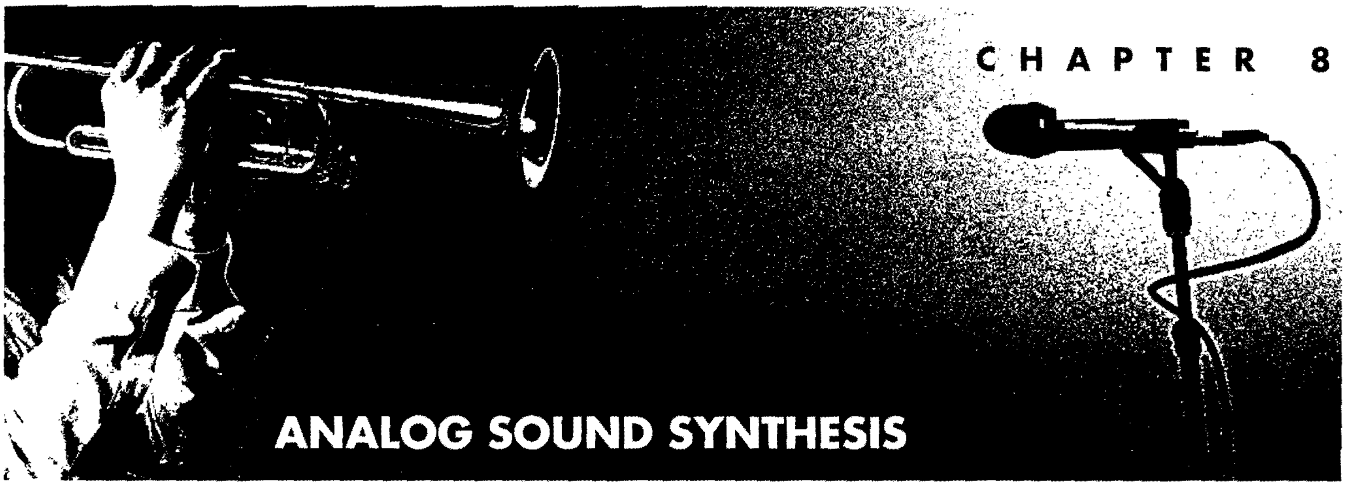


CHAPTER 8

ANALOG SOUND SYNTHESIS

Electronic sound synthesizers have been available to composers since the mid-1950s. Many synthesizers, particularly the most recent ones, deal primarily with numerical representations of electrical oscillations, and so are called **digital synthesizers.** These are described more completely in the next chapter.

Other synthesizers, generally older ones, work with the electrical oscillations themselves. These electrical waves are directly analogous to the vibrations of sound that are produced when the signal reaches the loudspeaker. For this reason, such synthesizers are identified as **analog synthesizers.** Although these are no longer in widespread use, many of the concepts of synthesizer operation introduced with them continue to be of great relevance.

The modern history of analog synthesis begins with the work of Herbert Eimert and his colleagues in the studio they established in 1951 at the station of the Northwest German Radio (or NWDR, for Nordwestdeutscher Rundfunk) in Cologne. The equipment in this studio included signal sources such as electronic oscillators and noise generators, and signal processors such as electronic filters, gates,[1] and related devices. Much of this equipment was generic test equipment that had been previously used elsewhere at the station. The essential equipment of the studio also included a variety of tape recorders. The signals that were generated and processed by the electronic equipment were recorded on tape, where they could then be edited, looped, mixed with other sounds, or otherwise manipulated.

1. A gate is an electronic device that regulates the passage of an electronic signal. When the gate is electronically open, the signal passes through; otherwise, it is blocked. By connecting the output of an electronic oscillator or noise generator (both of which produce a continuous signal) to the input of a gate device, it becomes possible to produce separate tones, with distinct beginnings and endings.

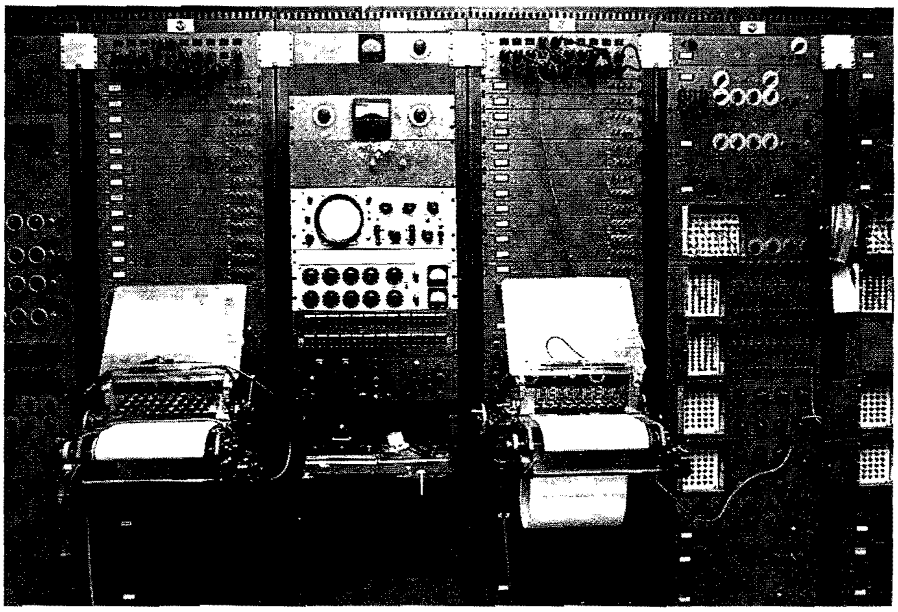


FIGURE 8.1
The RCA Mark II synthesizer. (Courtesy of the Electronic Music Center of Columbia University.)

The composers in the Cologne studio were attracted by the degree of control that these techniques provided (see the discussion in Chapter 10 of the *Gesang der Jünglinge* by Karlheinz Stockhausen). However, producing such ***elektronische Musik*** could be exceedingly tedious work. To record a simple series of pitches, for example, they had to tune the oscillator, record the tone, retune the oscillator, restart the recorder, and repeat the process until all the tones were on tape. Next, these separate tones had to be spliced together, with each segment of tape being measured carefully to ensure the accuracy of the rhythmic relationships among the tones.

The RCA Electronic Music Synthesizer, introduced in 1955, provided for the automation of many of these processes. The second model, called the Mark II, was built in 1957 at a cost of a quarter-million dollars; it was later placed with the Columbia-Princeton Electronic Music Center, jointly operated by Columbia and Princeton universities. The device, which included more than 1700 vacuum tubes, was housed in nine equipment racks, virtually covering an entire wall (and then some) in the studio in which it was located (see Figure 8.1). Needless to say, it was not an instrument intended for the onstage performances of an itinerant rock-and-roll band; it was most certainly a studio synthesizer.

Some rather remarkable claims were made at first regarding the capabilities of this synthesizer. It was said to be able to:

- "match the sound of any instrument or ensemble"[2]
- "project a speaking voice, even in familiar tones and accents"[3]

2. Howard Taubman, "Synthesized Piano Music Found to Have Tone Matching a Grand's: RCA Electronic Device Can Produce Ensemble and Voice Sounds Too—Musician's Role Still Important," *New York Times*, 1 February 1955, p. 35.

3. Ibid.

THE PREHISTORY OF ELECTRONIC MUSIC

The synthesizer was predated by dozens, if not hundreds, of electrical and elec[illegible] instruments. One of the most awesome of these was the Telharmonium (see F[illegible] developed around the turn of the century by the American inventor and lawyer T[illegible] Cahill. After working on some prototypes in the late 1890s, Cahill built the larges[illegible] the instrument at the beginning of this century. It was a large instrument indeed. R[illegible] have weighed at least 200 tons, it required 30 railroad cars to transport it to New Y[illegible] where it was installed in 1906 in the basement and first floor of a building at 39th S[illegible] Broadway called Telharmonic Hall. Its signal was sent out over special telephone [illegible] offered to subscribers for a monthly fee.

The vast bulk of the instrument was necessary because high voltages were require[illegible] operation. It was not until 1906 that the triode, a vacuum tube that made electronic a[illegible] tion possible, was invented—too late to be used in the Telharmonium. Although Cah[illegible] York Electric Music Company soon failed, many of his ideas were revived three decade[illegible] by Laurens Hammond and implemented with techniques of electronic amplification. B[illegible] this new instrument, the Hammond organ, was introduced; it has since become a class[illegible] instrument of gospel music and rock-and-roll.

One of the most unusual instruments ever invented is the theremin (see Figure 8.3[illegible] duced in 1920 by the Russian inventor Leo Theremin. It is unlike all other instruments [illegible] it is played by *not* touching it. The performer waves hands around its two metal antenna[illegible]— one for the control of pitch, and the other for amplitude. The pitch rises as the hand mo[illegible] closer to the pitch antenna. A vibrato is produced if the hand is shaken. Moving from one pitch to another requires sweeping through all of the pitches in between. Thus, great ski[illegible] required to slight these swoops so that the listener does not tire of the sound too quickly.

A few people established reputations as virtuosi on the theremin, including Clara Rock[illegible] whose performances can today be heard and seen on recordings and film. Theremin spent some time in New York City during the 1930s and then returned to the Soviet Union, where [illegible] was soon arrested by the Stalin regime. During his imprisonment, he developed aircraft tra[illegible]ing instruments and eavesdropping devices for the Soviet government; he was awarded a tec[illegible]nical prize for these "contributions" shortly after his release in 1947.

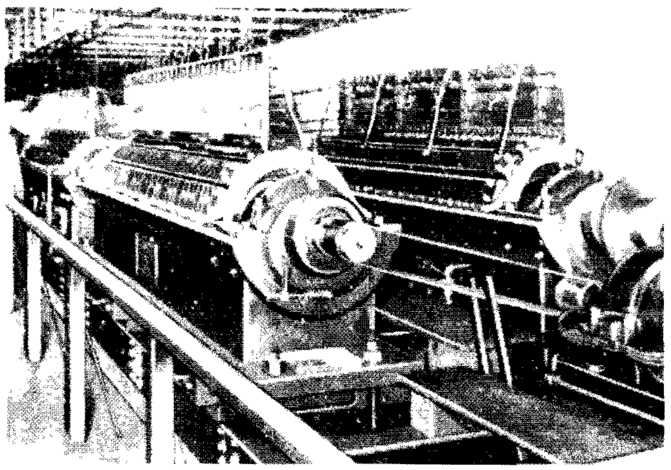
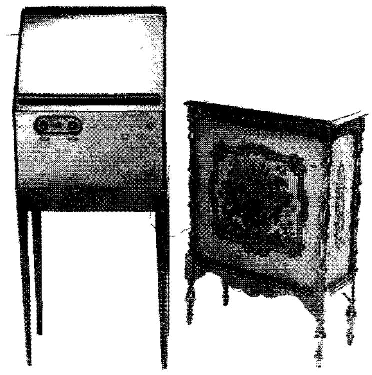


FIGURE 8.2
The dynamo room in the basement of Telharmonic Ha[illegible] This photograph (from the *Telharmonic Hall Program: Week of December 30th*, New York: Telharmonic Securities Co., 1907, p. 2) shows only a portion of Thaddeus Cahill's Telharmonium. (Copy photo courtesy of Reynold Weidenaar.)

FIGURE 8.3
A theremin. (Smithsonian Institution Photo No. 93-5639.)



