requirements might also influence the choice.

A reasonably good compromise is available. You can merge both the hard-wired and modular concepts. This approach has been taken on a number of commercially available synthesizers. The classic Arp 2600 instrument (no longer being manufactured, alas) is a prime example.

Common connections between modules are hard-wired as outlined earlier in this chapter. However, small jacks are also provided for each input and output, so that the internal connections can be bypassed with a patch cord. Switching jacks (see Fig. 10-25) should be used here. They are more expensive than regular jacks, but some means of breaking the hard-wired connections is needed. Separate switches would be more expensive, and would add considerably to the bulk of the control panel. They would also tend to make the instrument more inconvenient to use.

A *switching jack* has a set of switch contacts which are closed when no plug is inserted in the jack. When a plug is inserted into the jack, these contacts are separated.

This technique also offers a simple way to bypass unwanted hard-wired modules without replacing them with anything. This can

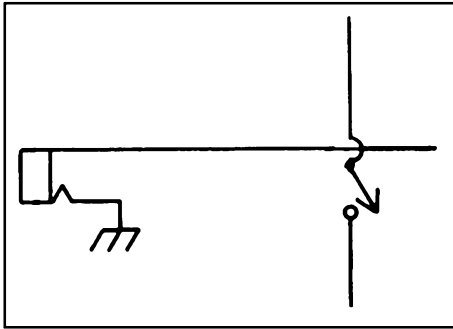


Fig. 10-25. Switched jack.

be done simply by inserting a plug with no patch cord attached. This will break the internal connection without inserting an external signal. Clearly, this allows you to save control panel space by eliminating many of the switches shown in the figures in the section on hard-wired instruments. Figure 10-26 demonstrates how four switches can be eliminated from the SOURCE super-module which was originally shown in Fig. 10-11.

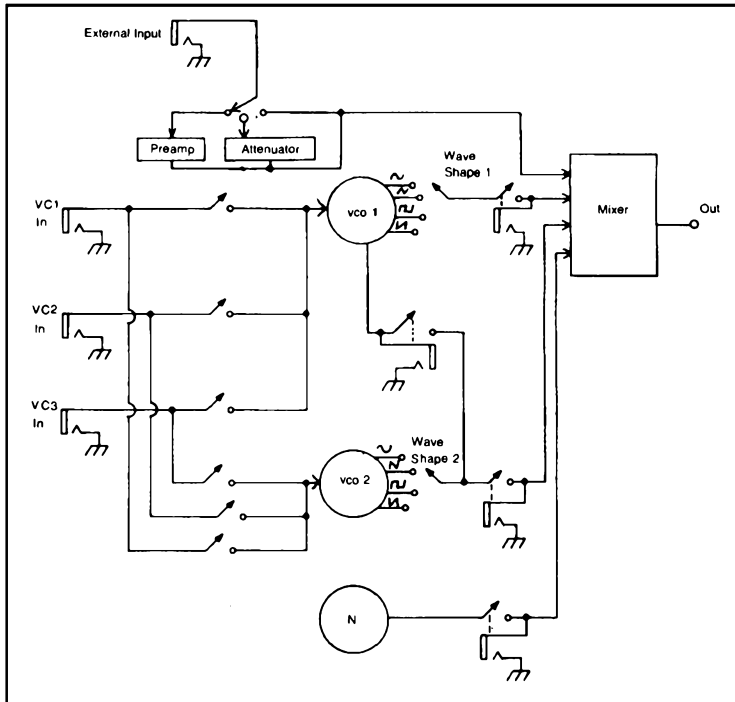


Fig. 10-26. Revised SOURCE super-module (refer to Fig. 10-11).

This compromise system makes it much easier to design the hard-wired portion of the synthesizer, since you don't really have to strive to allow for as many potential patches as possible. You should make provisions for all frequently used connections; only the occasionally used patches should require patch cords.

While the main instrument will not be expandable with this system (once the control panel has been designed, it will be difficult to make changes), you can use externally housed modules to expand your capabilities. Just use patch cords to connect the modules to the appropriate input/output jacks. For maximum convenience and portability, you can house the external expansion modules in small cases designed along the rack mounting concept discussed a few pages ago. Each case might hold six to twelve modules. A block diagram of a typical expanded compromise synthesizer system is shown in Fig. 10-27.

CONTROL PANELS

One of the most important parts of designing a practical electronic music synthesizer is laying out the controls. I have seen far too many instruments (both home-brew, and a few commercial units) with poorly thought out control panels.

The controls must be conveniently accessible, and logically located so the musician can always quickly operate the specific control he needs. Turning the wrong potentiometer can create obvious problems.

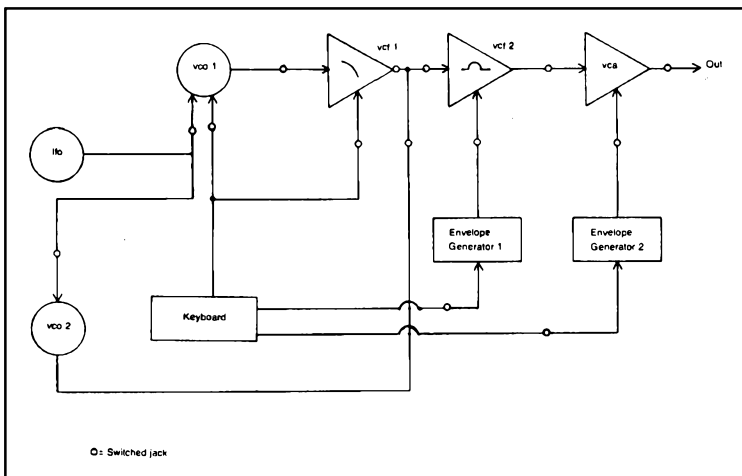


Fig. 10-27. Typical expanded compromise system.

The controls should be grouped together so that the musician can reach all of them from a single position. However, you must avoid cramming the controls too tightly together. A certain amount of fingerspace is essential. The musician has to be able to firmly grasp the needed control quickly, and without disturbing the settings of the adjacent controls. Allow a minimum of inch to inch between adjacent controls. If you have large hands, you should leave even more space between controls.

One way to fit the maximum number of controls in the smallest possible space is to stagger the controls, as shown in Fig. 10-28.

Related controls should be grouped together, either by function, or by module. In many cases you can use both groupings in a row and column arrangement.

Of course, in a modular system each module will have its own controls. The front panels of the modules themselves should be positioned in functional groups. For example, all of the vcos should be grouped together.

The controls on each individual module should be arranged in a fairly standardized pattern to avoid confusion. Duplicate modules should have identical control placement. Figure 10-29 shows one possible system of standardization.

All controls should be clearly and unambiguously marked. Calibration scales on potentiometer knobs and setting markings on switches are a must. All knobs should have some kind of position indicator, such as a pointer (see Fig. 10-30), or a positioning line (see Fig. 10-31).

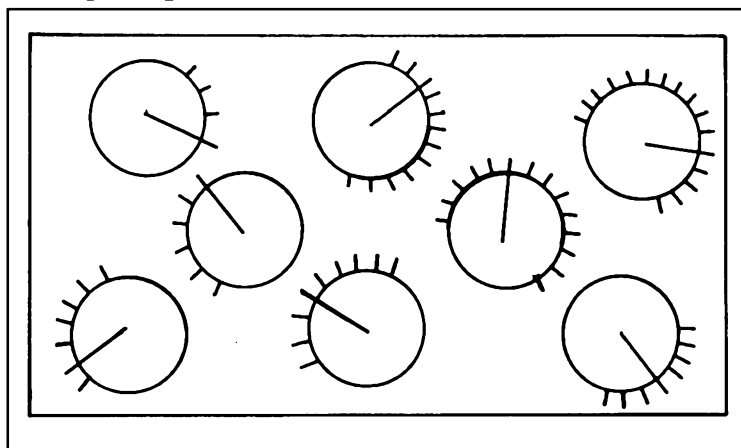


Fig. 10-28. Staggered controls.

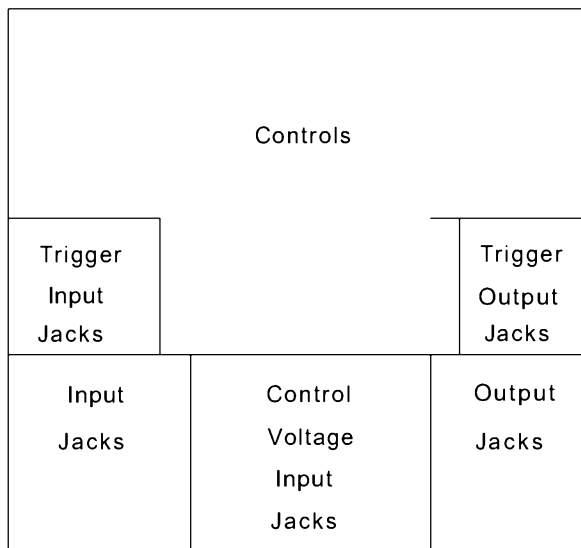


Fig. 10-29. Standardizing module control panels.

The most important component you will need in designing your synthesizer's control panel is a good dose of common sense. Remember, somebody is going to have to play the instrument. Why make it an obstacle course? A little extra effort in the design stage will save a lot of effort (and probable cursing) when you are trying to actually use what you've built.

PATCHING IDEAS

Of course the most deluxe electronic music synthesizer package in the world isn't going to be very much good if you don't know how to use the darn thing. A number of patch diagrams have been

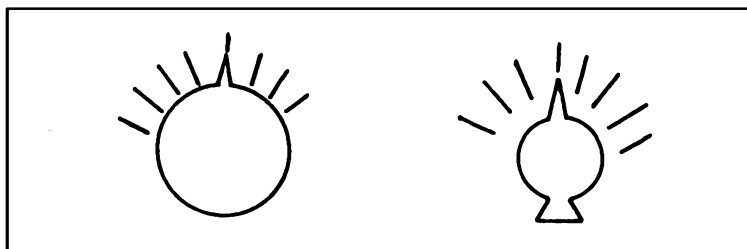


Fig. 10-30. Pointer knobs.

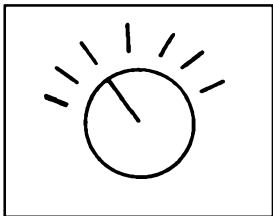


Fig. 10-31. Line positioned knob.

presented throughout this book to illustrate various points. In this section we will look at a number of additional patching ideas.

Remember, these are just suggestions. Feel free to experiment with any or all of the patches shown here. These diagrams are presented in the hopes that they will stimulate your imagination and get you started on creating your own unique sounds.

Standardized symbols are used to represent various types of modules. A number of notation systems have been developed. The one I use is summarized in Fig. 10-32.

Your customized system may or may not have all of the modules called for in these patches. If you come across an interesting patch that you don't have the modules for, try to think of what you have that can be substituted either to simulate the described effect, or (better yet) create an entirely new effect.

And by all means, experiment with different control settings. In many of these diagrams, specific control settings won't even be mentioned. Envelopes especially can create subtle or drastic differences in the sound of a given patch.

Definitely keep a notebook of your experimentation. You don't need anything fancy. An inexpensive spiral bound notebook from the drugstore is fine. Just jot down any patch you come up with that sounds interesting, even if you can't imagine how you might use it. You might just need exactly that sound at a later date, and nothing is quite as frustrating as not being able to recreate that perfect sound you chanced upon while fooling around last week. Be sure to keep notes on any relevant control settings too.

Even a relatively simple synthesis system will allow literally - hundreds, if not thousands, of sounds to be patched. Don't trust your memory.

In presenting these patching ideas, I have avoided saying much about the actual sounds created. For one thing, the control settings can often completely alter the character of a synthesized sound. In addition to this, the specific circuits you use may change the sound quality to a greater or lesser extent. For example, there are differences in filter cutoff slopes and the purity of waveforms generated

by oscillators. The exact same patch may sound noticeably different on two different synthesizers.

For these reasons, I have not emphasized realistic imitative sounds (trumpet, string ensemble, race car, etc.) in these sample patch diagrams. Instead, I feel more abstract patches are generally more useful. In some cases, one of the patches presented here could be used as the starting point in synthesizing the sound of a traditional instrument (or whatever), but don't count on a perfect imitation on the first try. Imitative synthesis is demanding and great

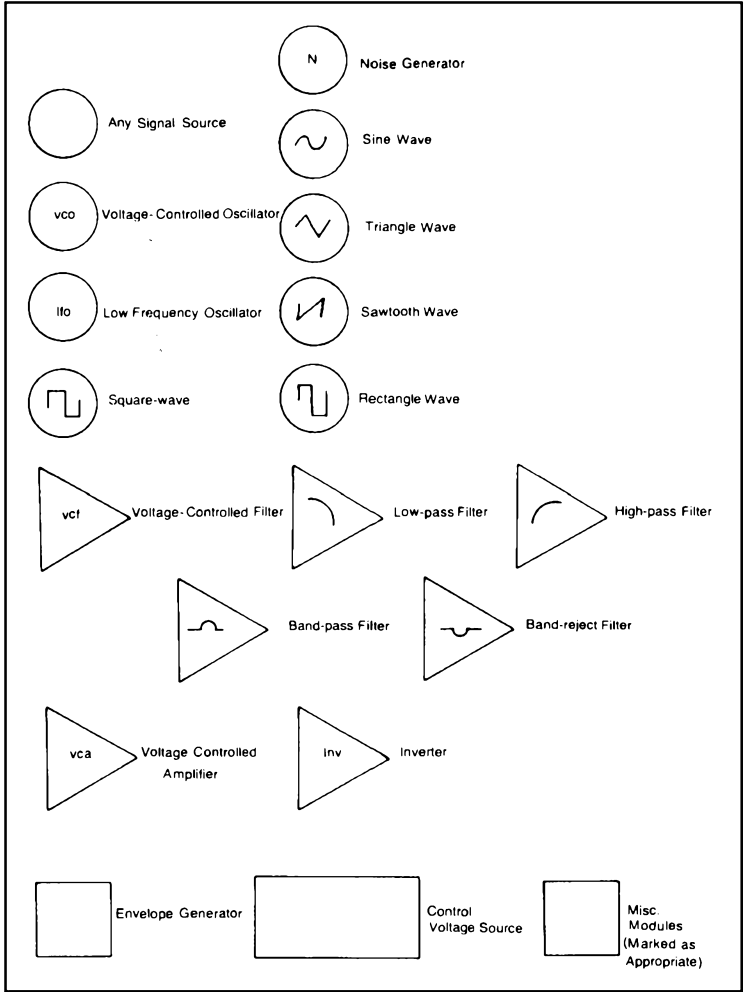


Fig. 10-32. Patching diagram symbols.