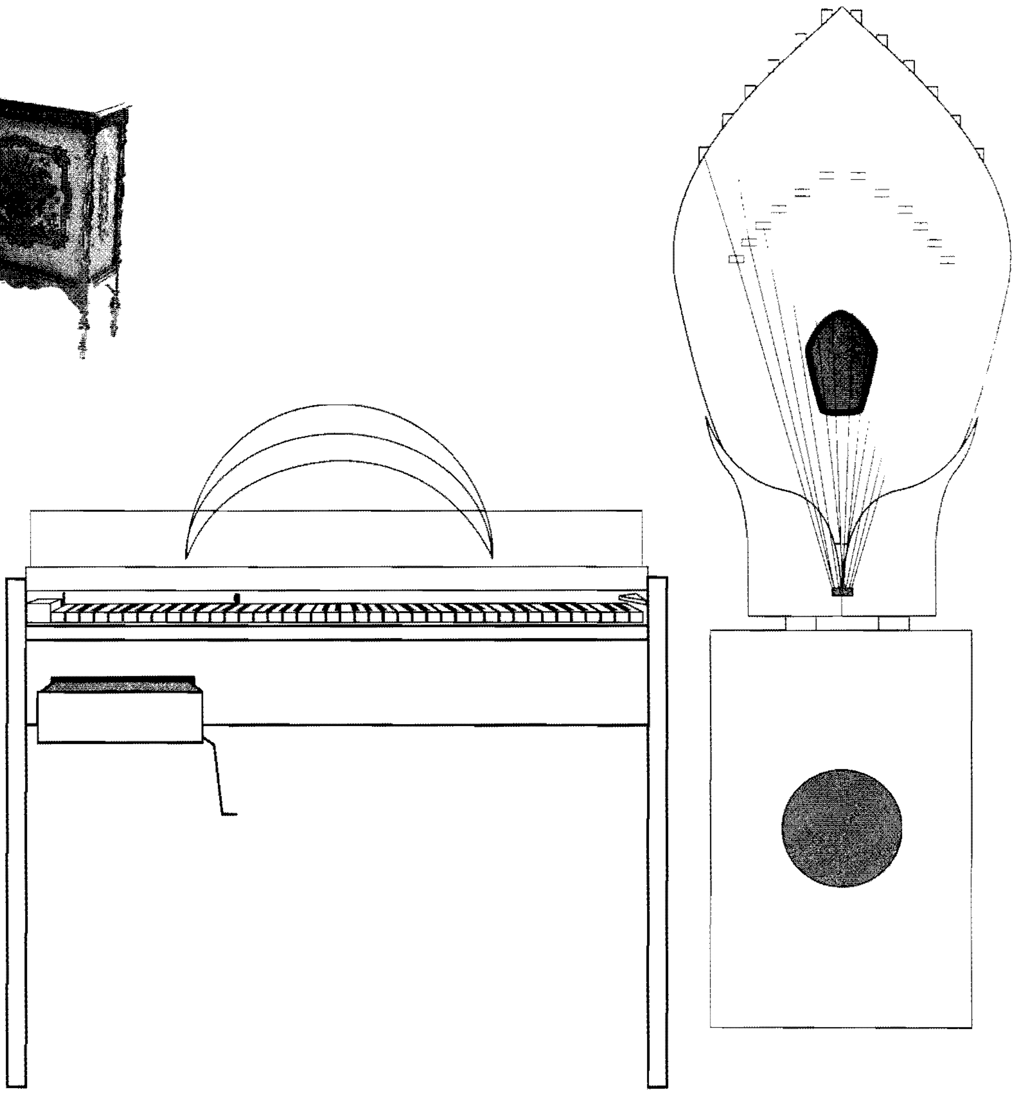


FIGURE 8.4
An *ondes martenot.*

Another well-established instrument from this period is the *ondes martenot* (see Figure 8.4), produced in Paris in 1928 by the French inventor Maurice Martenot. The instrument is based on the same principle of operation as the theremin, and sounds much the same, but is played by sliding a ring attached to a wire on a pulley. This mechanism is mounted over a painted keyboard to provide the performer with a reference for pitch (later versions of the instrument have had actual, working keyboards). Literally hundreds of pieces of music have been composed for the *ondes martenot,* including some excellent music by the eminent French composer Olivier Messiaen.

This provides only a glimpse of the early activities in this field. The reader is encouraged to explore many of the excellent writings and recordings cited at the end of this chapter.

FUNCTIONS ON A TYPICAL ANALOG SYNTHESIZER

The transistorized sound synthesizers that appeared in the 1960s were primarily studio instruments. Despite the reduction in size (at least in relation to the size of the RCA synthesizer) and the application of the concept of voltage control, they were still rather bulky and somewhat complicated to operate. Typically these instruments were designed as a collection of individual, electrically compatible components called **modules,** each dedicated to a specific function of sound synthesis, such as signal generation, gating, amplification, filtering, or control voltage generation. Thus, a particular sound synthesizer was likely to be a custom set of modules selected and arranged in the way considered best suited to the aesthetic aims of a particular studio.

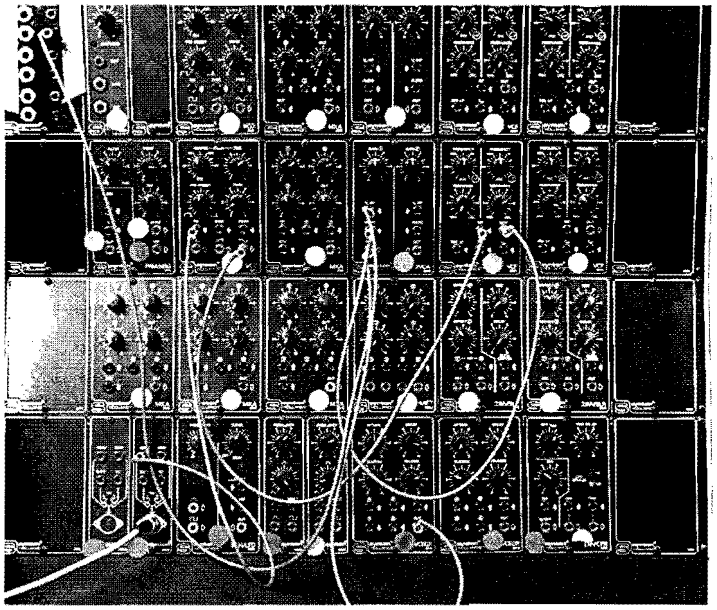
The inputs and outputs of a module are connected to those of other modules by patch cords (or, on some models, a matrix switch arrangement.) Thus, different subsets of the modules on a synthesizer can be connected in a variety of ways to produce an enormous variety of sounds (many of which may even be genuinely new and interesting). A particular combination of modules such as this is called a **patch.**

A module in a patch performs at least one of three basic functions: (1) It may function as the source of the audio signal, providing an oscillation or a band of noise. (2) It may process or modify the amplitude, spectrum, or some other attribute of the signal in some way. (3) A module may function as a source of control voltages that can be applied to the other modules in the patch.

Figure 8.5 includes a block diagram of a common synthesizer patch that illustrates how these three categories of modules relate to one another. The audio signal originates at the oscillator, then passes through the filter and the amplifier before being sent from the synthesizer to the tape recorder. The pitch of the oscillator is controlled by a voltage provided from the keyboard, and the level of the amplifier is controlled by a voltage from the envelope generator. The signals from the keyboard and the envelope generator are themselves never heard, but their effects on the oscillator and the amplifier are heard as changes in the pitch and loudness of the audio signal, respectively.

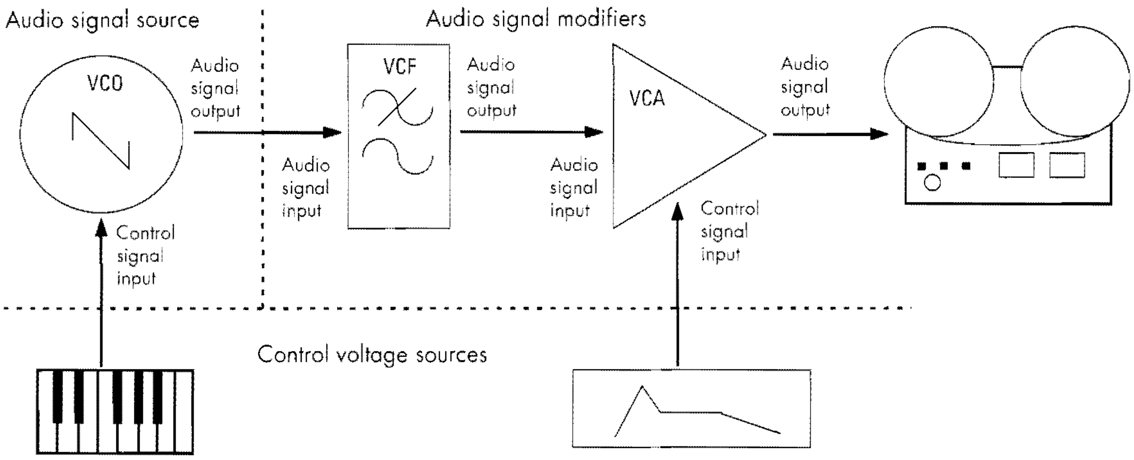
AUDIO SIGNAL SOURCES

Three types of modules can function as audio signal sources on an analog synthesizer: voltage-controlled oscillators, noise generators, and interfaces for external sources (such as microphones). The **voltage-controlled oscillator (VCO),** illustrated in Figure 8.6, is found on all analog synthesizers. In fact, there will most likely be at least two or three of them.



a. A photograph of the actual patch

FIGURE 8.5
A synthesizer patch.



b. A schematic diagram of the patch

FIGURE 8.6
A VCO.

A typical VCO will have a dial for coarse adjustments to the frequency of the signal produced by the module. The range of this dial usually covers a major portion of the range of frequencies audible to humans. Another dial, with a much narrower range, is used to make finer adjustments to the frequency. Some VCOs have a frequency range switch that is labeled after pipe organ stops—for example, 16'–8'–4'.[7] Such a switch is used to shift the frequency of the VCO up or down one or more octaves.

A VCO usually offers a selection of common waveforms. Some synthesizers have an output jack dedicated to each available waveform. Others have a single output jack, and the waveform is selected by a switch. One commonly available waveform is the sine wave, which has no overtones—only a fundamental frequency.[8]

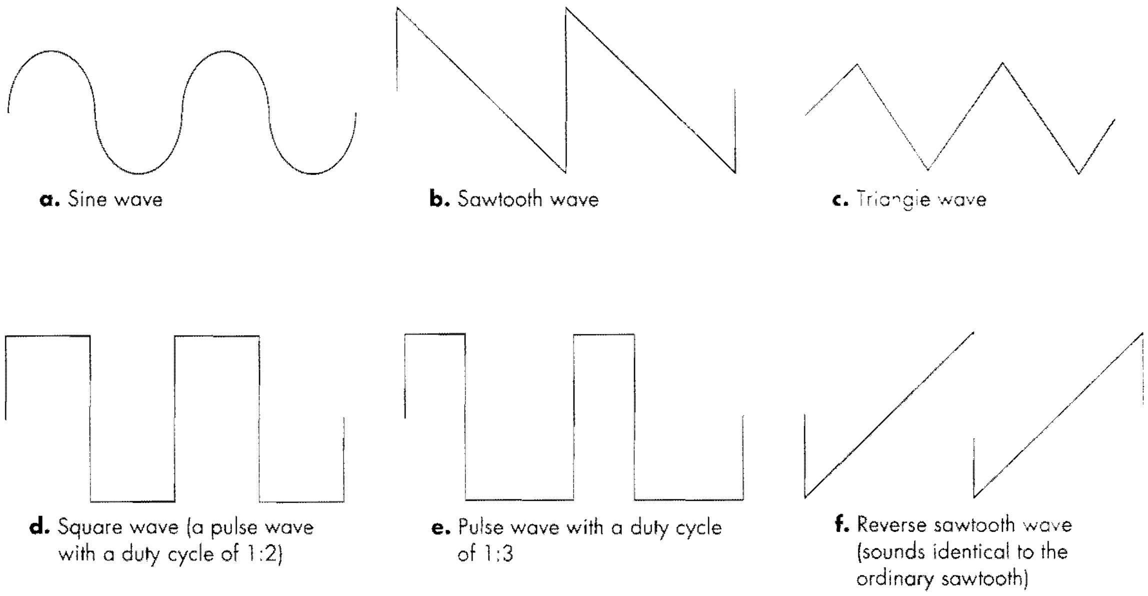
Virtually all VCOs can generate sawtooth waves (see Chapter 7), which have a spectrum that includes all partials. The amplitude of a particular partial, relative to that of the fundamental, is the inverse of its partial number ($1/n$, where n is the partial number). Thus, the amplitude of the second partial (the first overtone) is half that of the fundamental, the amplitude of the third partial (the second overtone) is one-third that of the fundamental, and so on (see Figure 7.5). The combined amplitude of the first three overtones is greater than that of the fundamental ($1/1 < 1/2 + 1/3 + 1/4$). Thus, most of the energy in the spectrum of a sawtooth wave is found in the overtones, making this a rather bright and penetrating timbre.

A **triangle wave** has a spectrum that includes only the odd-numbered partials. The amplitude of a particular partial, relative to that of the fundamental, is the square of the inverse of its partial number ($1/n^2$, where n is the partial number). Thus, the second partial is absent, the third partial has 1/9 the amplitude of the fundamental, the fourth partial is absent, the fifth partial has 1/25 the amplitude of the fundamental, and so forth. Most of the energy of a triangle wave is concentrated in the fundamental ($1/1 > 1/9 + 1/25 + 1/49 + \ldots$), resulting in a relatively mellow tone color.

A **pulse wave** has a waveform that is positive at a constant level for the first part of the cycle, then instantly drops to a negative level for the remainder of the cycle. The proportion of the positive portion of the cycle to the total length of the cycle is called the **duty cycle** of the wave. A **square wave** is a pulse wave with a duty cycle of 1:2. This means that the wave is positive at a constant level for the first half of the cycle and negative at a constant level for the other half of the cycle. The spectrum of a square wave includes only the odd-numbered partials; the amplitude of a particular partial, relative to that of the fundamental, is the inverse of its

7. An organ pipe 16 feet long, open at each end, will produce a pitch three octaves below middle C. An eight-foot organ pipe, open at each end, will produce a pitch two octaves lower than middle C, and a four-foot organ pipe will produce a pitch one octave below middle C. A two-foot organ pipe will produce middle C itself.