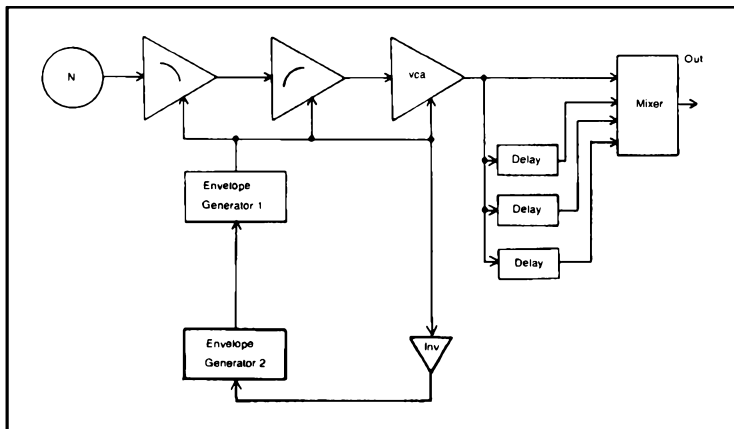


Fig. 10-33. Patch 1.

precision in settings is required. Duplicating the sound of most traditional instruments requires great patience, and (usually) a great many modules. It is not uncommon to find 5 or 6 vcas being used in a single imitative patch (not to mention the other types of modules).

Patch 1. The first sample patch is shown in Fig. 10-33. The modules required are outlined in Table 10-4.

Each of the delay units should be set to a different time to give the sound of a constantly changing phase effect.

The most novel feature of this patch is in how the two envelope generators are used to trigger each other. A specific voltage is required to trigger an envelope generator, so envelope generator 2 will trigger envelope generator 1 when the output voltage reaches the minimum trigger level. Different attack times will vary the time when envelope generator is triggered.

The output of envelope generator 1 is used to control the two vcfs and the vca. This envelope is also inverted and used to trigger

Table 10-4. Modules for Fig. 10-33.

Noise Generator	1
vcf, low-pass	1
vcf, high-pass	1
vca	1
Delay	3 (or more)
Envelope Generator	2
Inverter	1
Mixer	1

envelope generator 2. Because this signal is inverted, or reversed, the minimum triggering level will be reached during the decay, or release portion of the envelope. The two envelopes will continuously cycle, endlessly triggering each other, producing bursts of noise that swell and fade. You might be able to adjust this patch to produce a surf, or rainstorm effect.

Of course, you can try different types of vcfs, or signal sources. If a vco is used, it should generate a complex waveform, or the sound will be rather bland.

One of the filters could be controlled by envelope generator 2 rather than envelope generator 1. Alternatively, try controlling a filter with the inverted envelope that is triggering envelope generator 2.

Instead of feeding the four output signals (the straight signal, and the three delayed signals) into a monophonic (single output) mixer, you could run two of the signals to each channel of a stereo setup. This effect can be particularly impressive through headphones.

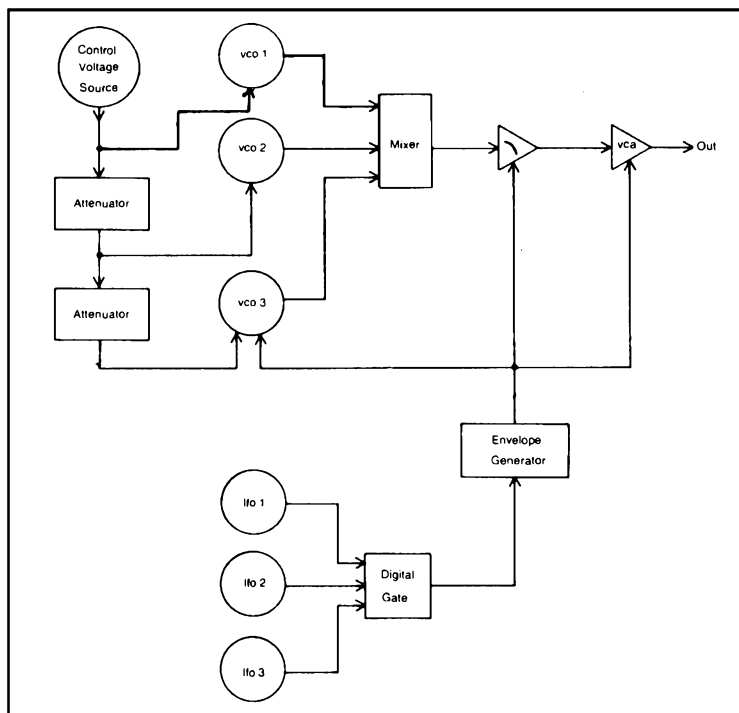


Fig. 10-34. Patch 2.

For an even more mindboggling effect, try adding a rotating speaker effect simulator (described in Chapter 9).

You can change the number of delayed lines too.

Patch 2. Figure 10-34 shows sample patch number 2. The required modules are listed in Table 10-5.

Three lfos are combined via a digital gate (any gate may be used—refer back to Chapter 9). The gated output is used to trigger the envelope generator which controls the vcf, the vca, and shares in controlling vco3.

A single control voltage source is used to drive the three vcOs. Attenuators set vco2 and vco3 to lower frequencies than vco1. The oscillators should be tuned to a triad chord when the envelope is not driving vco3.

Now, when the envelope generator is triggered, a chord will be heard at the output. The envelope driving vco3 will force the chord out of tune, and then it will resolve back into the original tuning. Slow envelopes work best here.

You might want to use a pseudorandom voltage source, or a sample and hold circuit (refer to Chapter 8) to control the vcOs for randomly changing chords. To prevent the chord from changing in mid-envelope, you could use the output of the digital gate as the clock input for the sample and hold module.

Other control voltage sources (keyboard, joystick, sequencer, etc.) could also be used.

Patch 3. The next sample patch is illustrated in Fig. 10-35. The module requirements are outlined in Table 10-6.

This patch is fairly simple. Two vcOs and a noise generator are fed into a band-pass vcf. Both vcOs and the filter are controlled by a single lfo. You could try using a vco in the subaudio range for the lfo and control it with a sequencer, pseudorandom voltage source, or sample and hold for more varied effects.

Table 10-5. Modules for Fig. 10-34.

Control Voltage Source	(see text)
vco (any waveform)	3
lfo	3
Schmitt Trigger	(3 if lfos are not rectangle wave generators)
vcf, low-pass	1
vca	1
Digital Gate	1
Attenuator	2
Mixer	1

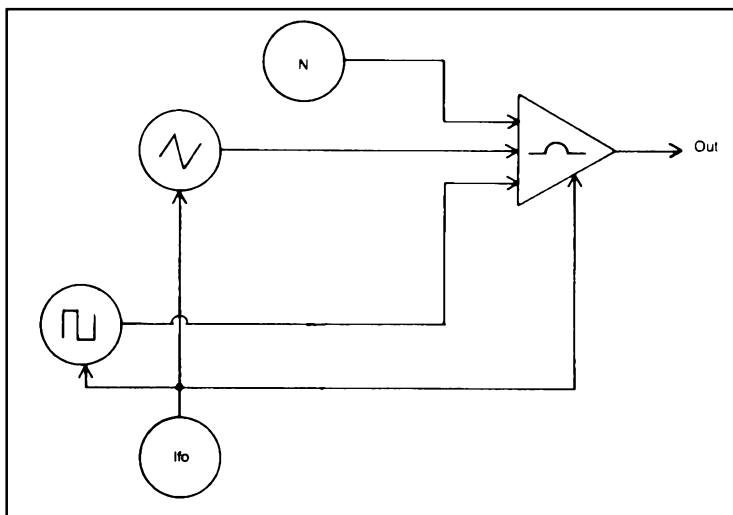


Fig. 10-35. Patch 3.

If the outputs of the vcOs are attenuated so the noise is predominant, and everything is tuned for a fairly high frequency, a fair approximation of a howling wind can be achieved with this patch.

Patch 4. Figure 10-36 shows the fourth sample patch. The required modules are listed in Table 10-7.

This patch creates some complex filtering effects. The harmonic content of the output signal seems to shimmer.

Try adjusting the mixer to give the various filters predominance. Of course, you can also substitute different filter types for those shown. Also, feel free to experiment with different control voltages for the vcfs.

An interesting variation on this patch is shown in Fig. 10-37. Here the output of each filter is fed through an independently controlled vca to cause the harmonic content of the signal to fade and surge in ever changing patterns. Ideally, the lfOs should not generate harmonically related frequencies. This gives the harmonic content a random quality that can be quite fascinating.

Table 10-6. Modules for Fig. 10-35.

vco, triangle wave	1
vco, square wave	1
Noise Generator	1
lfo	1
vcf, band-pass	

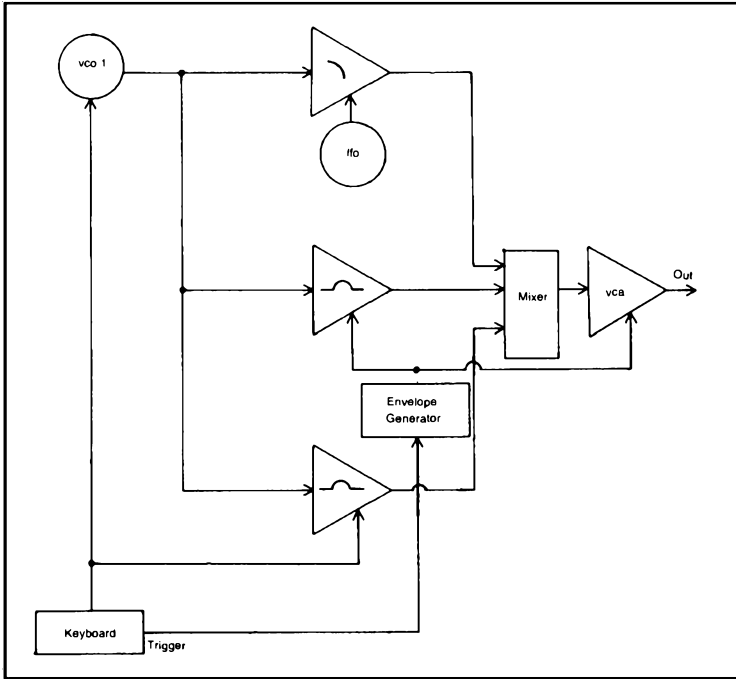


Fig. 10-36. Patch 4.

The outputs of these vcas are mixed together and fed through a master vca, which is controlled by an envelope generator.

The modules required to create this patch are outlined in Table 10-8.

Patch 5. Figure 10-38 and Table 10-9 show the next patch. In some ways it is similar to the two just described. By controlling two vcas with a lfo and inverter, the filtered frequencies will fade in and out of the output signal.

Table 10-7. Modules for Fig. 10-36.

vco, any waveform	1
lfo	1
vcf, low-pass	1
vcf, band-pass	1
vca	1
Envelope Generator	1
Keyboard	1
Mixer	1

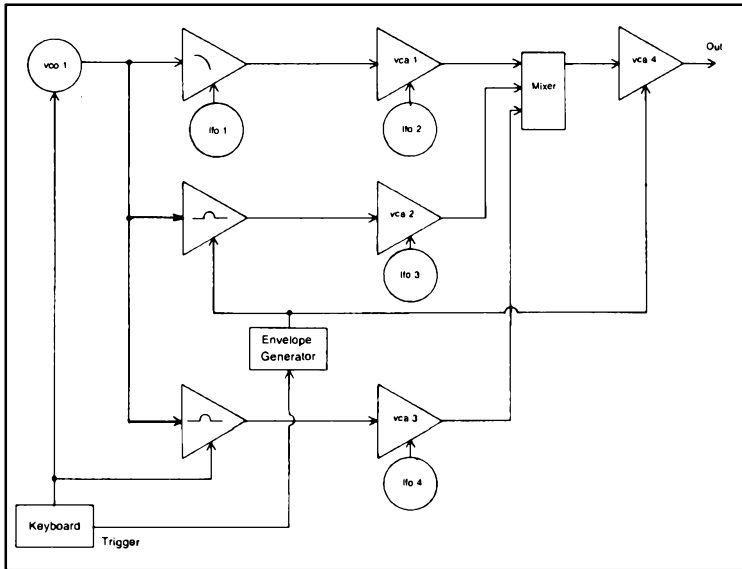


Fig. 10-37. Patch 4 variation.

Vca3 is a master vca used to put an envelope over the entire, fluctuating signal.

Patch 6. The next sample patch is shown in Fig. 10-39. The required modules for this patch are listed in Table 10-10.

The outputs of the two vcOs are ring modulated before being fed through a vcf and vca. An attenuator is used so vco2 will be a constant interval below vco1.

The straight signal and three delay signals are mixed together to give a pseudo-phasor effect. More delays may be used to increase the effect.

At certain settings, this patch will result in a very metallic, robot-like voice.