8. The terms used here to refer to the constituents of a tone color—*overtones, fundamental frequency, noise,* and so on—were introduced and defined in Chapter 7. The reader may wish to review this material before proceeding.



a. Sine wave
b. Sawtooth wave
c. Triangle wave
d. Square wave (a pulse wave with a duty cycle of 1:2)
e. Pulse wave with a duty cycle of 1:3
f. Reverse sawtooth wave (sounds identical to the ordinary sawtooth)

FIGURE 8.7
Common waveforms (two cycles of each), as displayed on the screen of an oscilloscope.

partial number ($1/n$, where n is the partial number). Thus, the second partial (the first overtone) is absent, the third partial (the second overtone) has one-third the amplitude of the fundamental, the fourth partial is absent, the fifth partial has one-fifth the amplitude of the fundamental, and so forth. In the absence of even-numbered partials, which reinforce odd-numbered partials at the octave (see Figure 7.1), the timbre of a square wave sounds "hollow."

A pulse wave with a duty cycle of 1:3 has every third partial absent from its spectrum, thus presenting a somewhat different timbre from that of a square wave. If the duty cycle is 1:4, then every fourth partial is missing, and so forth. Many VCOs include a dial for adjusting the duty cycle, or pulse width, of the pulse waveform so that these subtle variations of timbre can be achieved. Some VCOs even have an input jack for control voltages that can automatically vary, or modulate, the duty cycle of the pulse waveform.

The geometric origin of the names of these waveforms can be discerned when the output of a VCO is viewed on an oscilloscope (see Figure 8.7). As described in the previous chapter, however, these particular shapes appear only if all the partial frequencies in the wave are in phase with one another. If some of the partials are out of phase, the shapes will turn out somewhat differently. Nonetheless, the timbre will still be perceived as the same.

The audio signal output of a VCO is usually patched to an audio signal input on a voltage-controlled filter, voltage-controlled amplifier, mixer, or some other modifier module. The VCO module will also have at least one input for control voltages that might come from the keyboard, envelope generator, or some other source of control voltages.

Noise generator modules are typically very simple. As audio signal sources, they do not have inputs for an audio signal—they only have outputs. Separate outputs for white and pink noise may be available, or there may be a single output jack with a dial to adjust the coloration of the noise. The audio signal output of a noise generator is normally connected to the audio signal input of a voltage-controlled filter, voltage-controlled amplifier, mixer, or some other modifier module. Input jacks for control voltages are normally not provided on noise generator modules.

Some analog synthesizers include microphone preamplification modules so that a microphone signal can be boosted to the level expected by the filter and other modifier modules. There may also be input jacks for audio signals from tape recorders, mixing consoles, and other external, line-level sources. The capability to modify and control signals from such external sources greatly enriches the musical possibilities of an analog synthesizer.

SIGNAL MODIFIER MODULES

The function of a modifier module is to shape in some way the raw audio signal provided by a VCO, noise generator, or external source. As the audio signal passes through the module (on its way to its ultimate destination, the loudspeaker), the amplitude, timbre, or some other aspect of the signal will be modified or contoured in ways that result in a tone more appropriate for the musical context. This is similar to the way in which the vocal tract shapes the sound produced by the larynx, or the body of a guitar modifies the tone produced by the vibrating strings.

A **voltage-controlled amplifier (VCA)** modifies the amplitude, and therefore the intensity, of the audio signal. It has at least one input jack for an audio signal and at least one output jack, from which the signal can be connected to the audio signal input of another modifier module (or to the output lines of the synthesizer). Often there is a dial for adjusting the base amplitude level. Normally the base amplitude level of a VCA is set to zero so that the audio signal does not pass through the module until a control voltage is applied. There will always be at least one input jack on a VCA for control voltages. The most common source of control voltages for a VCA is an envelope generator (to be described soon).

Most voltage-controlled amplifiers do not actually amplify the signal—they attenuate it. If the dial for adjusting the base amplitude level is turned fully open, then the audio signal passes through at full strength. Lower settings represent the corresponding attenuation of the signal. The variations in amplitude that result from the application of a control voltage will build upon this base amplitude level, but the peak amplitude level will not likely exceed the original level of the signal at the audio input.

A **voltage-controlled filter (VCF)** can shape the timbre of a signal provided by a VCO, noise generator, or external source. "Active" filters accomplish this either by boosting the intensities of frequencies within a set

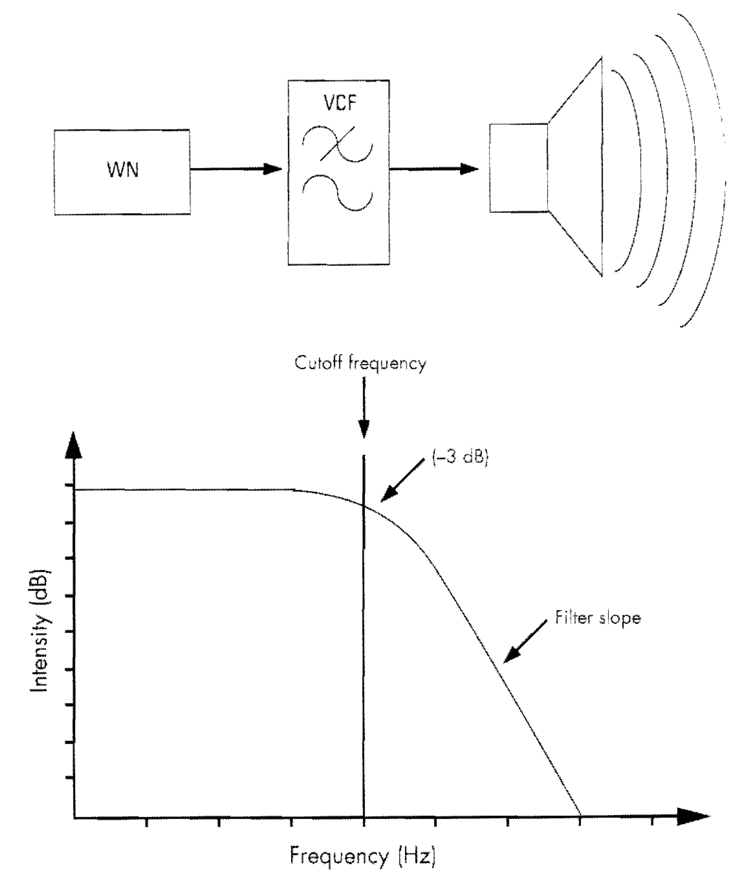


FIGURE 8.8
The effect of a low-pass filter on white noise.

range of the spectrum or by attenuating the intensities of frequencies within a range. "Passive" filters only attenuate frequencies within a range. Both types of filters can be found on analog synthesizers.

A **low-pass filter** is one that passes frequencies below a selected **cutoff frequency,** while filtering out the frequencies above the cutoff. Although the term *cutoff frequency* implies a distinct and perhaps even abrupt boundary between the range of frequencies that pass and the range of frequencies that do not, a graph of the response of a typical low-pass filter, shown in Figure 8.8, reveals that this is not necessarily the case. The cutoff frequency is simply the point at which the effects of the filter begin to be noticed. With a typical low-pass filter, the amount of attenuation increases gradually with frequencies above the cutoff frequency. This **filter slope** (or **roll-off,** as it is sometimes called) is usually expressed in decibels of attenuation per octave. For example, a low-pass filter with a roll-off of 12 dB per octave reaches the point at which attenuation is considered complete (signal strength is reduced by 60 dB) for frequencies that are five octaves above the "official" cutoff frequency. The cutoff frequency is set with a dial on most filter modules. An input jack is provided for control voltages that can vary the cutoff frequency above or below the point set manually with the dial.

The design of many low-pass filters provides for an internal path to feed some of the output of the filter back into the input. This recirculation of

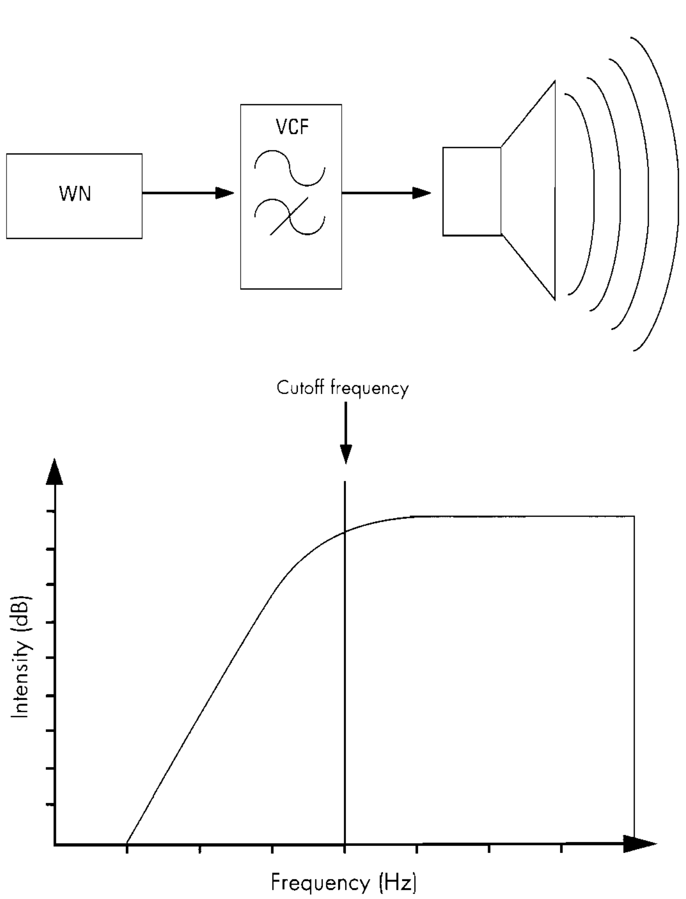


FIGURE 8.9
The effect of a high-pass filter on white noise.

part of the signal results in a resonance peak for any frequencies that happen to be near the cutoff frequency. The sharpness of this resonance peak is called the **Q** of the filter. Low-pass filters with this feedback capability have a dial that regulates the amount of the feedback and thus determines the Q of the filter. Such a dial is labeled Feedback, Q, Resonance, Emphasis, Regeneration, or something similar.

If a signal with a rich spectrum, such as a sawtooth wave, is patched into a filter tuned to a high Q, and if the cutoff frequency dial is swept across a broad range of frequencies, then each partial in the spectrum will be resonated in turn. This shimmering, even ethereal sound has become one of the classic sounds of analog synthesis. Another popular, and probably overused, technique is to patch white noise into such a resonating filter and then to sweep the cutoff frequency of the filter, thereby synthesizing the sound of a howling wind.

A filter that passes frequencies above the cutoff frequency, while filtering out the frequencies below, is called a **high-pass filter.** The effect of a high-pass filter is illustrated in Figure 8.9.