Table 10-8. Modules for Fig. 10-37.

vco, any waveform	1
lfo	4
vcf, low-pass	3 (see text)
vcf, band-pass	
vca	4
Envelope Generator	1
Keyboard	1
Mixer	1

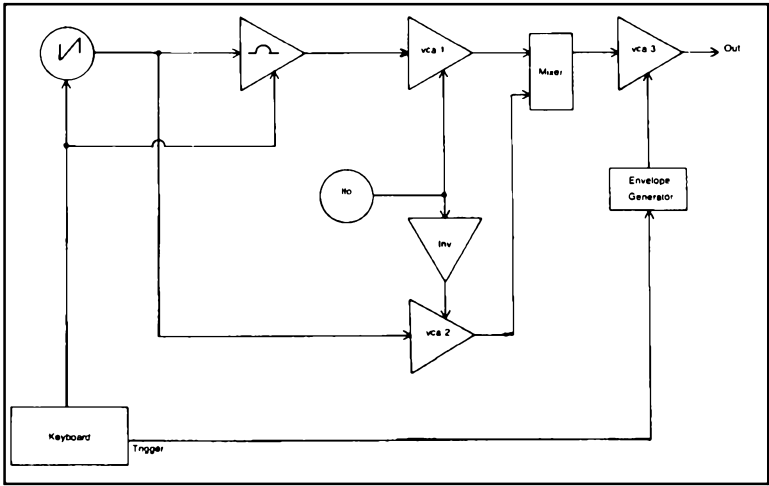


Fig. 10-38. Patch 5.

vco, ascending sawtooth wave	1
lfo	1
vcf, band-pass	1
vca	3
Inverter	1
Envelope Generator	1
Keyboard	1
Mixer	1

Table 10-9.
Modules for Fig. 10-38.

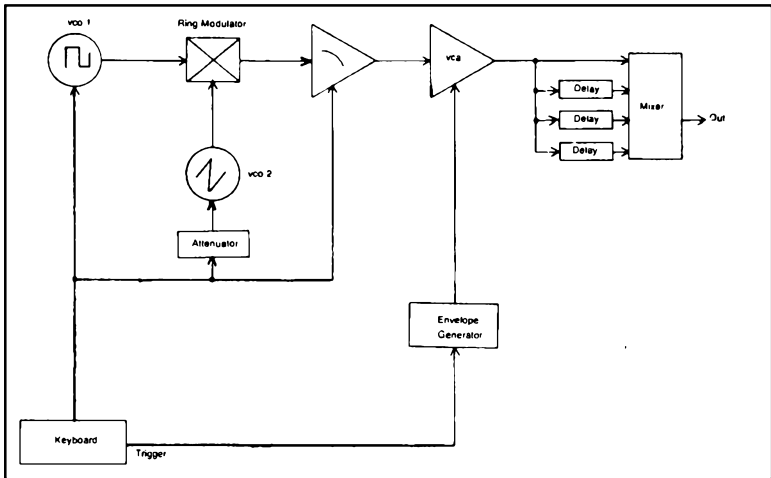


Fig. 10-39. Patch 6.

Table 10-10. Modules for Fig. 10-39.

vco, square wave	1
vco, triangle wave	1
vcf, low-pass	1
vca	1
Envelope Generator	1
Attenuator	1
Ring Modulator	1
Delay	3 (or more)
Keyboard	1
Mixer	1

Patch 7. The patch shown in Fig. 10-40 is entirely self-controlled, and it makes an excellent varying drone effect. The required modules are outlined in Table 10-11.

Vco2 should be set for a very low, sub-audio frequency. This oscillator is controlled by both lfos. Vco2 controls the vcf directly, and is fed through vca2 (controlled by lfo1) to provide a varying amount of control to vco1. Lfo2 also controls the master vca (vca1).

At certain settings, this patch will almost make your synthesizer seem to talk. A very tight Q is required for the band-pass vcf to create this effect.

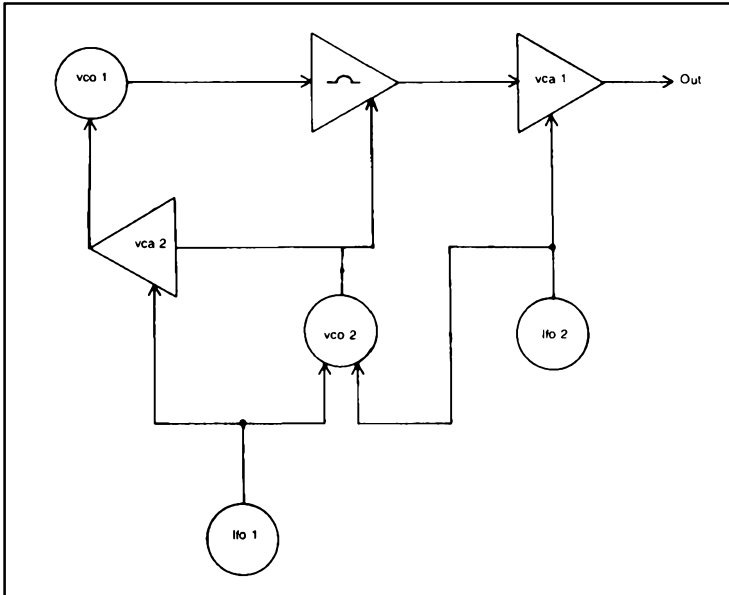


Fig. 10-40. Patch 7.

vco, any waveform	2
lfo	2
vcf, band-pass	1
vca	2

Table 10-11. Modules for Fig. 10-40.

Patch 8. The next sample patch is illustrated in Fig. 10-41. Table 10-12 lists the modules you will need to set-up this patch.

If the attenuator is set so that vco1 is a tenth above vco2 and vco3 and the envelope generator is set for a very short attack, no

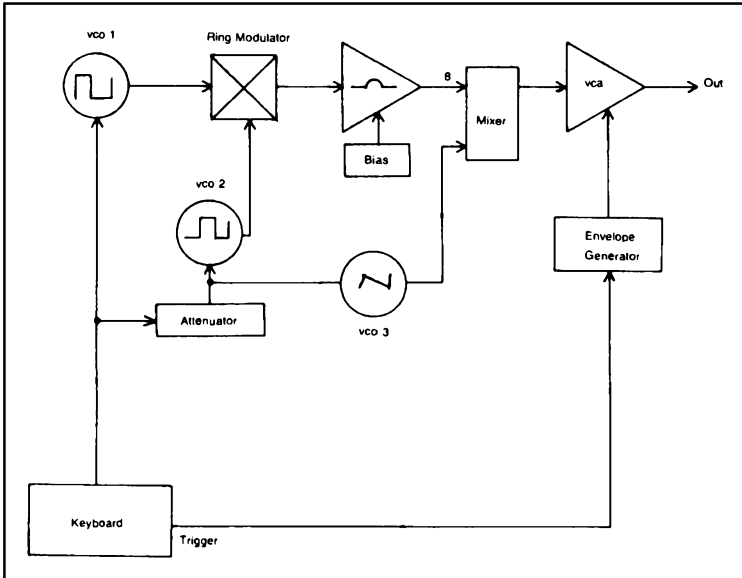


Fig. 10-41. Patch 8.

sustain, and moderately long decay, this patch will generate a pseudopiano effect. It will probably sound more like an electric piano than an acoustic instrument.

vco, square wave	2
vco, triangle wave	1
vcf, band-pass	1
vca	1
Envelope Generator	1
Keyboard	1
Attenuator	1
Bias (fixed voltage source)	1
Mixer	1

Table 10-12. Modules for Fig. 10-41.

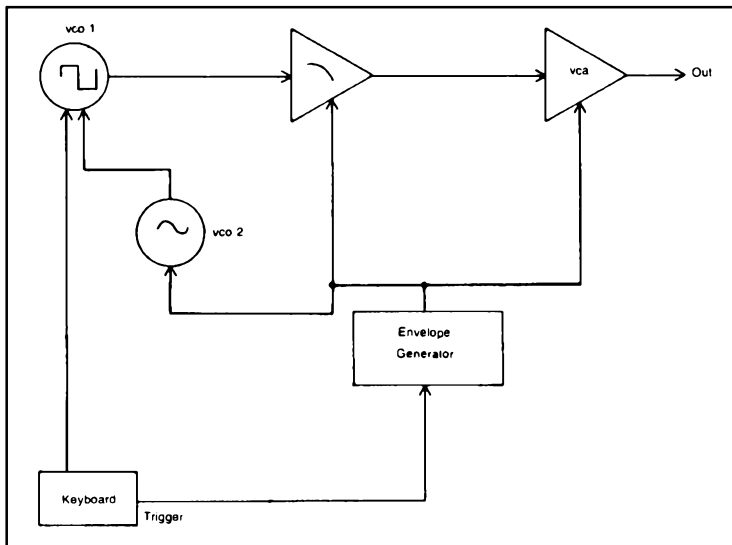


Fig. 10-42. Patch 9.

By very slightly changing the setting of the attenuator, an out of tune honky-tonk piano effect can be achieved.

Of course you should also feel free to experiment with other envelopes and settings of the attenuator. Also try substituting other waveshapes, and change the mixer input settings.

Patch 9. The ninth sample patch is shown in Fig. 10-42. The modules required are outlined in Table 10-13.

If vco2 is set for a sub-audio frequency, the rate of vibrato will increase as the envelope builds up (attack), and decrease as it dies away (decay-release). This gives the sound a very novel and expressive quality.

If vco2 is set in the audible range of frequencies, the output of vco1 will be frequency modulated. The sidebands will change as the envelope goes through its stages.

Patch 10. This patch, which is shown in Fig. 10-43, takes

vco, square wave	1
vco, sine wave (or triangle wave)	1
vcf, low-pass	1
vca	1
Envelope Generator	1
Keyboard	1

Table 10-13. Modules for Fig. 10-42.

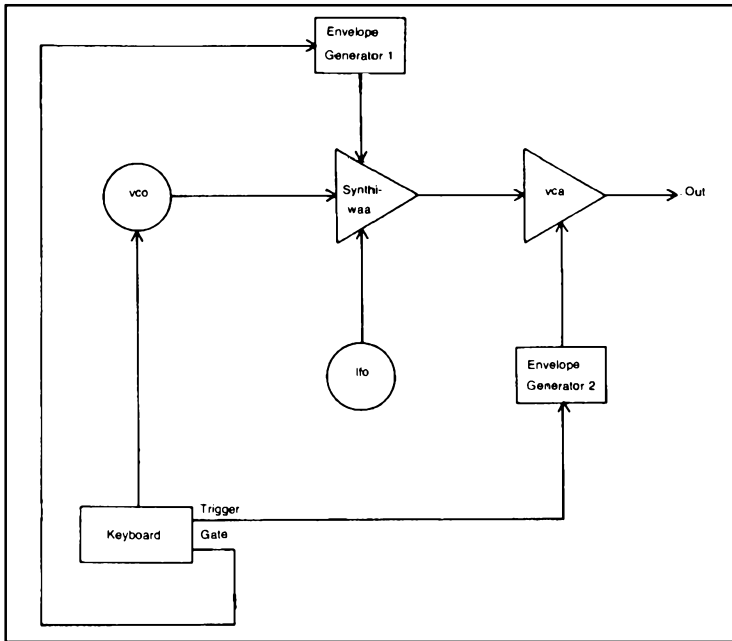


Fig. 10-43. Patch 10.

advantage of the two voltage control inputs to the synthi-waa module, discussed in Chapter 9. An lfo and envelope generator 1 are used to create an odd, fluttering effect.

The gate and trigger pulse signals may be reversed for a somewhat different effect.

The modules required for this patch are listed in Table 10-14.

Patch 11. This patch (Fig. 10-44, Table 10-15) can create some fascinating ring modulation effects. The attenuator is used to detune the two vcOs from unison. If the waveforms shown in the diagram are used, and the two oscillators are tuned an augmented fourth (C to F#) apart, a nice chime-like effect can be achieved. The envelope generator should be set for minimum attack time, no sustain, and a moderately long decay time.

VCO	1
lfo	1
Synthi-waa	1
vca	1
Envelope Generator	2
Keyboard	1

Table 10-14. Modules for Fig. 10-43.

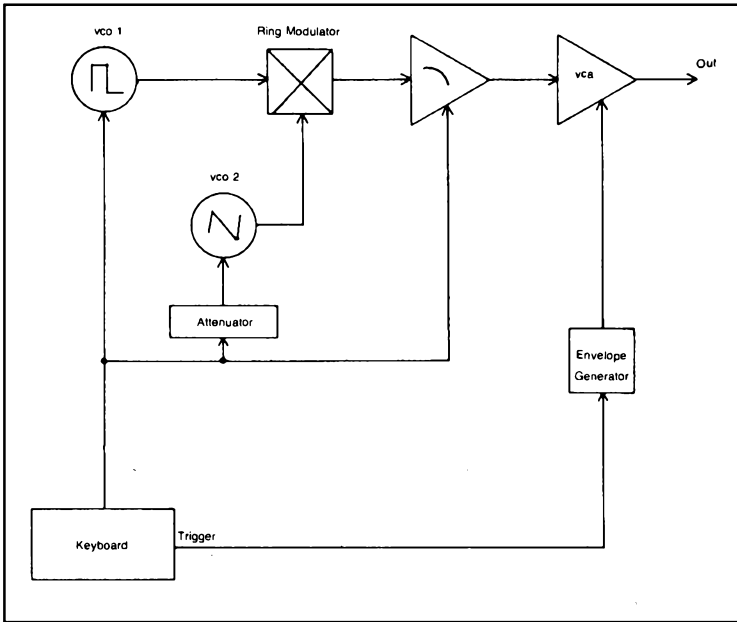


Fig. 10-44. Patch 11.

Of course, you could try other waveforms and envelopes for other effects. The tuning interval between the two oscillators may also be changed.

Mixing in this signal with a few delayed replicas of itself or adding a reverberation unit can strengthen the effect greatly.

Patch 12. The next sample patch is illustrated in Fig. 10-45. The required modules are outlined in Table 10-16.

This patch can produce a very hollow, electronic-sounding quality that is somewhat reminiscent of some of the early uses of synthesizers in pop recordings.

The oscillators should be just slightly detuned from unison. By changing the bias on the vcf, the filter can be manually swept.

vco, rectangle wave	1
vco, triangle wave	1
vcf, low-pass	1
Ring Modulator	1
vca	1
Attenuator	1
Envelope Generator	1
Keyboard	1

Table 10-15. Modules for Fig. 10-44.

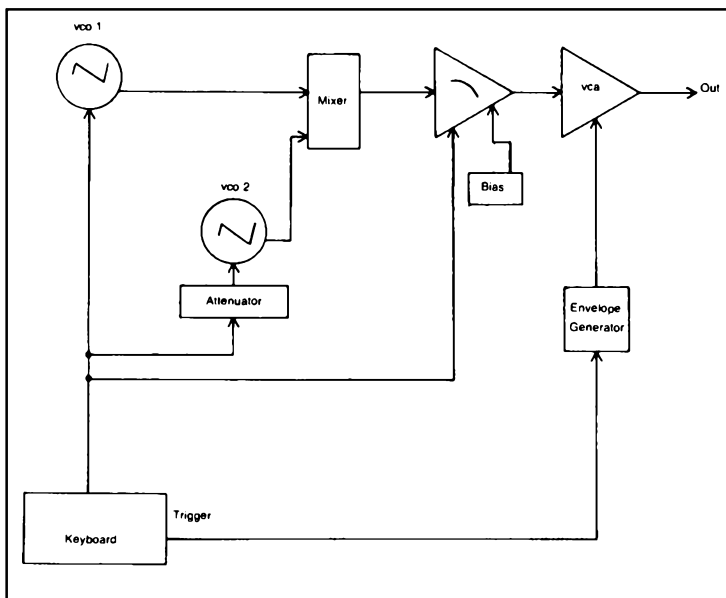


Fig. 10-45. Patch 12.

A variation on this patch is shown in Fig. 10-46. The only difference here is that a two axis joystick is used to manually detune the oscillators (actually, vco2), and sweep the filter. Some very dynamic effects can be achieved by playing the keyboard with one hand, and gently moving the joystick shaft with the other. The modules required to set-up this variation of Patch 12 are listed in Table 10-17.

Patch 13. Patch 13 is shown in Fig. 10-47. The required modules are listed in Table 10-18.

Notice how the output of the vco is split into two signal lines (a signal splitter may be needed—see Chapter 4). One of these signal

Table 10-16. Modules for Fig. 10-45.

vco, triangle wave	1
vco, ascending sawtooth wave	1
vcf, low-pass	1
vca	1
Envelope Generator	1
Keyboard	1
Bias (fixed voltage source)	1
Attenuator	1
Mixer	1

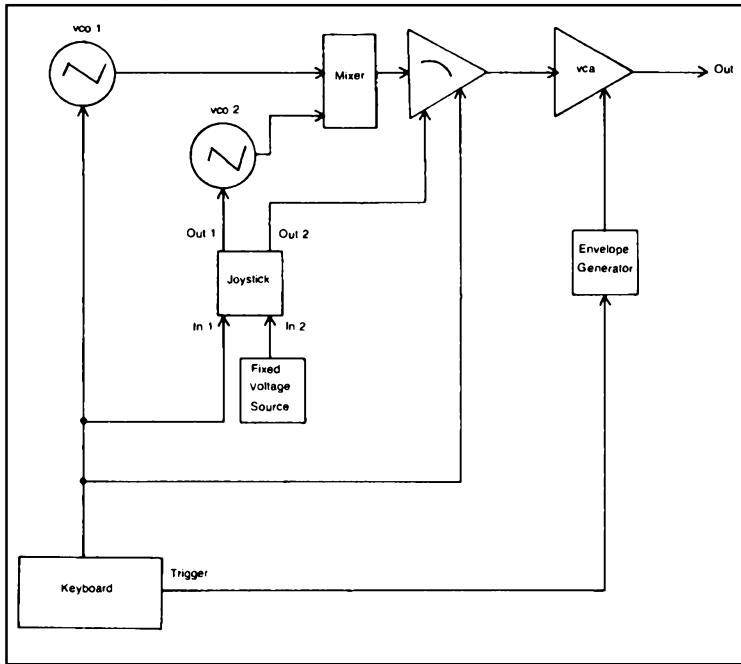


Fig. 10-46. Patch 12 variation.

lines is simply passed through a low-pass filter (vcfl). The other half of the signal is delayed slightly, passed through a rotating speaker effect simulator (see Chapter 9) and then band-pass filtered. The two signal lines are then mixed back together and an envelope is put over the signal via the vca.

A fairly complex waveform, such as the sawtooth wave indicated in the diagram, will tend to work best in this patch.

Patch 14. The next patch is intended for use with an external signal source, such as an electric guitar pickup, or a microphone.

Table 10-17. Modules for Fig. 10-46.

vco, triangle wave	1
vco, ascending sawtooth wave	1
vcf, low-pass	1
vca	1
Envelope Generator	1
Keyboard	1
Bias (fixed voltage generator)	1
Joystick	1
Mixer	1

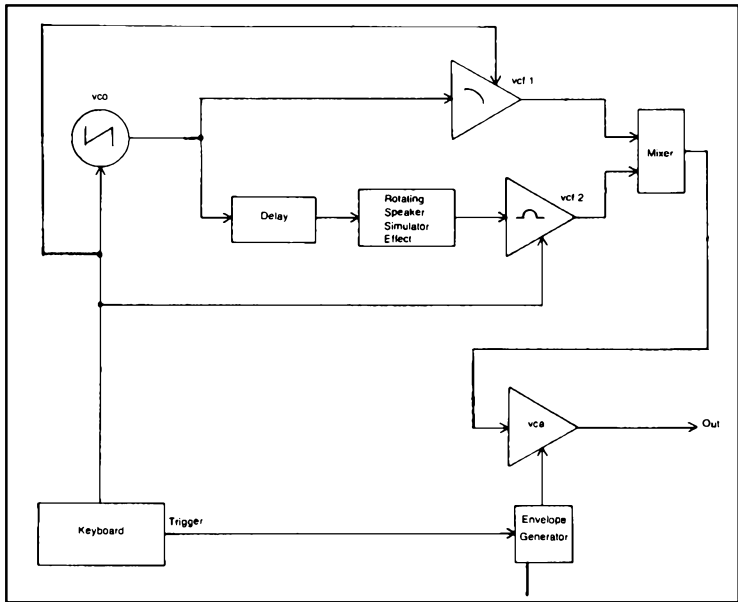


Fig. 10-47. Patch 13.

The patch is illustrated in Fig. 10-48, and the required modules are summarized in Table 10-19.

An envelope follower driven by the external signal source controls a vco. The vco signal is ring modulated with the original signal. The volume of the original signal source will control the ring modulation effect. This can be a very expressive effect. The vca is used to put the signal's original envelope over the entire output.

A simple variation on this patch is shown in Fig. 10-49. The modules required for the variation of the patch are listed in Table 10-20. Here the envelope follower triggers an envelope generator to put a new amplitude envelope over the output signal, rather than

Table 10-18. Modules for Fig. 10-47.

vco, ascending sawtooth wave	1
vcf, low-pass	1
vcf, band-pass	1
Rotating Speaker Effect Simulator	1
Delay	1
vca	1
Envelope Generator	1
Keyboard	1
Mixer	1

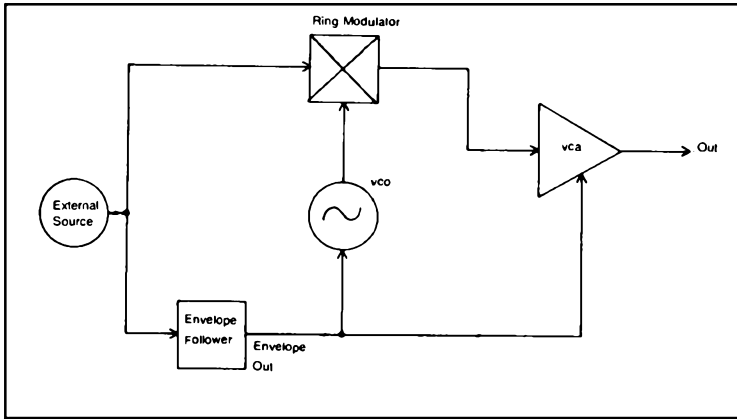


Fig. 10-48. Patch 14.

Table 10-19. Modules for Fig. 10-48.

VCO	1
Any external signal source (preamp or attenuator, if required)	1
Ring Modulator	1
vca	1
Envelope Follower	1

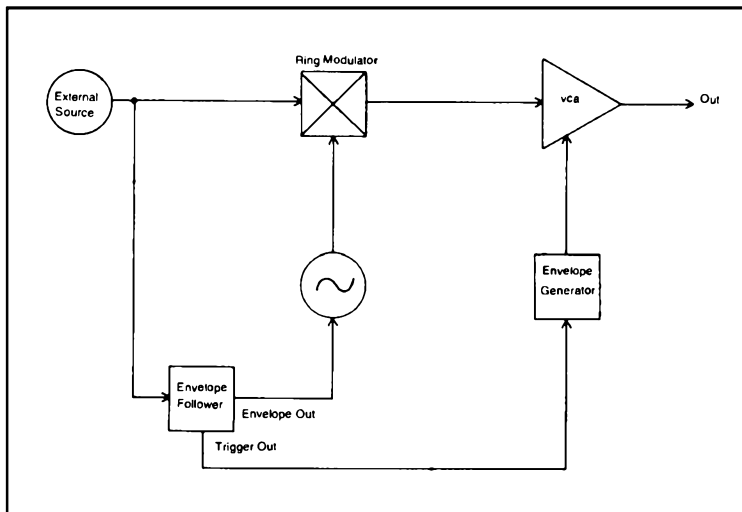


Fig. 10-49. Patch 14 variation.

Table 10-20. Modules for Fig. 10-49.

vco	1
Any external signal source (preamp or attenuator, if required)	1
Ring Modulator	1
vca	1
Envelope Generator	1
Envelope Follower	1

the external signal source's original envelope. This patch can render the original signal source totally unrecognizable, or the changes may be fairly subtle—experiment.

Patch 15. Sample Patch 15 is shown in Fig. 10-50. The modules that are needed to create this patch are listed in Table 10-21.

This is a self-playing patch, controlled by two lfoscillos. The vco is controlled by one of the lfoscillos and an envelope generator that is triggered by the second lfo. The attenuator is used to set the trigger point in the lfo's cycle. A Schmitt trigger module may be necessary to produce a clean trigger signal to the envelope generator.

The output of this patch will slip and slide smoothly from frequency to frequency. Running two or three patches like this

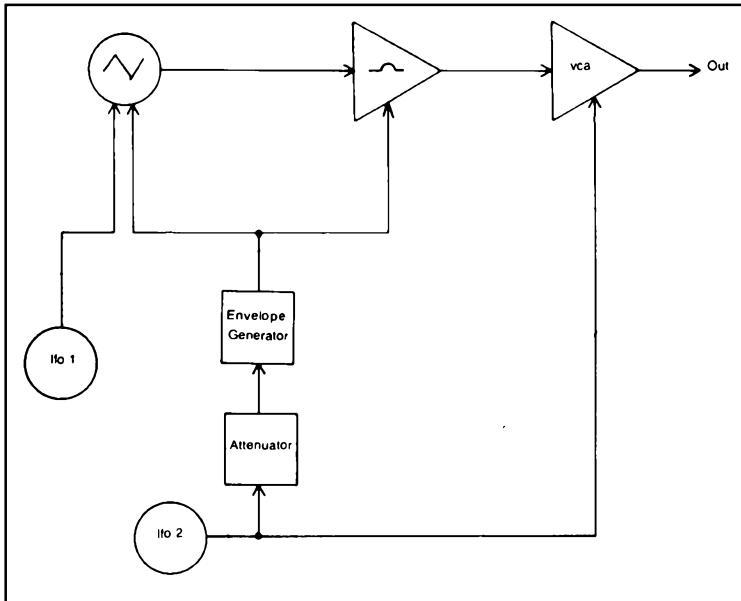


Fig. 10-50. Patch 15.

vco, triangle wave	1
lfo	2
vcf, band-pass	1
vca	1
Envelope Generator	1
Attenuator	1

Table 10-21. Modules for Fig. 10-50.

simultaneously, can produce some very interesting environment music.

Patch 16. The next patch (Fig. 10-51, and Table 10-22) is an unusual variation on the basic synthesizer patch which was pre-

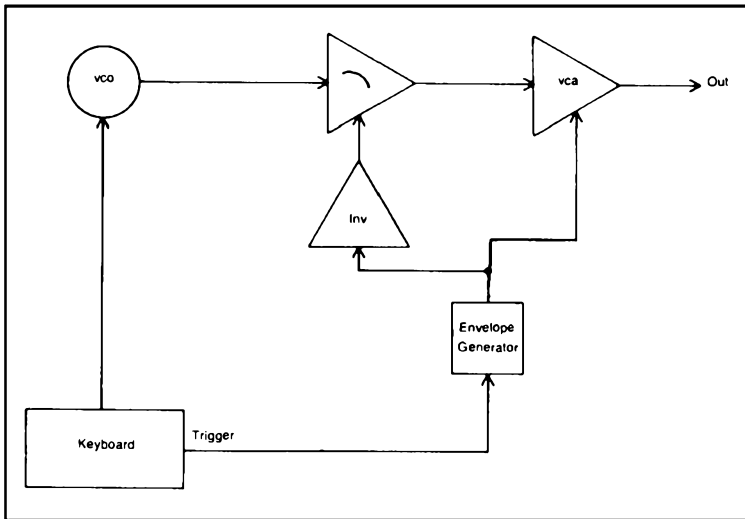


Fig. 10-51. Patch 16.

sented earlier in this chapter. The vcf and the vca are both controlled by a single envelope generator, however, the envelope signal is inverted before it is fed to the vcf, so the filter and the amplifier will run in contrary directions; as one goes up, the other goes down and vice versa.

vco	1
vcf, low-pass	1
vca	1
Inverter	1
Envelope Generator	1
Keyboard	1

Table 10-22. Modules for Fig. 10-51.