A **band-pass filter,** illustrated in Figure 8.10, removes frequencies above an upper cutoff frequency and also filters those frequencies that are below a lower cutoff frequency. The distance between the two cutoff frequencies is the width of the band of frequencies that passes (referred to as the **bandwidth**). A band-pass filter module may have separate dials to set the upper

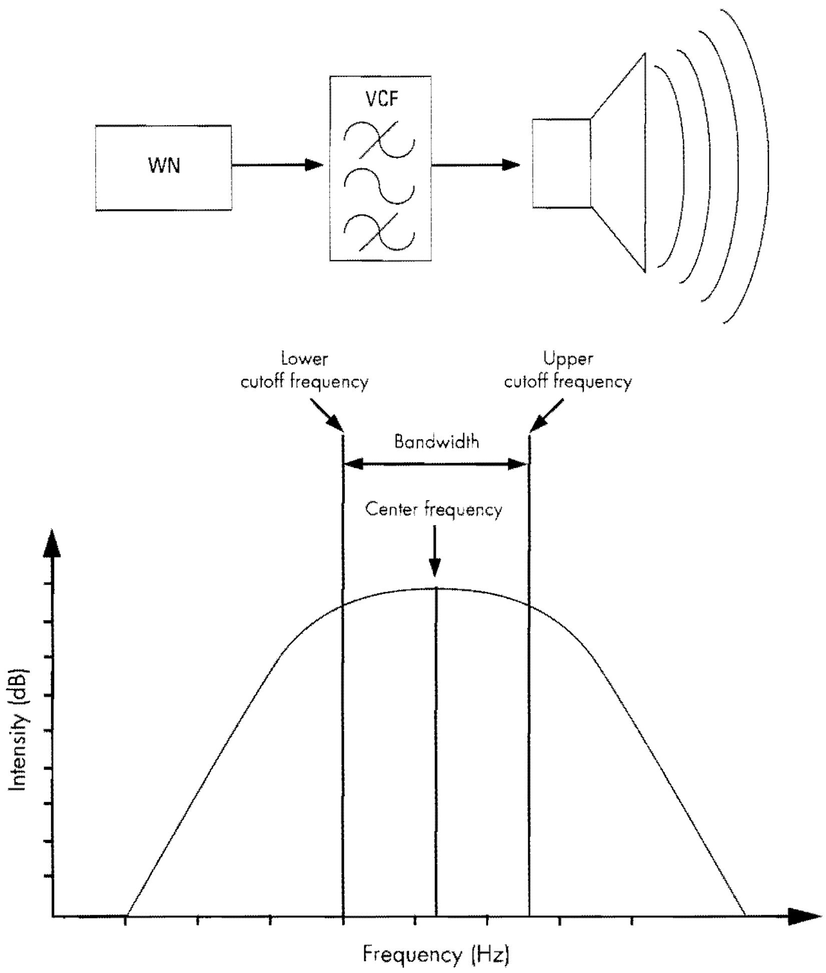
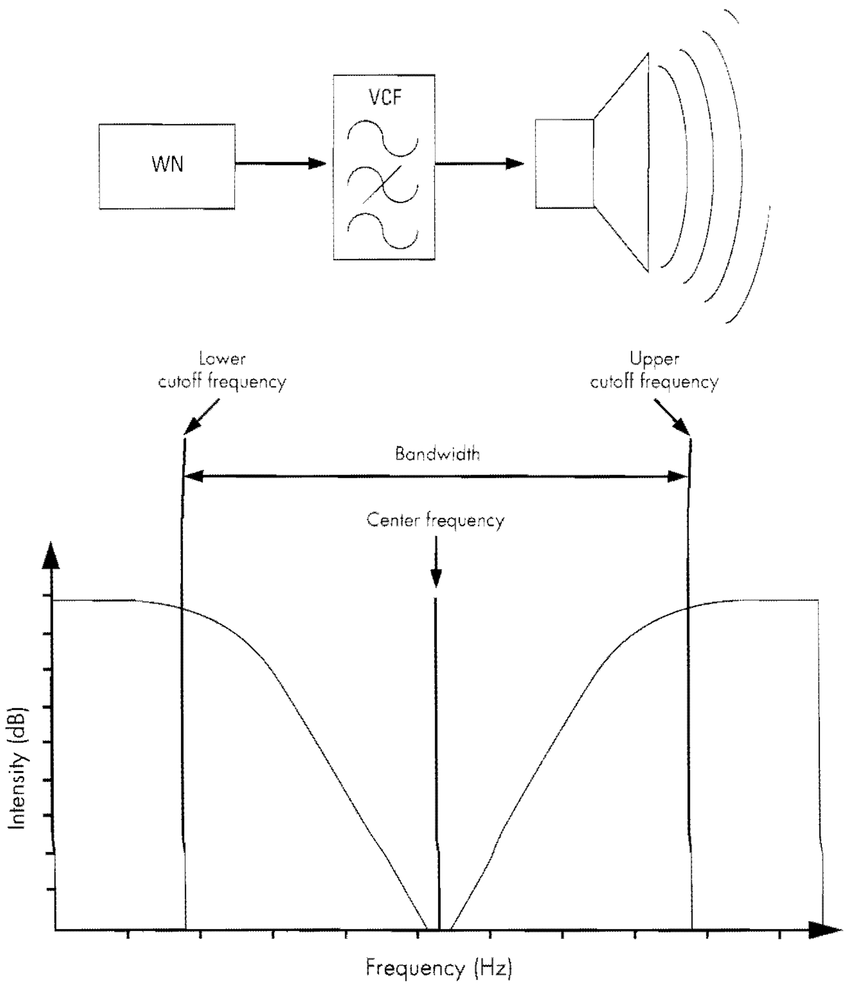


FIGURE 8.10
The effect of a band-pass filter on white noise.

and lower cutoff frequencies, or it may have a dial to set the bandwidth and another to set the center frequency of the band. An input jack for control voltages may be provided so that the center frequency can be shifted (meaning that the upper and lower cutoff frequencies are moved in tandem) in accordance with the pattern of voltage from a control voltage source, such as an envelope generator or keyboard. A control voltage input for the bandwidth of the filter may also be provided.

A **band-reject filter,** also known as a **notch filter,** is the obverse of a band-pass filter. Frequencies above the upper cutoff frequency are permitted to pass. So are frequencies below the lower cutoff frequency. The frequencies that fall between the two cutoff frequencies, however, are removed (see Figure 8.11). A band-reject filter will have a set of controls and control inputs similar to that of a band-pass filter.

A low-pass filter and a high-pass filter can be patched together to function as a band-pass or band-reject filter. If a signal is first filtered by a low-pass filter and then filtered by a high-pass filter with a lower cutoff frequency, then only a band of frequencies will remain, as illustrated in Figure 8.12. If the signal is split and one part goes through a low-pass filter and the second part goes through a high-pass filter that has a higher cutoff



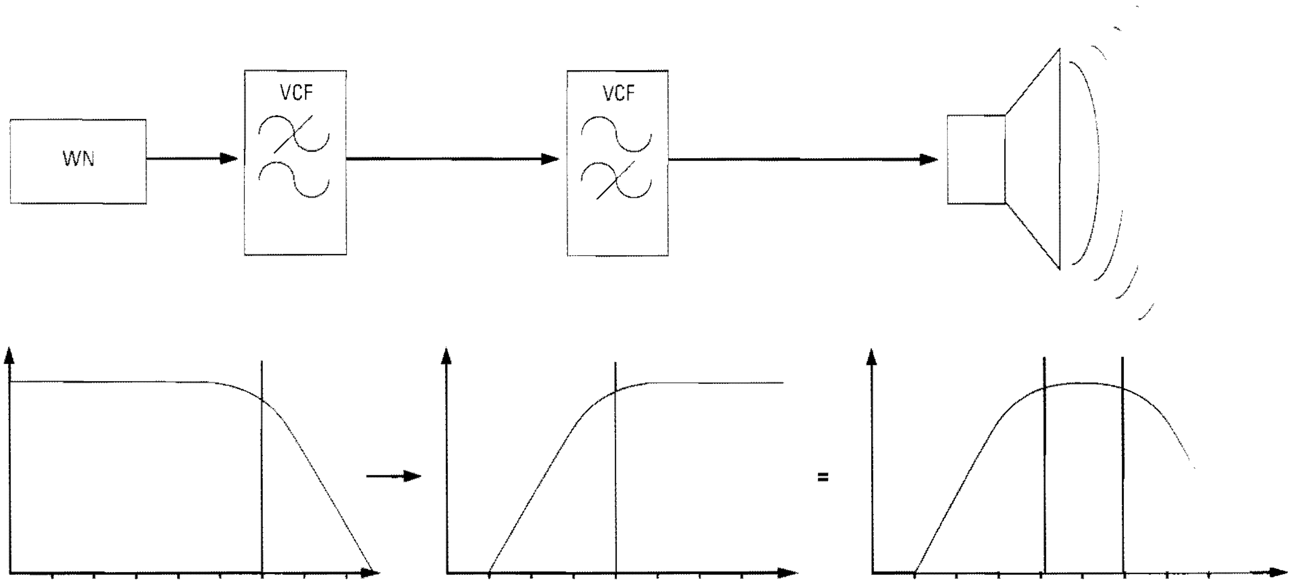


FIGURE 8.12
Creating a band-pass filter with a low-pass filter and a high-pass filter in series. The audio source signal is first sent through the low-pass filter. It is next sent through the high-pass filter. The frequencies that survive emerge as a band, as illustrated in the graph at the right.

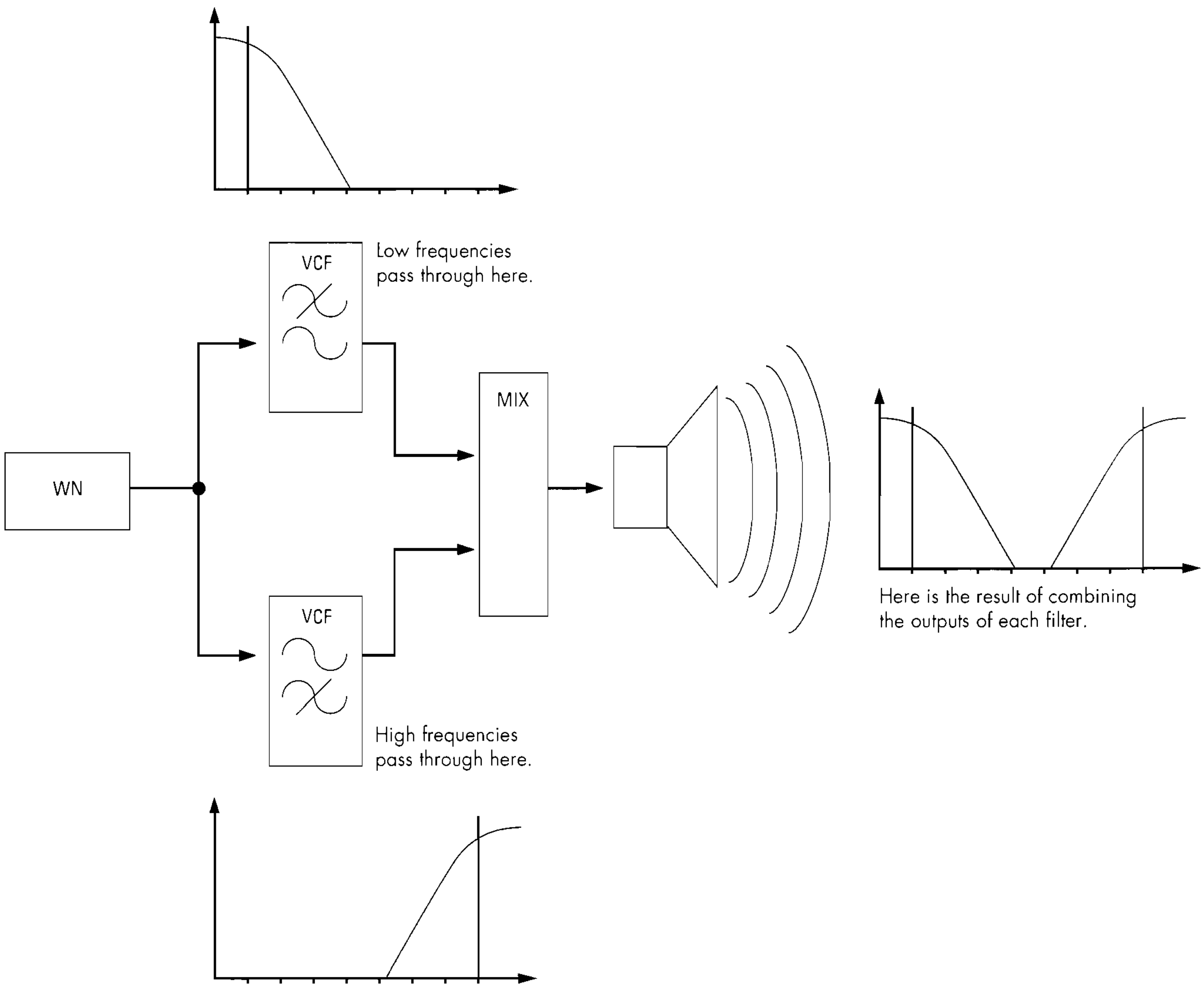


FIGURE 8.13
Creating a band-reject filter with a low-pass filter and a high-pass filter in parallel signal paths. The audio source signal is split. One part of the split is sent through the low-pass filter, and the other part of the split is sent through the high-pass filter. The signals from these two paths are then reunited by a mixer module.

frequency and if the two parts are then mixed back together, a band of frequencies will be discovered to be missing (see Figure 8.13). Because of the flexibility of techniques such as these, it is common with larger analog synthesizers to find low-pass, high-pass, band-pass, and band-reject filters conveniently located together in the same module, which might be called something like a "multi-mode filter."

A filter module can receive an audio signal from a VCO, a noise generator, or an external source, or it can receive the signal from the output of another modifier module. The output of the filter module can be patched to an audio signal input on another modifier module, or it can be connected to an output line of the synthesizer to become the signal that is monitored or recorded.

The technique of using filters to remove frequencies selectively from the standard timbres produced by VCOs and noise generators (or external sources) is called **subtractive synthesis.** It is an especially important and powerful technique of analog sound synthesis, particularly when a control voltage is used to change the effect of the filter over the duration of a sound. The synthesis of brass sounds provides an example of such dynamic filtering (in contrast to static filtering, in which the cutoff frequency is set and remains unchanged). As stated previously, an important characteristic of the timbre of a brass instrument is that the higher partials enter progressively later than the lower ones. This effect can be synthesized by patching a sawtooth wave through a low-pass filter. At the beginning of the tone the cutoff frequency is set relatively low, but it is swept up during the attack so that increasingly higher overtones can also begin to pass through the filter. Just as with a trumpet or trombone, the listener can hear the timbre brighten as the tone proceeds to develop. If the Q of the filter is high, the effects of such dynamic filtering can be especially dramatic.

ÉTUDE 8.1

Tune the frequency of a VCO to a bass pitch between two and three octaves below middle C. Select a sawtooth waveform. Patch the audio output of the VCO to the audio input of a low-pass filter. Then patch the audio output of the filter to an audio output line of the synthesizer. Connect the output of the synthesizer to an input of a multiple-track tape deck.

Adjust the Q of the filter so that it is set just below the point at which the filter oscillates on its own. Slowly sweep the cutoff frequency setting of the filter so that great arcs of overtones can be heard. Record approximately three minutes of arcs such as these on a track of the tape deck. Then record three more tracks with similar patterns of slowly unfolding arcs. If possible, monitor the four tracks with a quad speaker system (that is, with a loudspeaker in each corner of the room) for the greatest effect.

A mixer is a modifier module that simply combines audio signals from different sources. The mixer module is also used to recombine different branches of a patch (as, for example, in Figure 8.13). Typically, four audio signal inputs can be mixed to one audio signal output. A dial is provided with each input so that adjustments to the levels of each can be made. On some of the more advanced modular synthesizers, a control voltage input jack is also provided with each audio signal input so that the input level adjustments can be controlled simultaneously and independently by envelope generators or other sources of control voltage (essentially, each mixer input is functioning as an independent VCA). The output of a mixer module is generally connected to an audio signal input of another modifier module, or to an output line of the synthesizer.

Figure 8.14 shows a patch in which a mixer module is used to accomplish the **additive synthesis** of a timbre. Several independent VCOs pro-

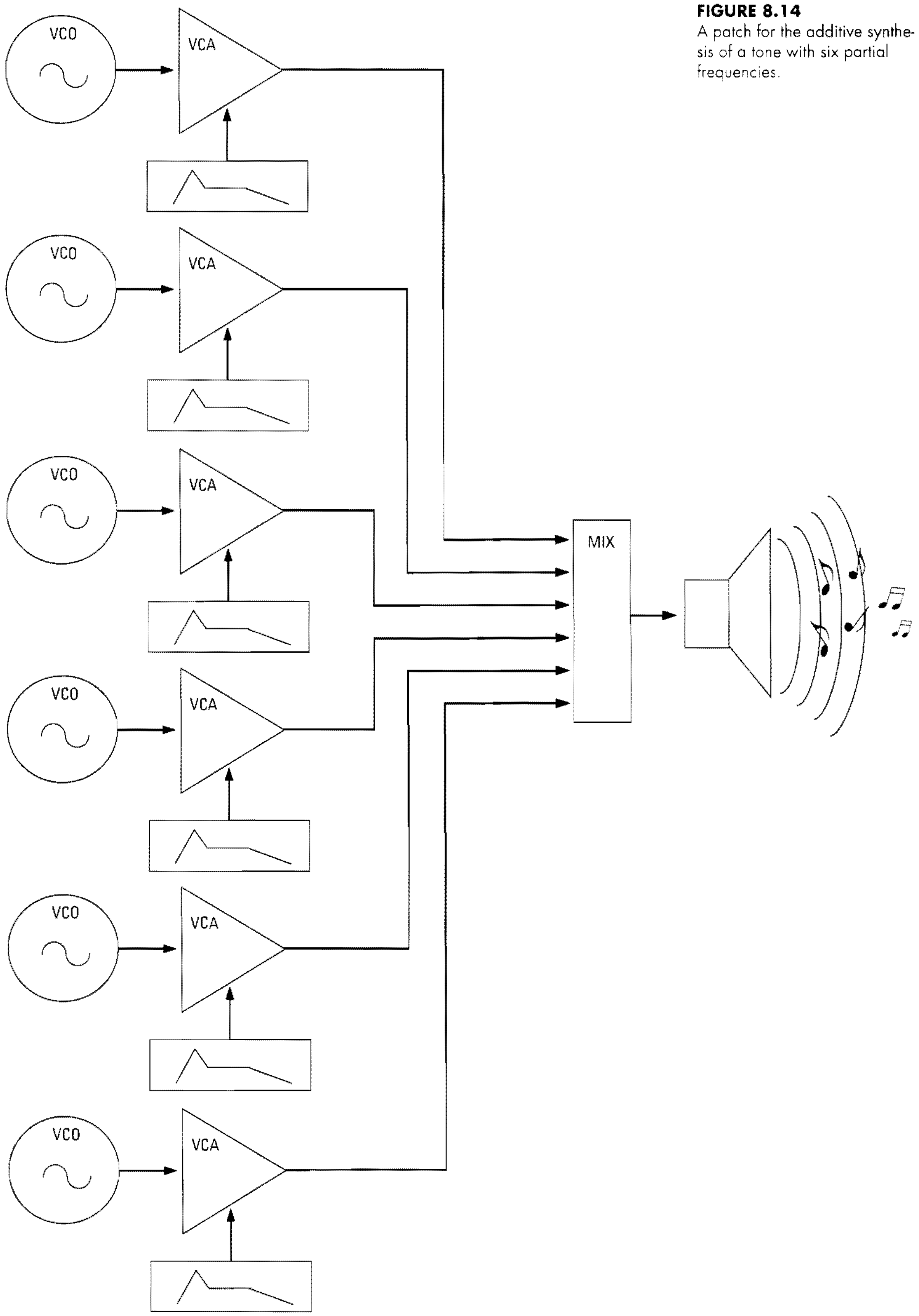


FIGURE 8.14
A patch for the additive synthesis of a tone with six partial frequencies.

vide the sine waves for the fundamental and the first few overtones. The output of each of these is first passed through a VCA controlled by an envelope generator. In this way, each partial frequency can have a unique envelope, as is the case with naturally produced timbres. The outputs of the VCAs are then connected to the audio signal inputs of the mixer (if the level of these inputs can be voltage-controlled, then the VCAs are not needed; the envelope generators can be patched to the control voltage inputs of the mixer module itself).

Unfortunately, this is not a very satisfactory way to synthesize a timbre on an analog synthesizer. First, a great number of modules are necessary, meaning that the synthesizer would have to be a large and probably expensive one. Second, it is doubtful that the VCOs can be tuned as precisely as required to simulate a gang of overtones; even if such precise tuning could be achieved, it is unlikely that the VCOs would stay in tune for very long. The successful implementation of the techniques of additive synthesis requires digital hardware, and this has become a realistic possibility for most electroacoustic musicians only relatively recently (more on this in the next chapter). For this reason, subtractive synthesis has been the main game in analog synthesis for some time.

CONTROL VOLTAGE SOURCES

The concept of voltage control is the key to the power of an analog synthesizer as a musical instrument. As stated earlier, control voltages themselves are not heard. They do not enter the audio signal path that leads to a recorder or loudspeaker. However, the effects of control voltages are most certainly heard as they are applied to the modules that generate audio signals or to the modules that subsequently modify the audio signals. Details of the structure of a sound, or of a group of sounds in a musical passage, can be shaped with some degree of precision and can be replicated fairly easily through the use of control voltages. By programming the production of sound in this way, a composer is able to work with sound in ways not readily achieved with other musical instruments.

One of the most important sources of control voltage on an analog synthesizer is a device called an **envelope generator.** This module creates a contour of voltage that is usually applied to the control input of a VCA to shape the amplitude envelope of a sound. Typically this contour has four stages: the **attack,** during which the voltage rises from zero to a peak value; the **initial decay,** during which the voltage falls off from the peak to the sustain level; the **sustain**; and the **release,** or **final decay,** during which time the voltage falls from the sustain level back to zero. Such an envelope, a very common one on analog synthesizers, is often referred to by the acronym **ADSR** (see Figure 1.8e).

An envelope generator usually includes four dials for setting the attack time, initial decay time, sustain level, and release time. Some of the more advanced systems also have input jacks for control voltages. These control

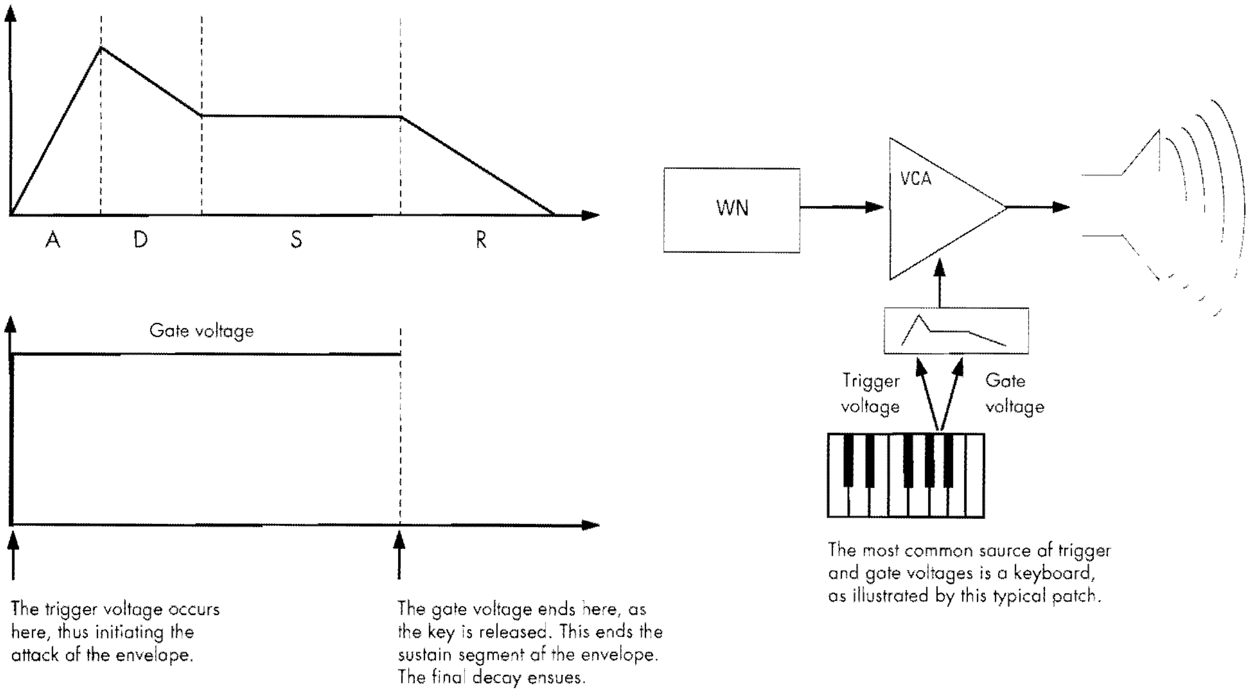


FIGURE 8.15
Trigger and gate voltages.

voltages can be used to adjust (or to override) the values set manually with each of these four dials.

A signal called a **trigger** is required for an envelope generator to commence its operation. Such a trigger may be provided by a push-button circuit on the envelope generator itself, by a keyboard when a key is depressed, or by a dedicated trigger-generator module.

Another signal, called a **gate** voltage, determines the duration of the envelope. Gate voltages are typically generated by the same mechanisms that produce triggers. Thus, for example, when a key on a keyboard is played, a trigger voltage is sent to the envelope generator. As long as the key is held down, the gate voltage is also present. The envelope proceeds through the attack and initial decay stages to the sustain level. The sustain *time* is determined by how much longer the key continues to be held down. When the key is released, the sustain time is over and the final decay gets underway (see Figure 8.15).

Although the output of an envelope generator is most commonly connected to the control voltage input of a VCA, it can just as easily be connected to the control voltage input of a VCO. This results in a sound with a pitch envelope: at first the pitch rises, then it falls back a bit, and at the end of the sound the pitch falls the rest of the way back to its initial level.

Often, an envelope generator is patched to a control voltage input on a VCF. As a result, a timbre envelope is formed as the cutoff frequency of the filter rises, falls back, holds steady for a while, then returns to its initial level. If the filter is resonant at the cutoff frequency, then a "wow" or "wa-wa" effect is heard. Unfortunately, this sort of sound is one that rather quickly became one of the clichés of analog synthesis. It is best approached with care if it is used at all.

ÉTUDE 8.2

Patch the output of a white noise generator module to the audio input of a band-pass filter. Then patch the audio output of the filter to an audio output line of the synthesizer, and connect the output of the synthesizer to an input of a two-track recorder. Tune the upper and lower cutoff frequencies of the band-pass filter so that only a narrow band of frequencies passes. Also, set the center frequency to a relatively low part of the audio range.