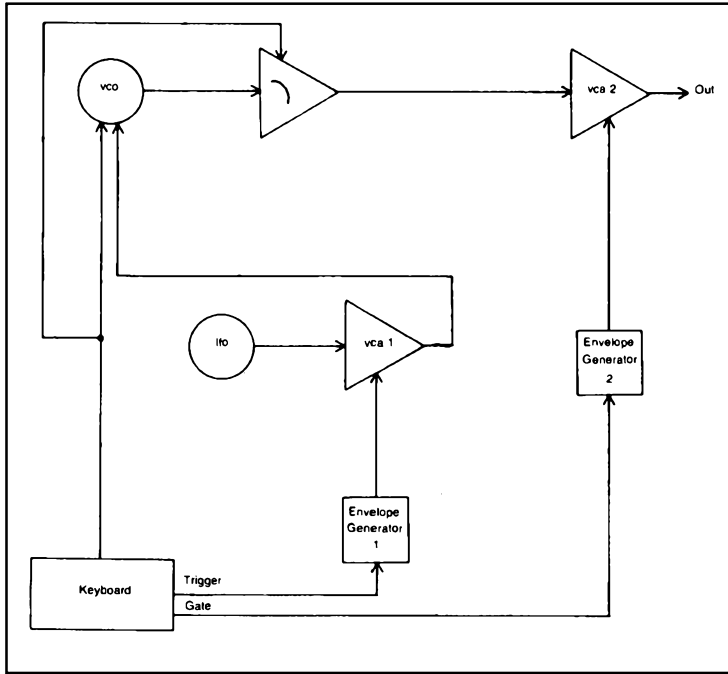


Fig. 10-52. Patch 17.

Patch 17. Figure 10-52 shows our next sample patch. The required modules for setting up this patch are outlined in Table 10-23.

Here we have the basic patch with a modification. The keyboard controls the vco and vcf and triggers the envelope generator which triggers the patch's main vca (vca2). An lfo is also used to control the vco to add some vibrato. However, rather than being applied directly to the vco's control voltage input, the lfo signal is first passed through vca 1 and given an envelope by envelope generator 1. Rather than a straight vibrato, the vibrato will start out very shallow (and probably inaudible) and will increase in

VCO	1
vcf, low-pass	1
lfo	1
vca	2
Envelope Generator	2
Keyboard	1

Table 10-23. Modules for Fig. 10-52.

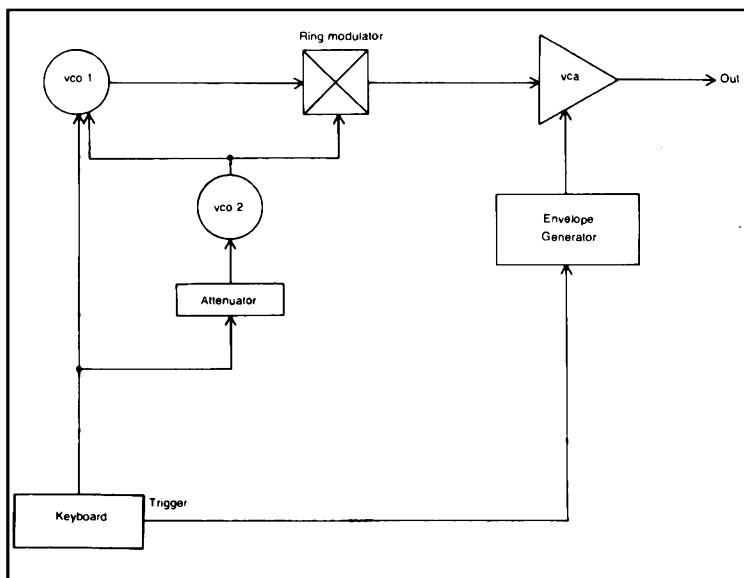


Fig. 10-53. Patch 18.

depth with the envelope's attack and then decrease back to a minimum during the decay (release) portion of the envelope.

While this trick, like any other, could get a little tiresome if overdone, it is an excellent occasional alternative to straight, continuous vibrato.

Patch 18. The next patch is shown in Fig. 10-53, and the required modules are listed in Table 10-24.

The keyboard controls two vcOs, and their outputs are ring modulated together. The attenuator is used to set the vco frequencies to an interval other than unison. The results can be especially intriguing if vco2 is set exactly one octave lower than vco1.

You should also notice that vco2 frequency modulates vco1. This adds to the striking effects produced by this patch.

Remember that FM and ring modulation effects tend to sound the best with relatively simple waveforms (sine waves, or triangle

VCO	2
Ring Modulator	1
vca	1
Envelope Generator	1
Attenuator	1
Keyboard	1

Table 10-24. Modules for Fig. 10-53.

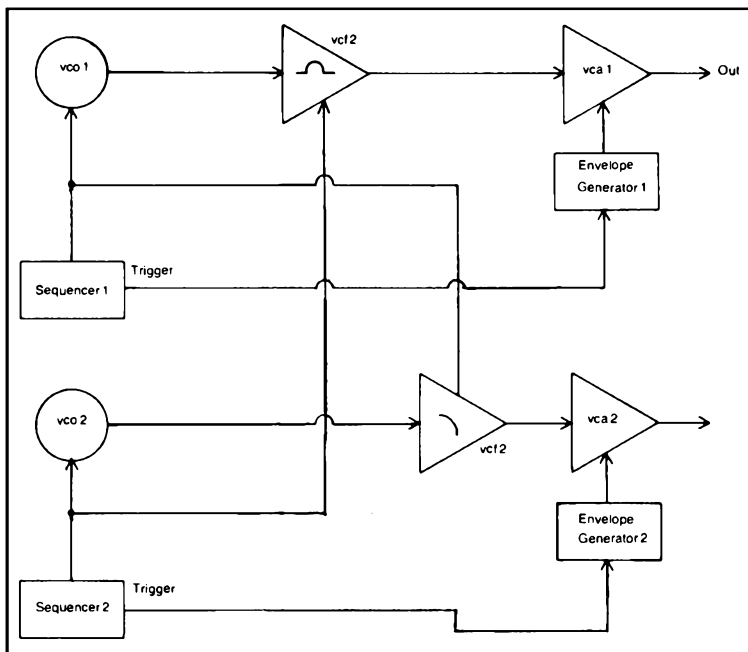


Fig. 10-54. Patch 19.

waves). This is especially true in this patch where both FM and ring modulation are used simultaneously. Complex waveforms from the vcOs will tend to create a rather muddy sounding and overly complex output without a strong sense of pitch.

Patch 19. The nineteenth sample patch is illustrated in Fig. 10-54. The modules needed for this patch are listed in Table 10-25.

This is a self-playing patch controlled by two sequencers. The results can be particularly fascinating if the sequencers are clocked at slightly different rates.

One of the sequencers controls vco1 and vcf2, which filters the output of vco2. Vco2's output frequency is controlled by the other sequencer, which also controls vcf1. Vcf1 filters the output of vco1.

vco	2
vcf, low-pass	1
vcf, band-pass	1
vca	2
Envelope Generator	2
Sequencer	2

Table 10-25. Modules for Fig. 10-54.

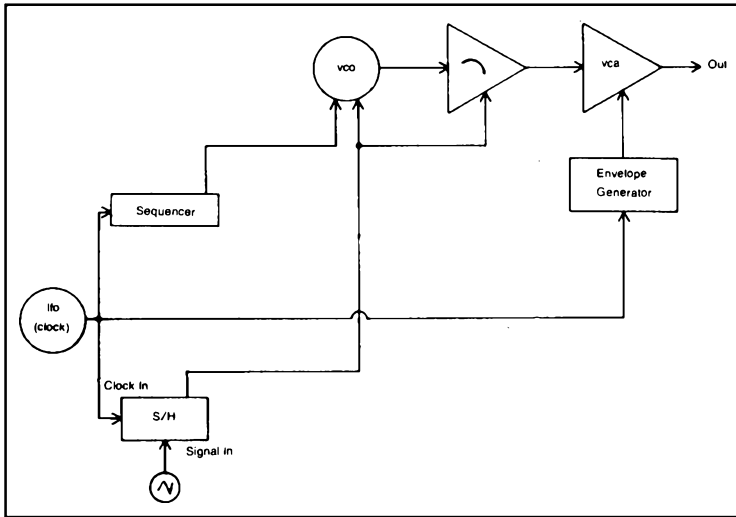


Fig. 10-55. Patch 20.

This cross control can create a charming, continuously changing sound quality.

Separate stereo outputs could be used, as shown in the diagram (this patch is especially effective when heard through stereo headphones), or the two signals may be combined with a mixer.

Patch 20. Continuing with self-playing patches, we come to Patch 20 which is illustrated in Fig. 10-55. The required modules are listed in Table 10-26.

A low frequency oscillator is used to simultaneously control the vco. A single lfo is used to clock the sequencer and the S/H and trigger the envelope generator controlling the vca. The sequencer will try to impose a regular pattern on the output frequency of the vco, while the S/H will force a strong element of randomness into the output. The result will sound random, although not quite as random (and musically pointless) as in a simple sample and hold patch.

vco	1
Noise Generator	1
lfo, clock	1
vcf, low-pass	1
vca	1
Envelope Generator	1
Sample and Hold	1
Sequencer	1

Table 10-26. Modules for Fig. 10-55.

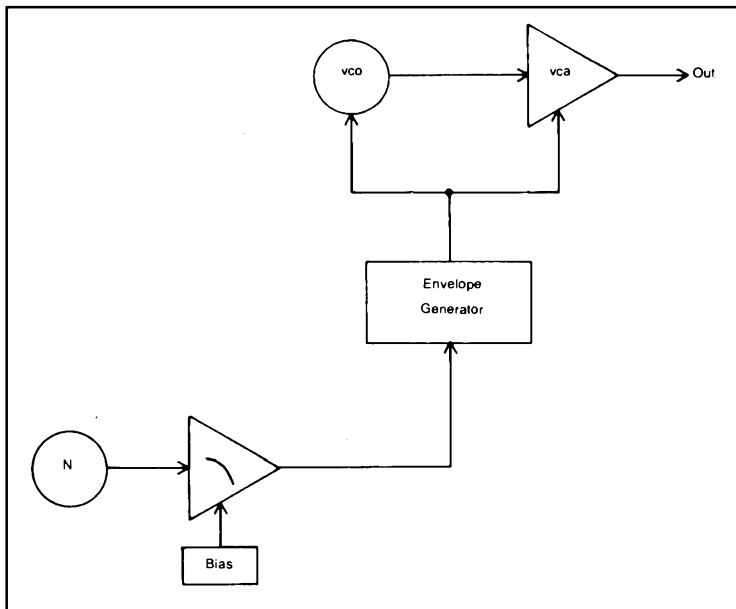


Fig. 10-56. Patch 21.

Patch 21. Patch 21 is shown in Fig. 10-56. The required modules are outlined in Table 10-27.

The envelope generator controls both the vca level and the vco frequency. The envelope generator is triggered by low-pass filtered noise. The bias voltage should be set so that the envelope generator is not triggered too often.

This patch works best with very short envelopes (short attack, no sustain, short decay or release). At some settings, the output will sound rather like a coffee perculator or like popcorn popping.

Patch 22. The next patch, which is shown in Fig. 10-57, is a rather unusual self-playing patch. The modules needed to set this one up are listed in Table 10-28.

vco	1
Noise Generator	1
vco, low-pass	1
vca	1
Envelope Generator	1
Bias (fixed voltage source)	1

Table 10-27. Modules for Fig. 10-56.

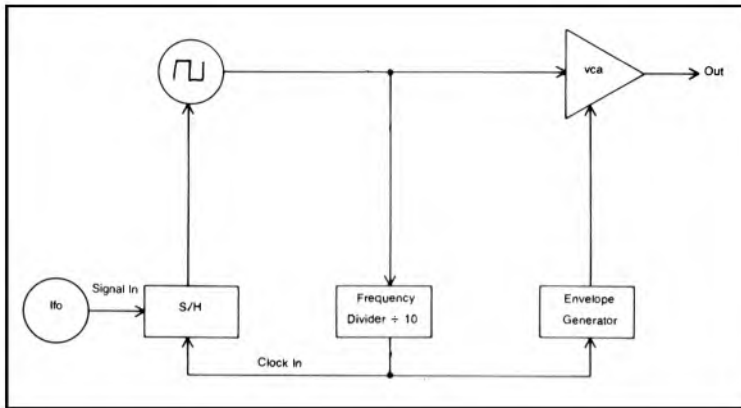


Fig. 10-57. Patch 22.

The output frequency of the vco is controlled by the sample and hold, which in turn is triggered by the output of the vco. The vco's output is stepped down to a tenth of its original frequency with the frequency divider. The new frequency is used to clock the sample and hold and trigger the envelope generator. Every ten cycles of the output signal will cause the frequency to change. Very short envelopes are mandatory for making this patch work properly. The vco should also be biased to generate relatively low frequencies.

You could try cascading two or more frequency dividers to make each note longer.

Patch 23. A frequency divider is also employed in Patch 23 (Fig. 10-58 and Table 10-29). Here the vco signal is split with one portion passing through a vca and the other being divided and passed through a separate vca. Both vcas are controlled by a single envelope generator, but the envelope signal is inverted before being fed to one of the vcas. This means that as one of the frequencies increase in volume the other will decrease. The two outputs are mixed together and band-pass filtered.

vco, square wave	1
lfo	1
vca	1
Frequency Divider (÷10)	1
Sample and Hold	1
Envelope Generator	1

Table 10-28. Modules for Fig. 10-57.

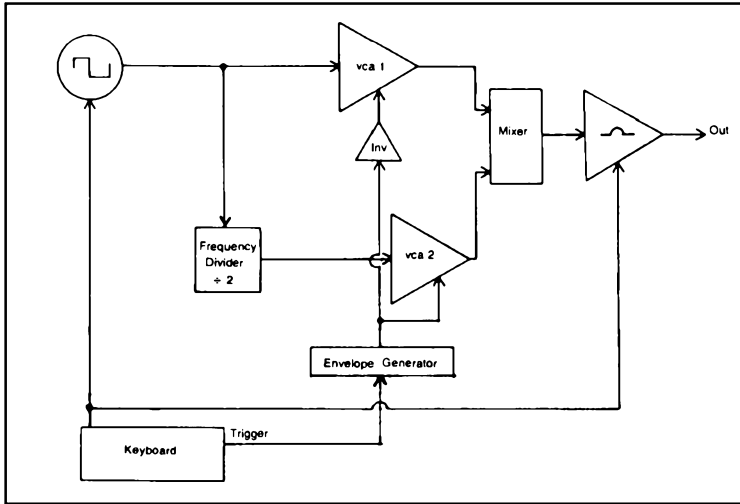


Fig. 10-58. Patch 23.

While not shown in the diagram, it might be a good idea to include a master vca (controlled by its own envelope generator) right after the vcf to prevent any signal from leaking through to the output when no key is being depressed.