Next, patch the output of an envelope generator to the control voltage input for the center frequency of the filter. Patch a trigger/gate voltage source, such as the keyboard, to the trigger/gate input of the envelope generator.

Trigger an envelope, and listen to how the signal passing through the filter is changed. Experiment with different settings of the attack time, initial decay time, sustain level, and release time of the envelope. Record a passage, of approximately one or two minutes' duration, that consists of a series of such enveloped noise patterns. Then, on the other track of the tape deck, record a similar series as a counterpoint to the patterns recorded on the first track.

Another control voltage source on an analog synthesizer is the keyboard. It provides a discrete, sustained voltage whenever a key is pressed (along with trigger and gate voltages that may be of use to an envelope generator). A quite ordinary application of the keyboard control voltage is to patch it to the control voltage input of a VCO. The VCO frequency is thus set according to which key has been pressed. This patch, illustrated in Figure 8.16, brings an analog synthesizer perilously close to being not much more than a mutant electronic organ. In fact, many analog synthesizers, such as those by Buchla, have been designed to avert this situation by providing only a rudimentary keyboard, or none at all. A great variety of modules have been developed as alternatives to the keyboard for the generation of control voltages. If the habits of thought associated with keyboard playing hinder the exploration of some of these other possibilities, and thereby limit the range of expression of the medium, then perhaps the keyboard is best avoided. Or, if it cannot be avoided, then at least it should be approached with a certain amount of circumspection.

The keyboards of early analog synthesizers were **monophonic**—capable of producing only one control voltage at a time (apart from the trigger and gate voltages). This in itself is not a particular disadvantage, since many of the world's great musical instruments—clarinets, saxophones, trumpets, and kazoos—are also monophonic. On a monophonic keyboard, some sort of priority scheme must be engaged to determine which control voltage is produced when more than one key is pressed. Usually the lowest key is given priority, although alternative systems assign priority to the highest note, the first note, or the last note played. Performers who had developed their keyboard skills on pianos and organs were rather unnerved by this,

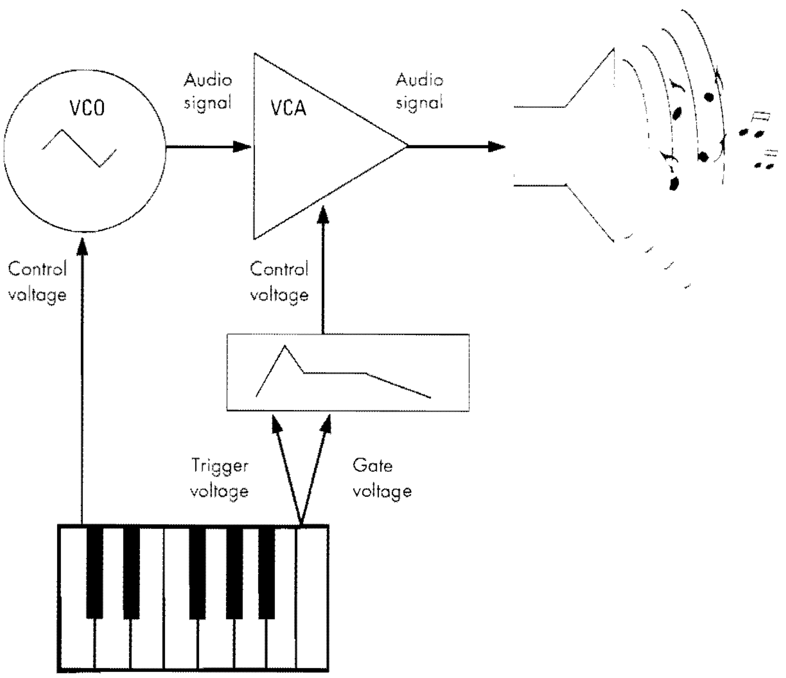


FIGURE 8.16
A very ordinary patch. The keyboard generates control voltages that determine the frequency produced by the VCO. The keyboard also generates trigger and gate voltages for the envelope generator. The envelope generator produces a control voltage that determines the gain level of the VCA. The amplitude of the audio signal produced by the VCO is shaped by this VCA. The signal is then passed to the external output line and to the studio monitor system.

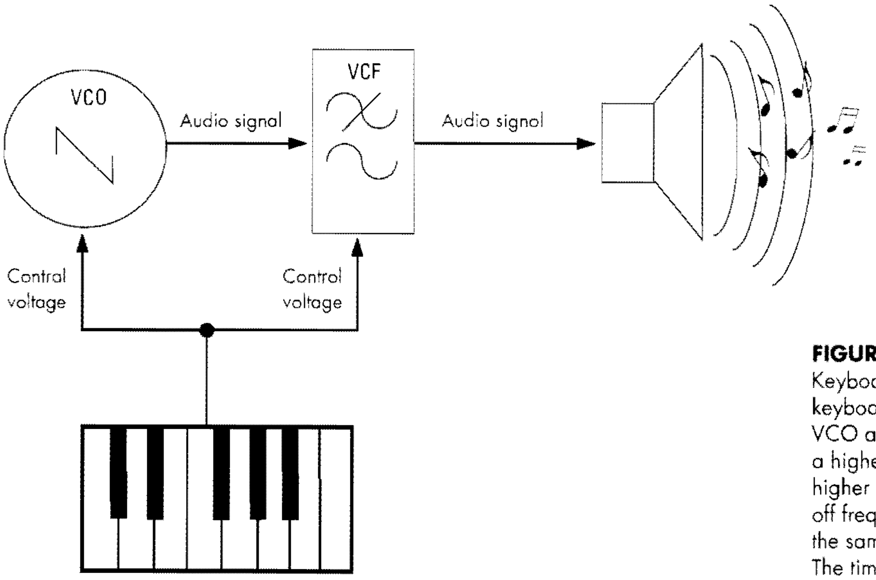


FIGURE 8.17
Keyboard tracking. Here the control voltage from the keyboard is used to determine both the frequency of the VCO and the cutoff frequency of the low-pass filter. Thus, a higher control voltage from the keyboard will cause a higher pitch to be produced, and will also cause the cutoff frequency of the filter to move to a higher level so that the same number of overtones will pass through the filter. The timbre can therefore remain fairly constant.

however, and pressed for the development of keyboards that were **polyphonic.** These could provide up to eight independent voltages that could be patched to several different VCOs or other modules.

A control voltage output from a keyboard can be patched into the control voltage input of a VCF so that the cutoff frequency of the filter can be moved in tandem with the pitch changes of the VCO, as illustrated in Figure 8.17. This **keyboard tracking** by the filter ensures that the waveform remains unchanged, regardless of the frequency of the sound. If the

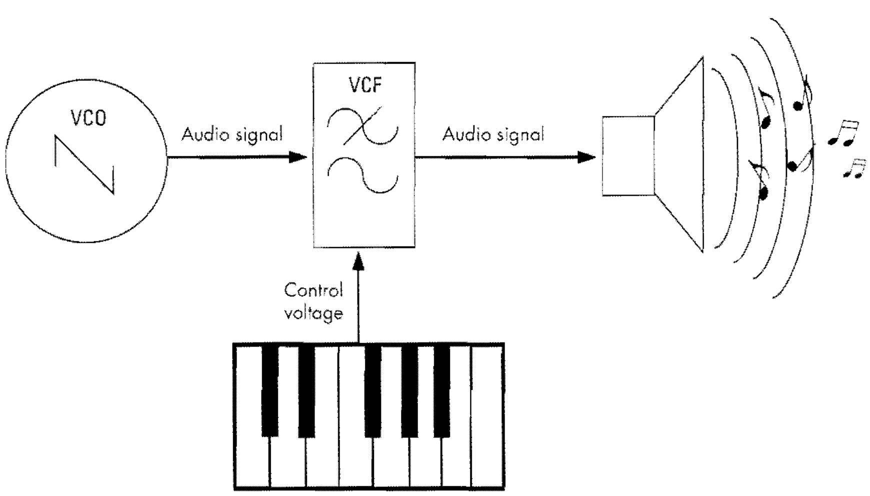


FIGURE 8.18
A synthetic jaw harp. The control voltage from the keyboard is connected only to the low-pass filter. Thus, the cutoff frequency of the filter rises and falls to discrete levels, determined by the key that is being played, while the frequency of the sound remains constant. For best results, a bright waveform, rich in overtones, should be used. Also, it helps if the resonance setting of the VCF is turned up to at least a moderate level.

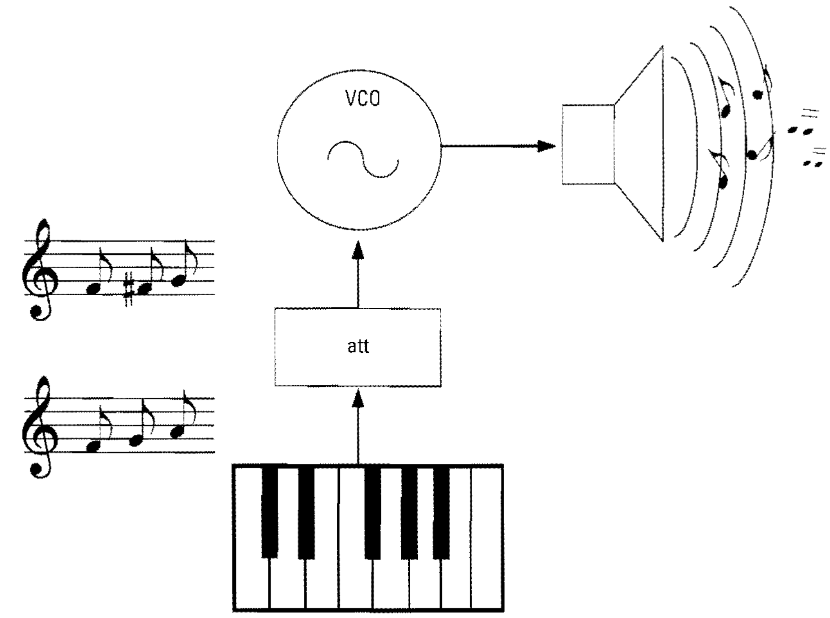
cutoff frequency of the filter were to remain stationary, then many of the overtones of the higher pitches would fall above the cutoff frequency and would be filtered out. This would result in a more rounded waveform with a more mellow, perhaps somewhat dull sound.

A more interesting sound, perhaps, is produced by a patch in which the control voltage of the keyboard is connected to the filter, but is *not* connected to the VCO (see Figure 8.18). The pitch of the oscillator remains constant, but whenever a different key is pressed, the timbre of the sound changes as the cutoff frequency of the filter is set to a different level by the control voltage. The sound that results is much like that of a jaw harp or of the Australian aboriginal instrument, the didjeridu.

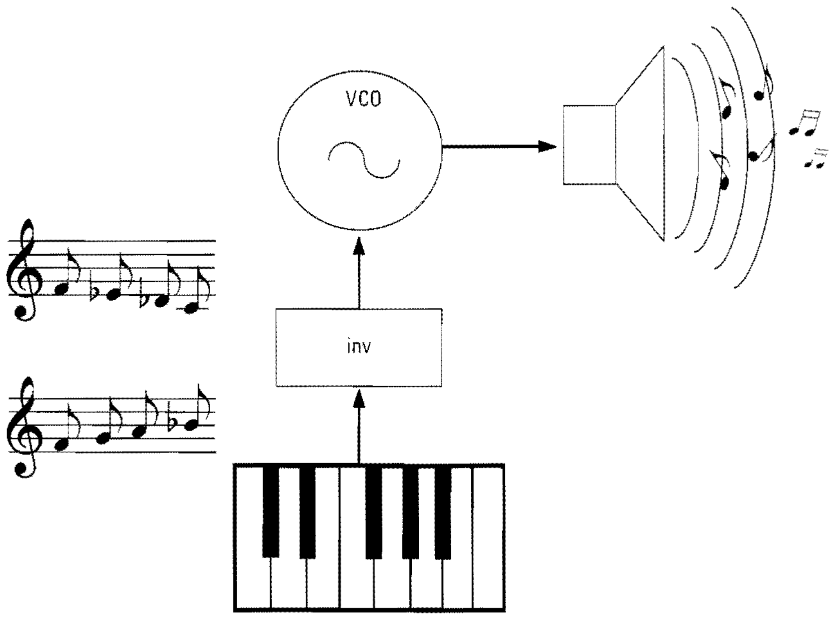
Control voltages can themselves be modified, usually by modules such as **control voltage attenuators** or **control voltage inverters** (often the functions of these two are combined in a single module). For example, if the control voltage output of a keyboard is passed through an attenuator with its dial set at midpoint, then the keyboard voltages will be halved (see Figure 8.19a). If these reduced voltages are patched to a VCO, then the intervals of pitch will also be halved. A major second (a "whole step") played on the keyboard will thus sound a minor second (a "half step"); a minor second played on the keyboard will sound a **quarter tone.** This creates an opportunity to explore systems of tuning other than the currently extant system of 12 equal steps per octave.