Patch 24. The next sample patch is illustrated in Fig. 10-59, with the required modules listed in Table 10-30.

This is essentially the standard basic synthesizer patch, except the vcf is controlled by the keyboard and a delayed replica of the envelope driving the vca. This causes a shifting of harmonic content that is not quite in step with the changing amplitude, and therefore sounds a bit more natural.

Patch 25. In Patch 25 (Fig. 10-60 and Table 10-31), the keyboard controls three vcOs in unison, each with a different waveform. Each vco signal has its own individual vca controlled by its own envelope generator. The trigger signal is delayed before

vco, square wave	1
vcf, band-pass	1
vca	2
Frequency Divider	1
Inverter	1
Envelope Generator	1
Keyboard	1
Mixer	1

Table 10-29. Modules for Fig. 10-58.

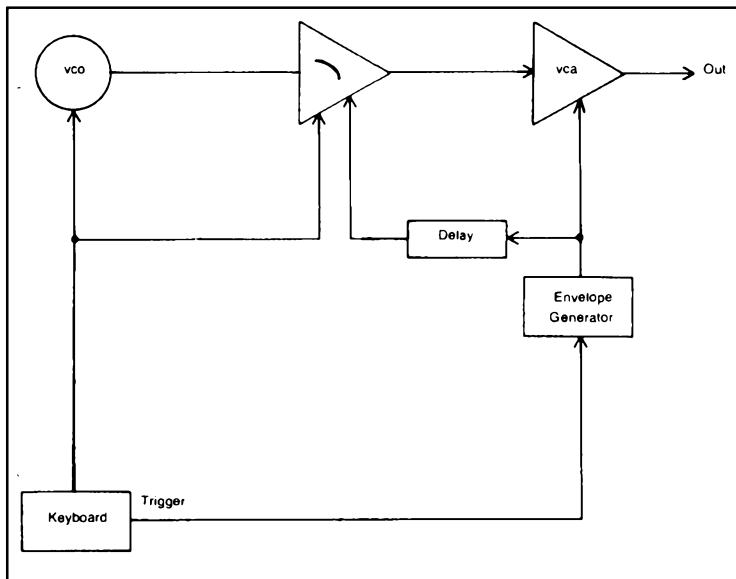


Fig. 10-59. Patch 24.

triggering envelope generator 2 and delayed even more before it reaches envelope generator 1. The three outputs are mixed together, resulting in an output with a complexly shifting pattern of harmonics. Personally, I find the sound created by this patch fascinating.

Patch 26. The next patch, shown in Fig. 10-61, is based on a similar idea as the previous example. Multiple signals are faded in and out of the output by delayed envelopes.

Signal path one includes vco1 and the synthi-waa module which was presented in Chapter 9. This signal is fed directly into the mixer. Only the master vca envelope (vca3 and envelope generator 2) controls its amplitude.

Signal path two begins with vco2. This signal is band-pass filtered. The vcf is controlled by the same lfo governing the

vco	1
vcf, low-pass	1
vca	1
Delay	1
Envelope Generator	1
Keyboard	1

Table 10-30. Modules for Fig. 10-59.

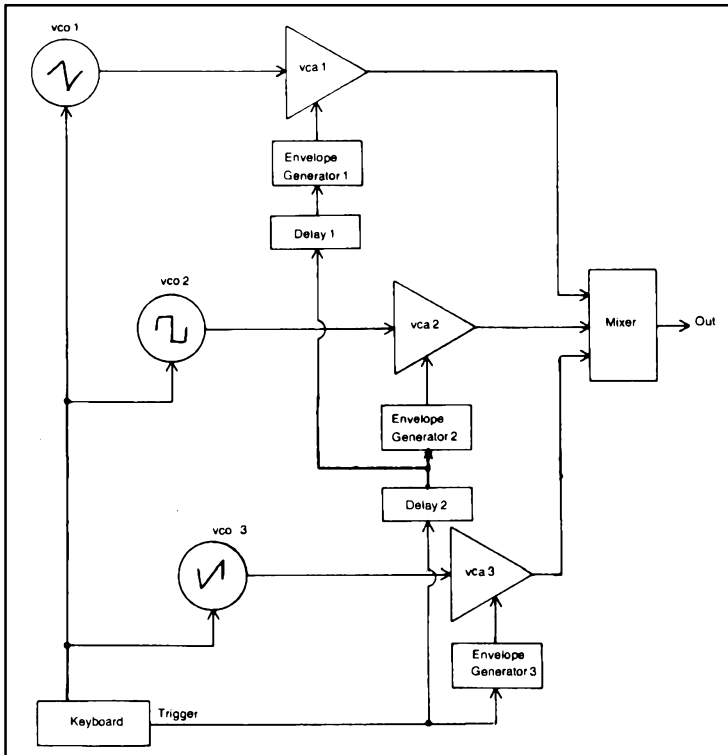


Fig. 10-60. Patch 25.

synthi-waa module in signal path one. This signal path includes an individual vca which is controlled by the delayed signal from envelope generator 1.

Signal path 3 includes vco3, the rotating speaker effect simulator (see Chapter 9), and vca2. Envelope generator 1 controls vca2 directly. Notice that the trigger input for this envelope generator is delayed.

vco, triangle wave	1
vco, square wave	1
vco, sawtooth wave	1
vca	3
Delay	2
Envelope Generator	3
Keyboard	1
Mixer	1

Table 10-31. Modules for Fig. 10-60.

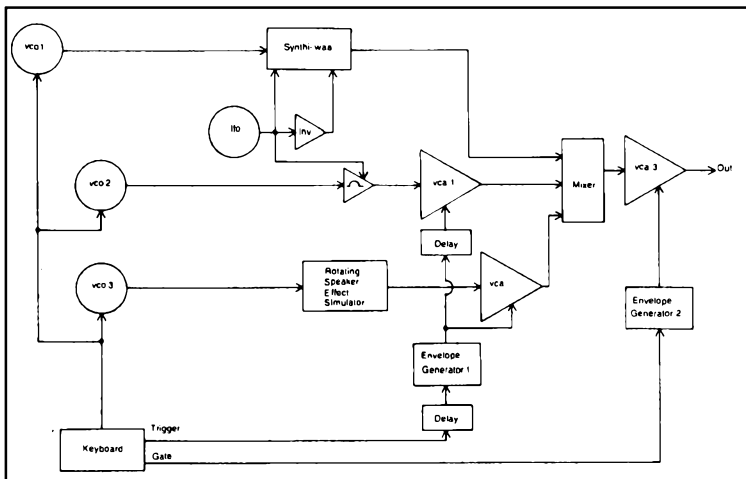


Fig. 10-61. Patch 26.

All three vcos are controlled by the keyboard. They may be tuned to unison, or to any desired interval. The waveshapes may be identical for all three oscillators, or contrasting waveshapes may be used.

This is a highly versatile patch, and many different effects can be achieved simply by re-tuning the vcos, by changing one or more of the waveshapes generated, or by adjusting the settings of the two envelope generators. Figure 10-62 gives a clearer idea of how this patch works by illustrating a typical set of envelopes and their effects on the output.

Many variations on this patch are also possible. You can substitute almost any module in any of the signal lines in place of those shown. The modules specifically shown in the patch diagram of Fig. 10-61 are listed in Table 10-32.

vco	3
lfo	1
vcf, band-pass	1
vca	3
Synthi-waa	1
Rotating Speaker Effect Simulator	1
Inverter	1
Delay	2
Envelope Generator	2
Keyboard	1
Mixer	1

Table 10-32. Modules for Fig. 10-61.

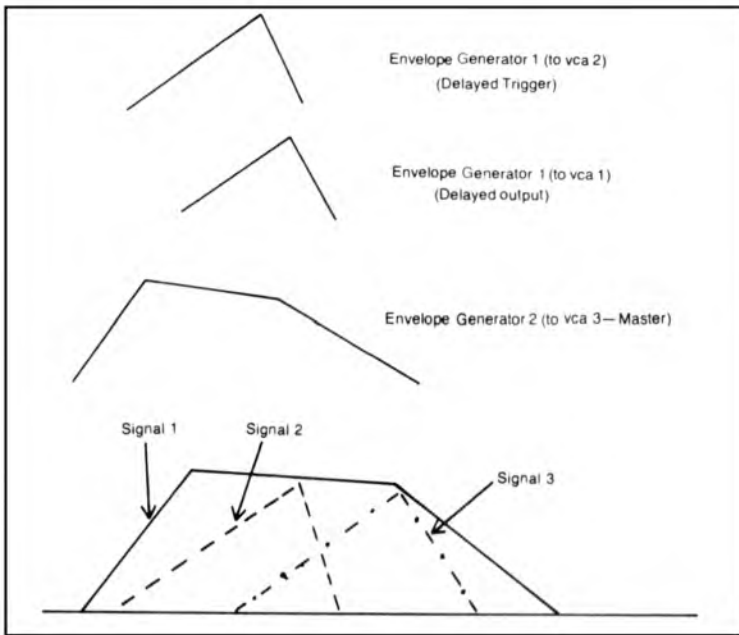


Fig. 10-62. Typical envelopes for patch 26 (Fig. 10-61).

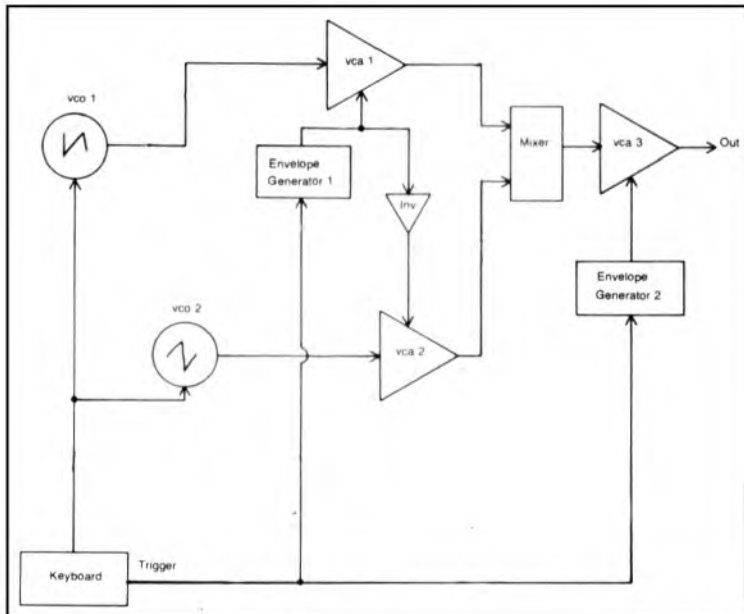


Fig. 10-63. Patch 27.

vco, triangle wave	1
vco, sawtooth wave	1
vca	3
Inverter	1
Envelope Generator	2
Keyboard	1
Mixer	1

Table 10-33. Modules for Fig. 10-63.

Patch 27. In the patch shown in Fig. 10-63, the output signal will start out as a triangle wave, fade into a sawtooth wave, and back into a triangle wave. The required modules for this patch are listed in Table 10-33.

For best results, envelope generator 1 should be set to produce a slightly shorter envelope than the one from envelope generator 2.

A variation on this patch is illustrated in Fig. 10-64. The required modules are listed in Table 10-34. Here are two initial signal selecting vcas (vca1 and vca2) are controlled by a lfo rather than an envelope generator. The two waveshapes will change back and forth at a rate determined by the lfo frequency. Some very

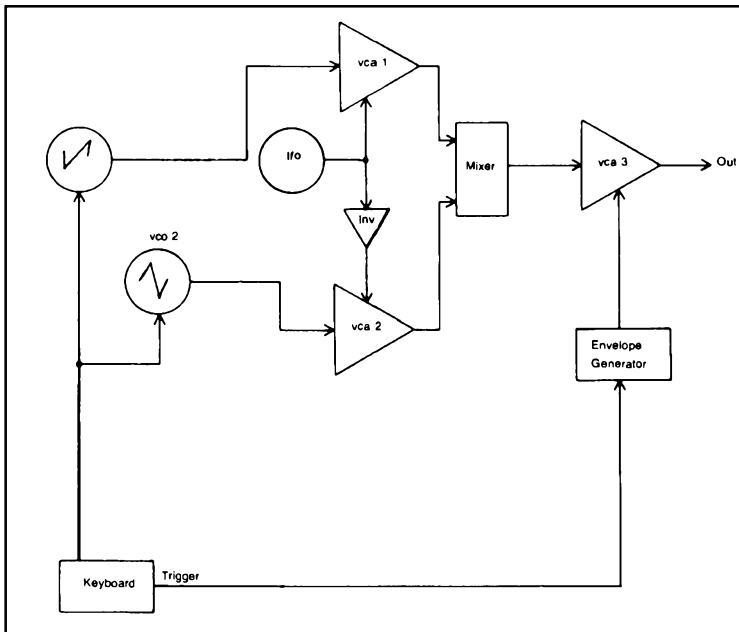


Fig. 10-64. Patch 27 variation.

vco, triangle wave	1
vco, sawtooth wave	1
lfo	1
vca	3
Inverter	1
Envelope Generator	1
Keyboard	1
Mixer	1

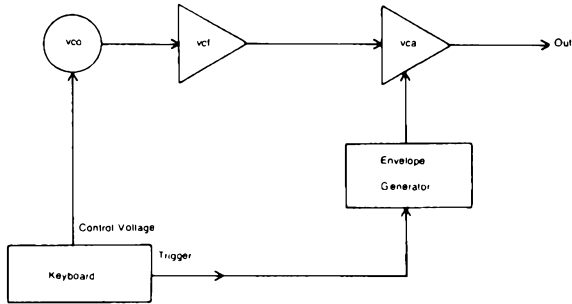
Table 10-34. Modules for Fig. 10-64.

offbeat modulation effects can be achieved by raising the lfo output into the audible range.

The two vcOs should be tuned in unison for either version of this patch.

Your Own Ideas. In this section, several patching ideas have been presented. Some were fairly simple, and others were fairly complex, but all offer a multitude of possibilities. Experiment freely, and don't be afraid to try out your own ideas. You may come up with some absolutely atrocious sounds, but what harm does that do? If a patch sounds bad, change it until you come up with something you like. Remember, you might stumble upon some really terrific sounds—but not if you don't experiment!

Chapter 11



Using a Computer

With the growing popularity of relatively inexpensive home computers, it is inevitable that more and more electronic music enthusiasts are employing their computers to musical ends.

Computer music covers a broad area, and would warrant a full volume devoted to the subject to even begin to do it justice. Computers can be used to synthesize sounds directly, or to control, or modify external signals. Computers have been used to analyze, and even compose musical scores. Traditional sounds can be digitally analyzed for more accurate synthesis or storage. A number of other applications have also been found in recent years, and more will probably follow.

Since this book is dedicated to hardware modules, we will restrict our discussion here primarily to the methods of interfacing a computer with an external electronic music synthesizer system.

ANALOG AND DIGITAL SIGNALS

Most of the circuits that have been presented in this book have been analog circuits. This means a theoretically infinite number of specific voltages may appear at the output of such a circuit. Analog circuits are often called linear circuits, because the output signals are capable of smoothly changing from value to value. Possible analog circuit outputs are illustrated in Fig. 11-1.

A digital output, on the other hand, can have only two possible states: high voltage, or low voltage (usually close to ground). No

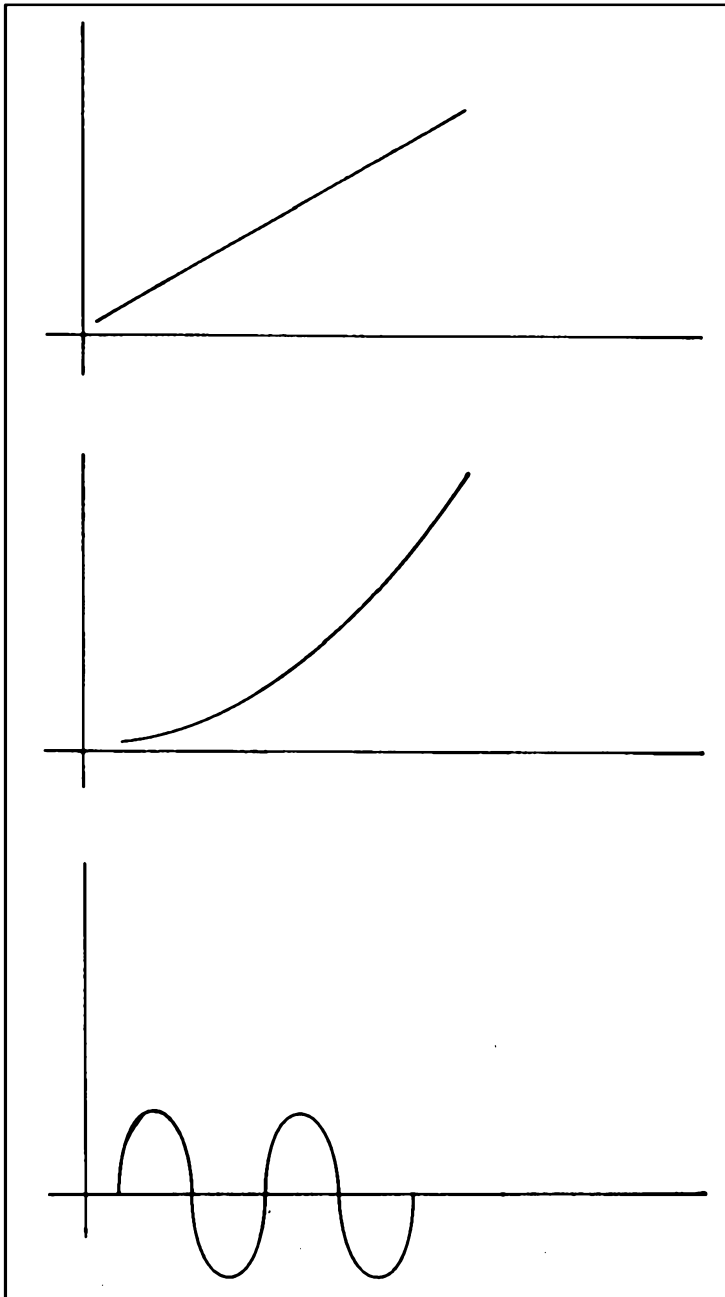


Fig. 11-1. Typical analog signals.

in-between states may ever occur. A typical digital signal is shown in Fig. 11-2. Notice its resemblance to rectangle waves.

Some digital circuits are three state devices, with an extra high impedance output, but this need not concern us here.

Now a single digital signal like this is not terribly versatile in its applications. Most practical digital circuits have multiple outputs. Each output can have only two possible states. Usually, these outputs are ranked from the most significant digit to the least significant digit. The combined outputs are written as binary numbers. A binary number uses only two digits: 1 and 0. There is no need for you to worry about doing any math or conversions with binary numbers here. You should know that each binary number has a decimal (the way we usually count) equivalent.

For example, a digital device with four outputs has sixteen possible output combinations that can be numbered from 0 to 15, as follows:

0000	0
0001	1
0010	2
0011	3
0100	4
0101	5
0110	6
0111	7
1000	8
1001	9
1010	10
1011	11
1100	12
1101	13
1110	14
1111	15

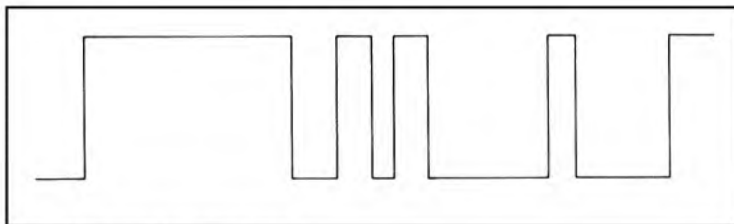


Fig. 11-2. Typical digital signal.

Notice that there is no such thing as, say, 8 . No in between values are possible.

Computers can only deal with binary numbers. Obviously, conversion circuits are required to use a computer with analog equipment like the synthesis modules presented in this book.

D/A CONVERTERS

A circuit that turns a multiple line digital signal (or binary number) into an analog voltage is called a *D/A converter*. D/A, of course, is short for digital to analog.

The simplest approach to D/A conversion is to use increasing resistances to give different weights to each digit (input line) of the binary number. This idea is illustrated in Fig. 11-3. Each digit's resistor has exactly half the value of the next lower digit, so the most significant digit will face the least resistance and produce the highest output voltage, while the least significant digit will face the maximum resistance and produce the smallest output voltage.

Binary numbers with more than one digit set at 1 produce values that are the sums of the appropriate digits. Each combination of digital inputs produces a specific and unique output voltage. There is no way to produce in between voltages.

This simple D/A converter can be functional, but it is not very practical. The resistors must be precise multiples of each other. Also, the resistance values vary greatly in step with the number of digital inputs. For an eight input D/A converter we would need additional resistances of $R/16$, $R/32$, $R/64$, and $R/128$. Besides being difficult to obtain the resistors in the correct values, the wide range of resistances could cause severe impedance and current draw problems.

A more practical approach to D/A conversion is the R-2R ladder network shown in Fig. 11-4. Here only two resistor values

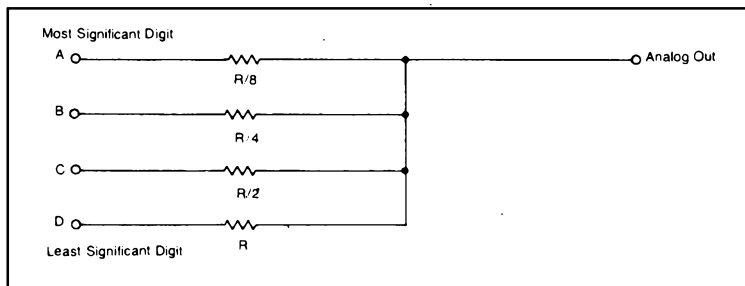


Fig. 11-3. Simple D/A conversion.

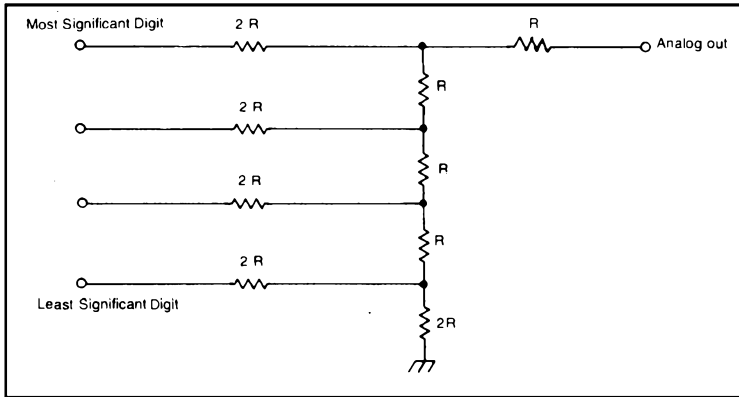


Fig. 11-4. R-2R ladder D/A converter.

are required: R , and its double $2R$. In fact, a circuit like this could be constructed using all resistors of a single value if each $2R$ resistance is made up of two R resistors in series.

Essentially, the R-2R ladder network works in about the same way as the simple D/A converter that was shown back in Fig. 11-3. The least significant digit faces more resistance (and thus voltage

Table 11-1. Outputs for a Typical D/A Converter.

Digital Inputs				Decimal Equivalent	Output Voltage
A	B	C	D		
0	0	0	0	0	0
0	0	0	1	1	0.31
0	0	1	0	2	0.62
0	0	1	1	3	0.94
0	1	0	0	4	1.25
0	1	0	1	5	1.56
0	1	1	0	6	1.87
0	1	1	1	7	2.19
1	0	0	0	8	2.50
1	0	0	1	9	2.81
1	0	1	0	10	3.12
1	0	1	1	11	3.44
1	1	0	0	12	3.75
1	1	0	1	13	4.06
1	1	1	0	14	4.37
1	1	1	1	15	4.69

(Output voltages are rounded off to two decimal places.)

drop) than the most significant digit. Combinations of digits are combined to create intermediate steps. However, no voltages between the step values can be achieved. Typical step values are listed in Table 11-1.

This simple resistor network can be badly loaded down by the impedance of the module it is driving. Moreover, the output voltages might not be in range needed. These problems can be solved by adding a preamplifier circuit to the output, as shown in Fig. 11-5. Op amp IC1 is the preamp. Since this amplifier inverts the signal, a second op amp (IC2) is added as a unity gain inverter to restore the original polarity.

The output of a D/A converter can be used as a control voltage source for virtually any voltage controlled synthesis module. This allows almost full computer control of your synthesizer. The computer may act as a super sequencer, or it may perform internal calculations with the results determining the output voltage.

A computer and D/A converter combination can also be used as a versatile ac signal source. By repeatedly outputting a specific pattern of binary numbers, almost any waveform may be created. For example, the following sequence simulates a triangle wave:

0000
0001
0010
0011
0100

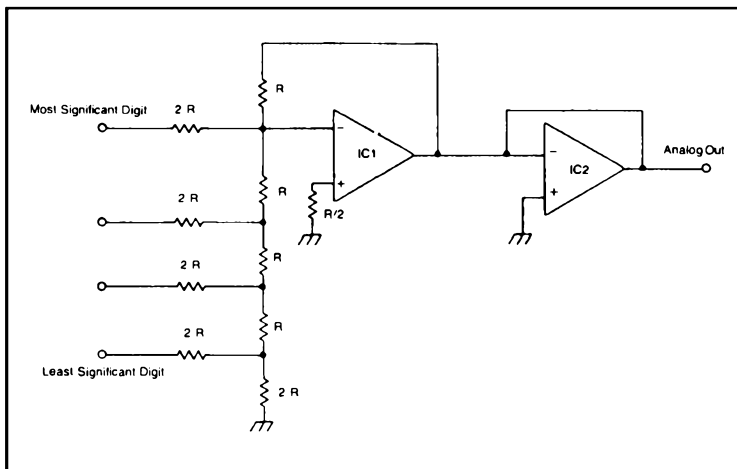


Fig. 11-5. R-2R ladder D/A converter with preamp.

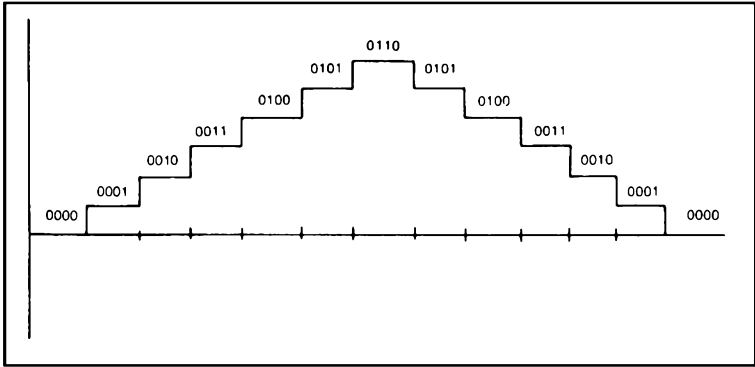


Fig. 11-6. Digitally generated triangle wave.

0101
 0110
 0101
 0100
 0011
 0010
 0001
 0000

This digital waveform is shown in Fig. 11-6. It can be improved by placing a relatively large capacitor across the D/A converter output to smooth out the steps of the waveform, as shown in Fig. 11-7.

The frequency of the waveform will be determined by how rapidly the computer cycles through the pattern.

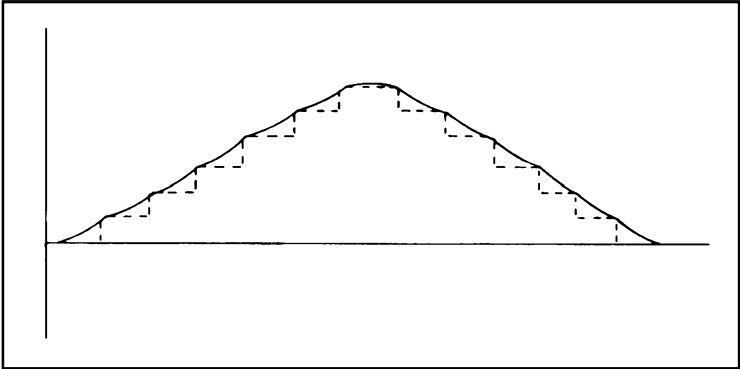


Fig. 11-7. Digitally generated triangle wave smoothed by a capacitor.