If the control voltage of a keyboard is passed through an inverter, and the output of the inverter is patched to the control voltage input of a VCO (as in Figure 8.19b), then the higher voltages produced by playing higher keys will be inverted to lower voltages, and lower pitches will be sounded. Likewise, if lower keys are played, the lower voltages that are thus generated will be inverted to higher voltages, and the VCO will respond by producing higher pitches. Ascending scales played on the keyboard will be heard as descending scales, and vice versa.

FIGURE 8.19
Two common techniques of control voltage modification.



a. With a control voltage attenuator inserted between the keyboard and the VCO, the voltage generated by the keyboard is halved, and the intervals of pitch produced by the VCO are halved as well—in this example, from whole steps to half steps.

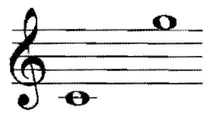


b. A control voltage inverter transforms higher voltages to lower voltages, and vice versa. In this example, playing an ascending series of keys on the keyboard produces a descending series of pitches from the VCO.

ÉTUDE 8.3

Patch the voltage output of the keyboard to the input of a control voltage attenuator. Then patch the output of the control voltage attenuator to the control voltage input of a VCO. Patch the audio output of the VCO to the audio input of a VCA, and patch the audio output of the VCA to an output line of the synthesizer. (You may also want to patch the output of an envelope generator, perhaps triggered by the keyboard, to the control voltage input of the VCA, so that each tone is shaped with an amplitude envelope.)

Alternately play these two keys on the keyboard, and adjust the level of the control voltage attenuator until the interval of pitch heard is a perfect octave:



You have now tuned your synthesizer to a 19-tone-per-octave scale. This particular tuning is noted for the purity of its minor thirds and major sixths, which come very close to their ideal ratios of 6:5 and 5:3, respectively (you may want to review the description of interval ratios provided in Chapter 1). Experiment with other tunings by adjusting the control voltage attenuator level. (Be aware, however, that the stability of pitch from an analog oscillator is usually not too great; the oscillator is likely to drift out of tune within a few minutes.)

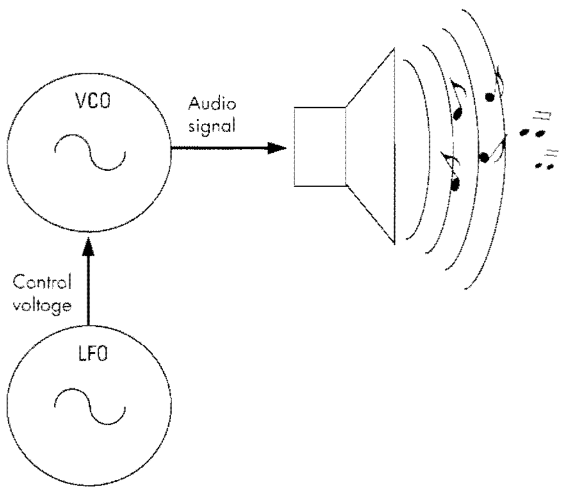
A **low-frequency oscillator (LFO)** is one with a frequency that is usually below the range of human hearing. Since it cannot be heard, its only use is to function as a control voltage source. When the output of an LFO is patched to the control voltage input of a VCO (see Figure 8.20a), the audio frequency of the VCO will begin to fluctuate up and down at the rate of the frequency of the LFO. The extent to which the audio frequency is raised and lowered is determined by the amplitude of the signal from the LFO. This amplitude can be regulated if a control voltage attenuator is patched in between the LFO and the VCO (see Figure 8.20b).

The periodic fluctuation of an audio frequency at a rate less than 20 times per second (so that the ear can follow the individual changes of frequency) is called a vibrato. The rate of the vibrato is the same as the frequency of the LFO. The width of the vibrato is determined by the amplitude of the LFO signal.

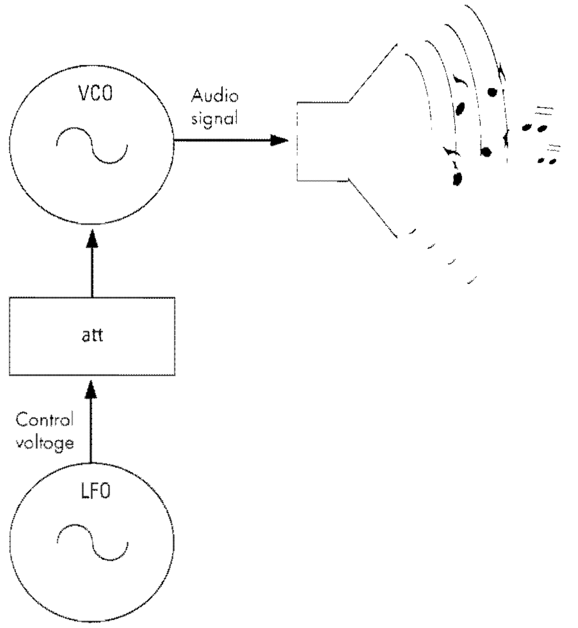
If the output of an LFO is patched to the control input of a VCA (as illustrated in Figure 8.21), then a similar modulation of amplitude, called tremolo, is produced. A "wa-wa" effect is produced if the cutoff frequency of a filter is controlled by an LFO (as depicted in Figure 8.22).

Such subtle modulations of pitch, amplitude, or timbre are often applied to longer tones, much as a trained singer, violinist, or woodwind player shapes long tones. For example, the tone begins without vibrato, but will begin to vibrate after it is about halfway through its duration. This gives

FIGURE 8.20
The use of a low-frequency oscillator (LFO) to create a vibrato in the audio signal.



a. A simple patch



b. With a control voltage attenuator used to regulate the width of the vibrato

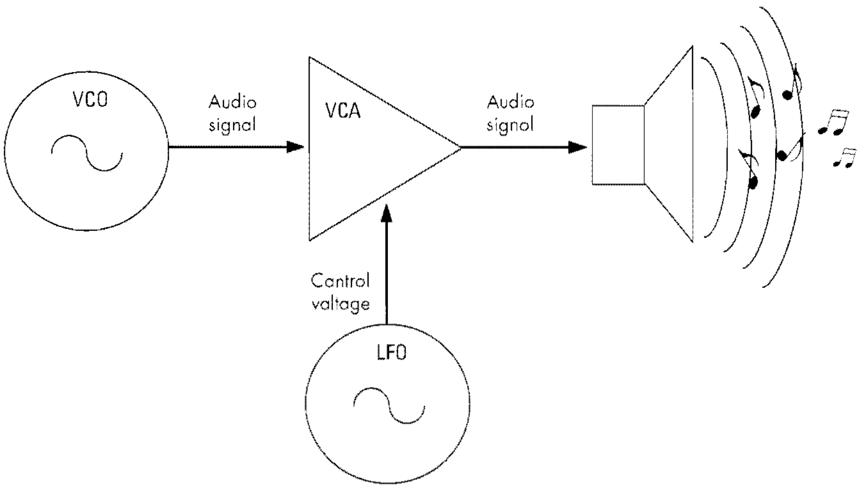


FIGURE 8.21
The use of a low-frequency oscillator (LFO) to create a tremolo in the audio signal.

direction to the tone by drawing the attention of the listener forward into the next sound. The result is a warmer, more expressive, and more lifelike sound. To facilitate this technique, many LFOs have a dial for setting a delay time for the onset of the low-frequency oscillation that controls the modulation.

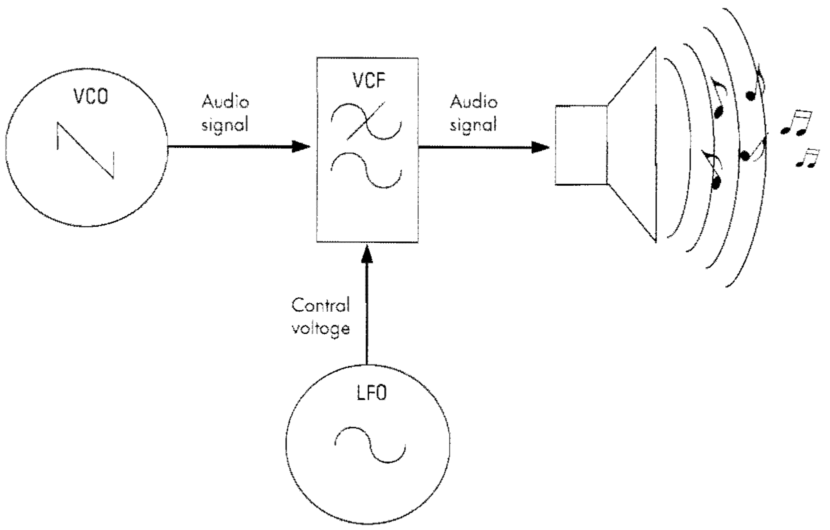


FIGURE 8.22
The use of a low-frequency oscillator (LFO) to create a "wa-wa" effect. The rising and falling voltage produced by the LFO causes the cutoff frequency of the filter to rise and fall accordingly, thus producing a slow undulation of the timbre of the signal that originated in the oscillator.

audio output line of the synthesizer. Connect the output of the synthesizer to an input of a two-channel tape deck. Adjust the Q of the filter so that it is set just below the point at which the filter oscillates on its own.

Patch the control voltage output of the keyboard to the control voltage input of the VCF. Play a random assortment of keys in a markedly rhythmic pattern. Record approximately two minutes of this synthetic didjeridu.

Now remove all the patch cords and start a new patch. Tune the frequency of a VCO to a pitch in the octave above middle C. Select a sine waveform. Patch the audio output of the VCO to the audio input of a VCA. Patch the audio output of the VCA to an output line of the synthesizer, and connect the output of the synthesizer to the input of the other track of the tape recorder.

Next, patch the output of an envelope generator (to be triggered by the keyboard) to the control voltage input of the VCA. Set a long decay time for the envelope.

Patch the output of an LFO to the input of a control voltage attenuator (if available), and patch the output of the control voltage attenuator to the control voltage input of the VCO. Select a square waveform on the LFO, and set the frequency of the LFO in the range of 4–10 Hz. Adjust the level of the control voltage attenuator so that the audio frequency produced by the VCO is alternated no more than a major third or perfect fourth by the control voltage originating from the LFO. Also, if possible, try to delay the arrival to the VCO of the control voltage from the LFO to coincide with the beginning of the final decay of the envelope of the tone.

Record a series of a few of these tones on the second track of the tape recorder. Perhaps, for variety, reset the frequency of the LFO for each tone recorded on this track.

An audio frequency oscillator, a VCO in fact, can also be used as a control voltage source, and this can produce some very unusual effects. If an oscillating voltage with a frequency greater than 30 Hz is patched to the

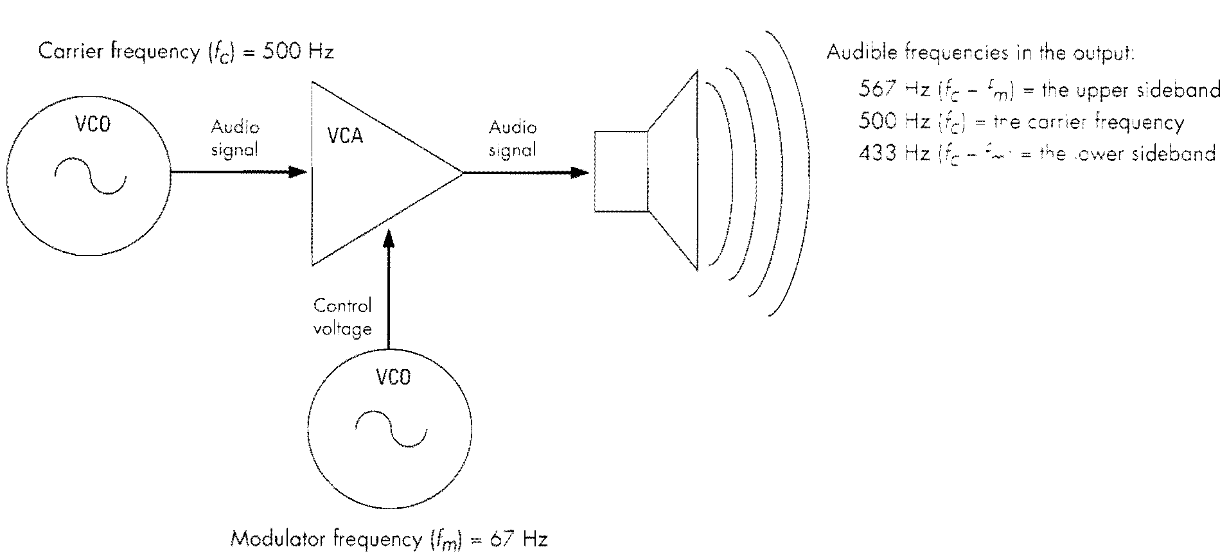


FIGURE 8.23
The use of an audio frequency oscillator as a control voltage source to modulate the amplitude of the audio signal produced by the first VCO. As a result of the rapid fluctuations of the amplitude of the audio signal, additional frequencies called sidebands become audible.

control voltage input of a VCA that is passing an audio signal, then additional frequencies, called **sidebands,** will be heard around the frequencies in the audio signal. For each of these frequencies in the audio signal, there will be an upper sideband with a frequency equal to the sum of the audio frequency, called the **carrier** (f_c), and the controlling frequency, called the **modulator** (f_m). There will also be a lower sideband, with a frequency equal to the difference between the carrier and the modulator frequencies (f_c–f_m). This technique of producing sidebands by causing the amplitude of a signal to fluctuate periodically at a rate greater than 30 times per second is called **amplitude modulation,** or **AM** (in contrast to tremolo, the fluctuation of amplitude at a rate less than the lower limit of the range of audible frequencies—approximately 20 Hz).