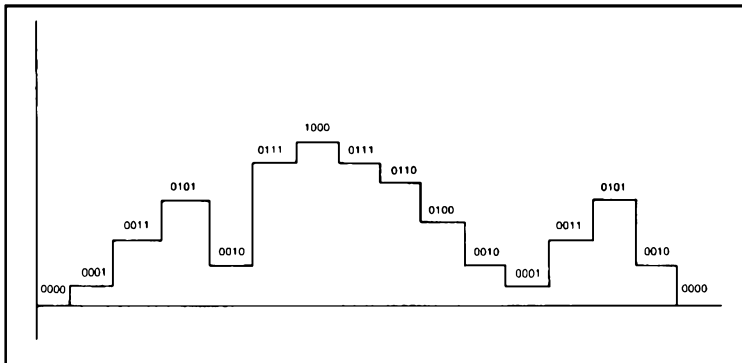


Fig. 11 -8. Unusual digitally generated waveform.

Besides synthesizing traditional waveshapes, a computer can easily generate many unusual waveforms that would be difficult to create by analog circuitry. For example, the waveform shown in Fig. 11-8 can be created with the following binary sequence:

0000
 0001
 0011
 0101
 0010
 0111
 1000
 0111
 0110
 0100
 0010
 0001

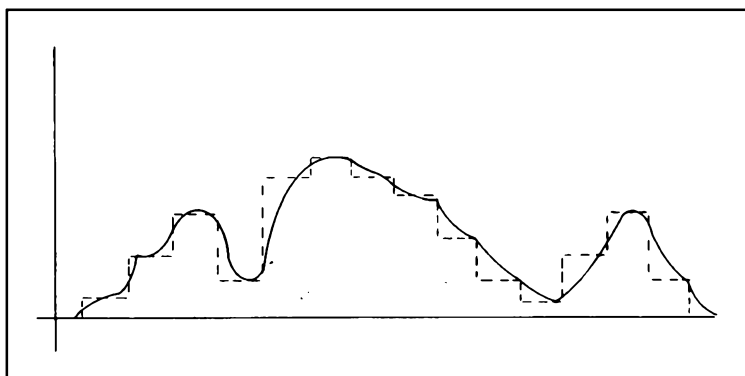


Fig. 11 -9. Smoothing the waveform from Fig. 11 -8 with a capacitor.

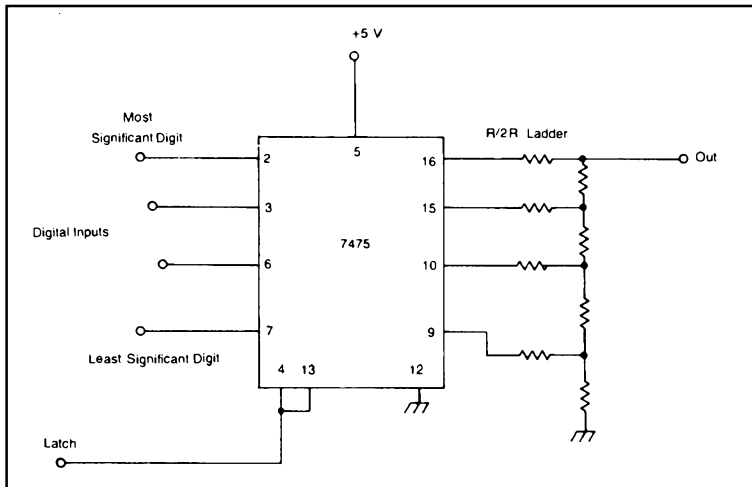


Fig. 11-10. Adding flip-flop latches to a D/A converter.

0001
 0011
 0101
 0010
 0000

Figure 11-9 shows the waveform when smoothed with a capacitor.

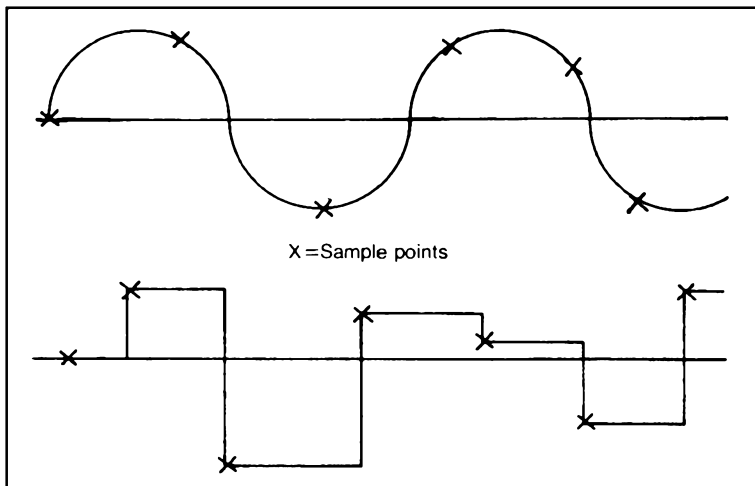


Fig. 11-11. A/D conversion with too few samples.

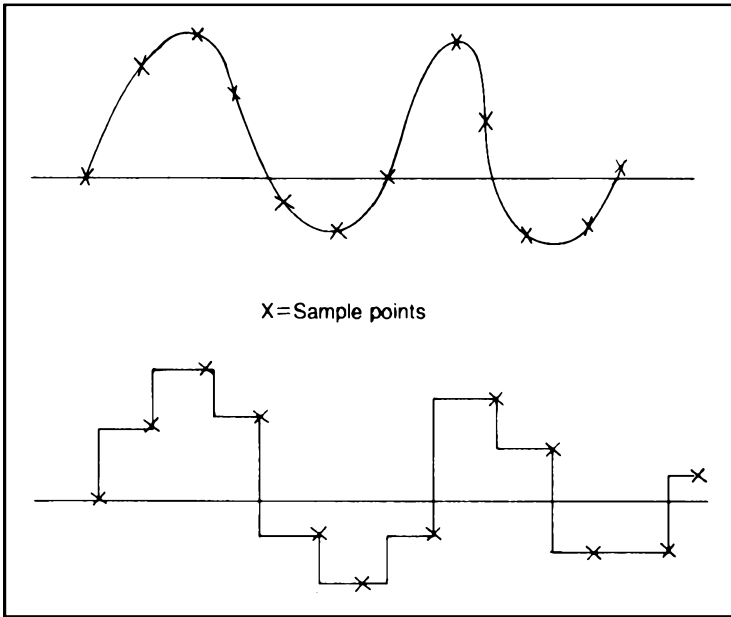


Fig. 11-12. Improved A/D conversion.

Computers are capable of very high frequencies. Most can output a new digital number at a rate of 1 MHz (1,000,000 Hz) to 8 MHz (8,000,000 Hz), or even higher. Sound synthesis requires much lower frequencies. The computer can repeatedly output the same number over and over as long as it is needed, but this wastes much of its capabilities.

To take better advantage of the computer's capabilities, the input of a D/A converter can be preceded by flip-flop latches, as illustrated in Fig. 11-10, to hold the current binary number until the computer tells the circuit it is ready to feed out a new number. This frees the computer to perform other operations while an output value is being held constant.

A/D CONVERTERS

The opposite of a D/A converter is an *A/D converter* (analog to digital). Obviously this is a device used to feed an external analog voltage into a computer in the form of a binary number for storage and/or mathematical manipulation. The digital signal will eventually be fed back out of the computer through a D/A converter.

A/D converters are not as widely used as D/A converters. This is partially because the circuitry required is more complex.

The analog signal must be sampled repeatedly at brief intervals to create a digital representation of the signal. If the sampling rate is too low, confusion about the signal could result, as illustrated in Fig. 11-11. A faster sampling rate improves fidelity (see Fig. 11-12) but increases cost and circuit complexity.

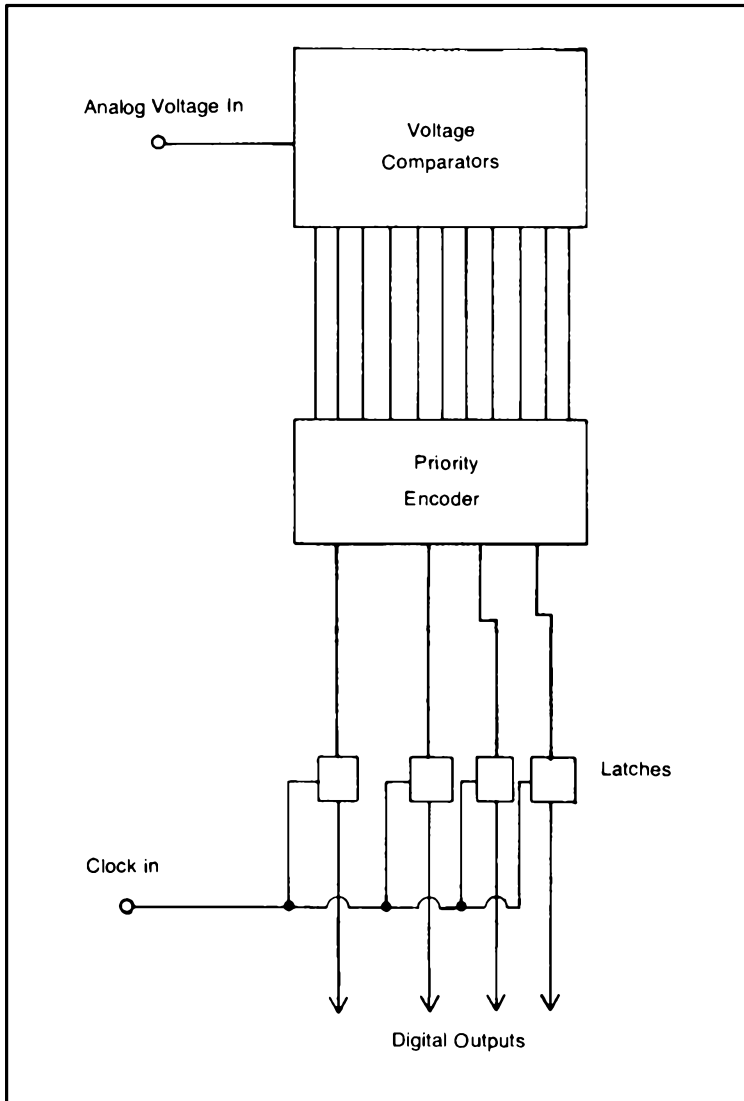


Fig. 11-13. Block diagram for a A/D converter.

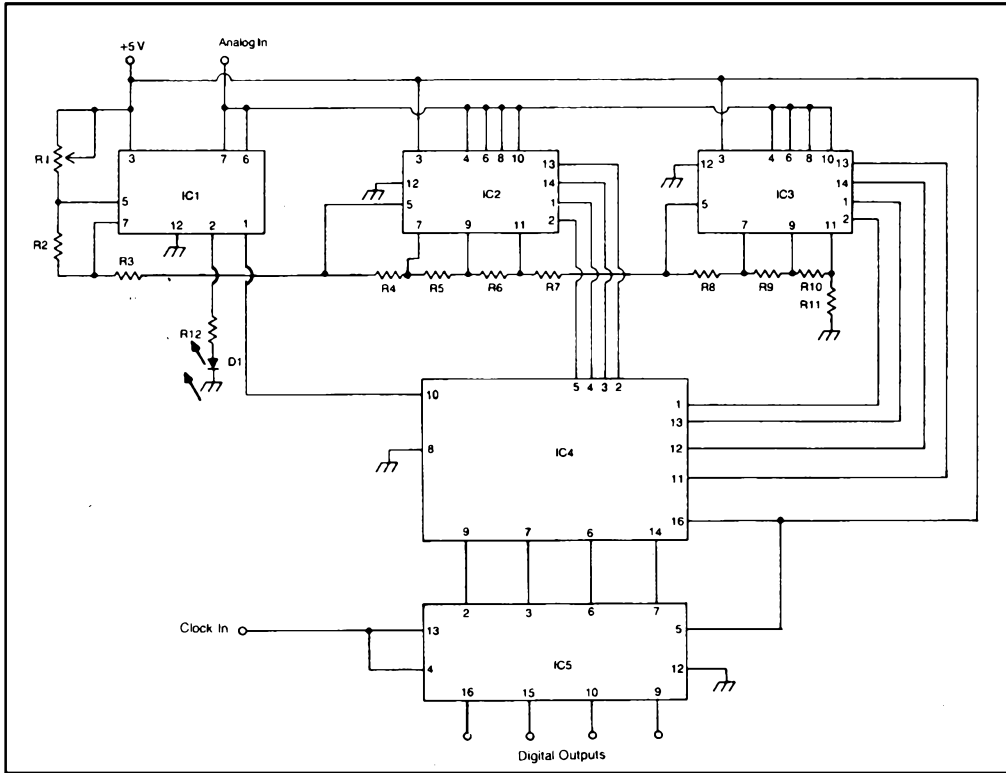


Fig. 11-14. Simple A/D converter.

Table 11-2. Parts List for Fig. 11-14.

R1	100 k Ω potentiometer (sensitivity control)
R2-R11	1 k Ω resistor
R12	390 Ω resistor
D1	LED (overflow indicator)
IC1, IC2, IC3	LM339 quad comparator IC
IC4	74147 priority encoder IC
IC5	7475 quad latch IC

Most A/D converters are based on sample and hold circuits. The clock frequency must be at the very least twice the maximum analog frequency to be converted, and preferably even higher.

A block diagram for a typical A/D converter circuit is shown in Fig. 11-13. A schematic for a relatively simple A/D converter is illustrated in Fig. 11-14.

There are a great many ways to use a computer with your electronic music synthesizer. In this chapter, we have barely skimmed the surface of the basics. As with everything else discussed in this book, use your imagination and experiment, experiment, experiment!

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Music Synthesizers: A Manual of Design & Construction

by Delton T. Horn

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Delton Horn is an electronics hobbyist who has written five previous books for TAB, including *The Beginner's Book of Electronic Music*, and *Electronic Music Synthesizers*.

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