In Figure 8.23, a sine wave with a frequency of 500 Hz is produced by the VCO and patched to the audio input of the VCA. A sine wave with a frequency of 67 Hz is produced by another VCO and patched to the control voltage input of the VCA. One audio frequency goes into the VCA, but three audio frequencies are present in the audio signal output: 433 Hz, the lower sideband; 500 Hz, the carrier frequency; and 567 Hz, the upper sideband.

If the carrier waveform is a sawtooth wave, as in Figure 8.24, or any other complex waveform, then sidebands around the overtones also appear in the output. If the modulator waveform is a sawtooth, as in Figure 8.25, then each overtone in the modulator will also act as a modulator and produce sidebands in the output. If carrier and modulator frequencies do not have an obvious, common arithmetic factor (for example, a sawtooth wave with a frequency of 100 Hz for the carrier and a sine wave with a frequency

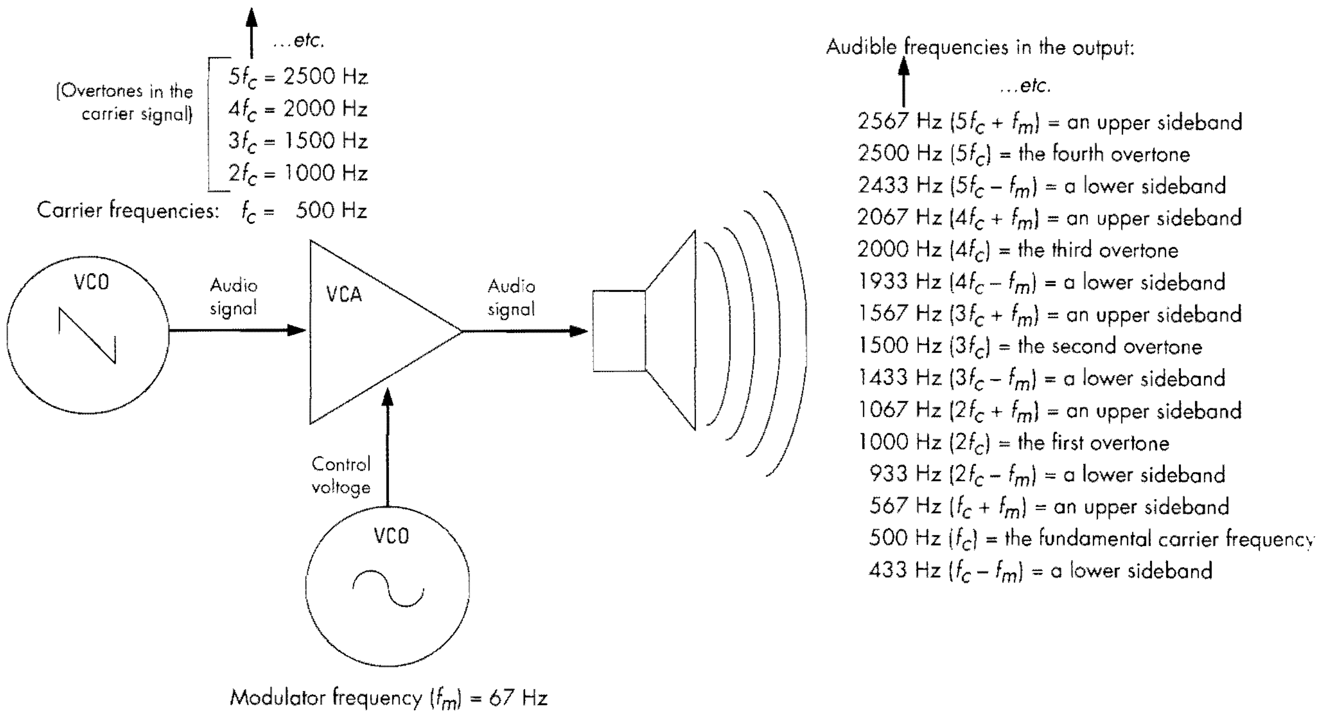


FIGURE 8.24
The use of a sawtooth wave as a carrier signal results in a more complex output signal, as sidebands are produced around each overtone of the sawtooth.

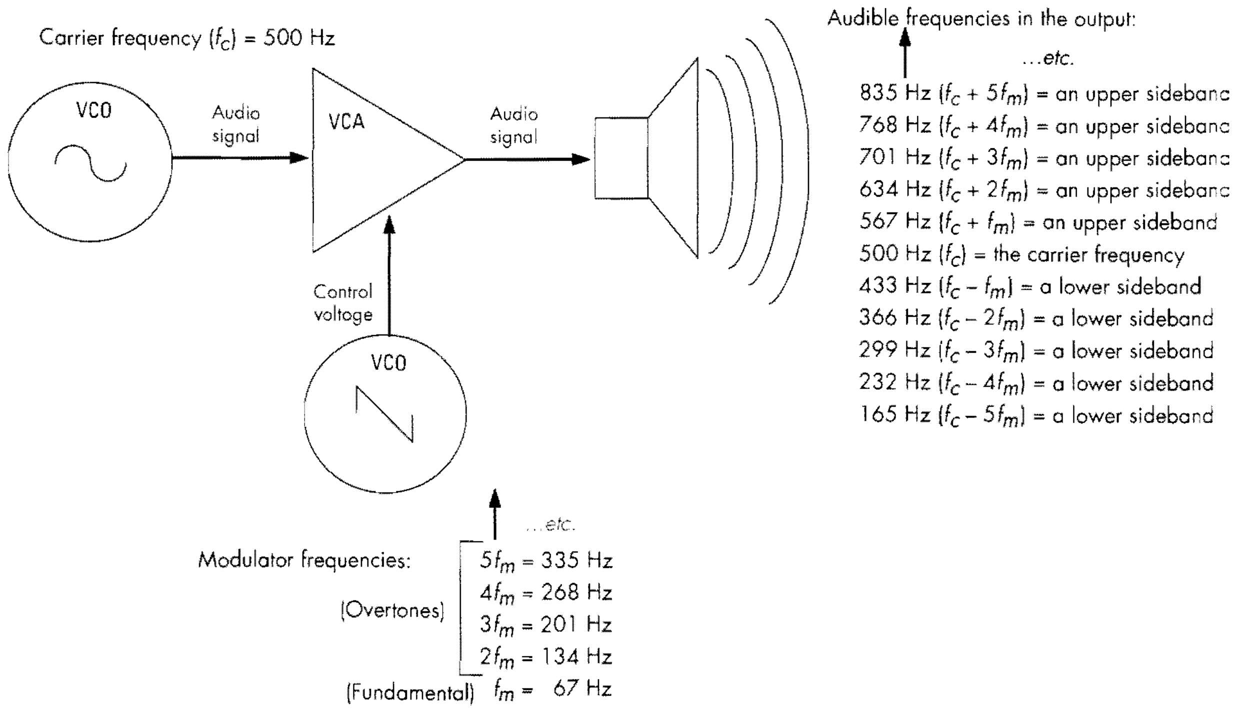


FIGURE 8.25
The use of a sawtooth wave as a modulating signal also results in a more complex output signal, as each overtone of the sawtooth creates sidebands around the carrier frequency.

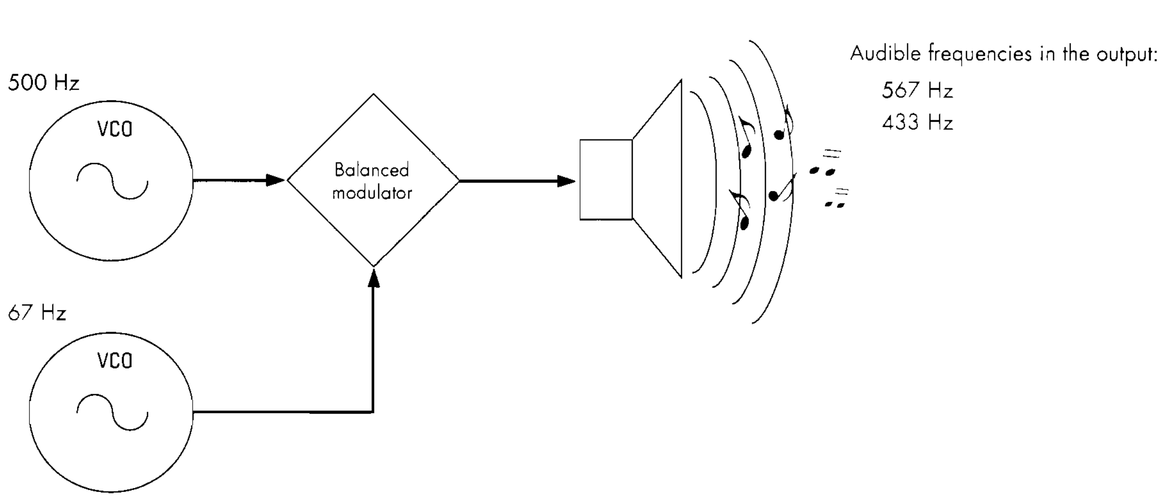


FIGURE 8.26
A balanced modulator. Note that neither of the input frequencies is audible in the output. The two frequencies that are heard at the output are the sum of the input frequencies and the difference between the input frequencies.

of 43 Hz for the modulator), then fairly complex, inharmonic spectra, like those of bells, can be created with relative ease.

A **balanced modulator,** also called a **ring modulator,** is a modifier module found on many larger analog synthesizers. Its operation is based upon a special application of the technique of amplitude modulation. The module has two inputs and one output. The signal that is available at the output consists exclusively of the upper sideband and lower sideband frequencies produced by the modulation of the amplitude of one of the input signals by the other input signal (see Figure 8.26). The carrier frequencies themselves are blocked and so are not present in the output. The effect of a balanced modulator module is particularly striking when one of the input signals is patched from an external source—a microphone that is picking up a speaking voice or a tape recorder playing prerecorded music. The radical transformations performed by a balanced modulator can even do justice to some such sources, such as political speeches or dull classroom lectures.

Frequency modulation, or **FM,** is the technique of producing sideband frequencies by modulating the frequency of an audio signal at a rate greater than 30 times per second (a sort of super-vibrato). Figure 8.27 illustrates a patch for frequency modulation. As with the production of a vibrato (illustrated in Figure 8.20a), the output of a controlling oscillator is connected to the control voltage input of the audio frequency oscillator. The difference with FM is that the frequency of the controlling oscillator is greater than the lower limit of the range of audible frequencies. We are unable to hear the individual fluctuations of frequency; instead, we hear sidebands.

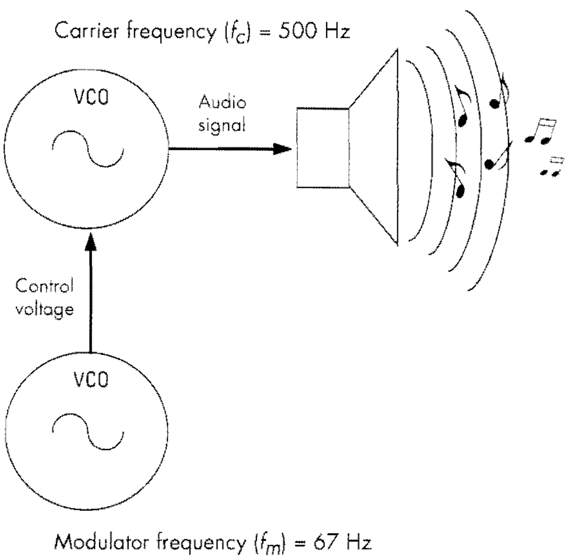


FIGURE 8.27
The use of an audio frequency oscillator as a control voltage source to modulate the frequency of the audio signal. This is frequency modulation, or FM.

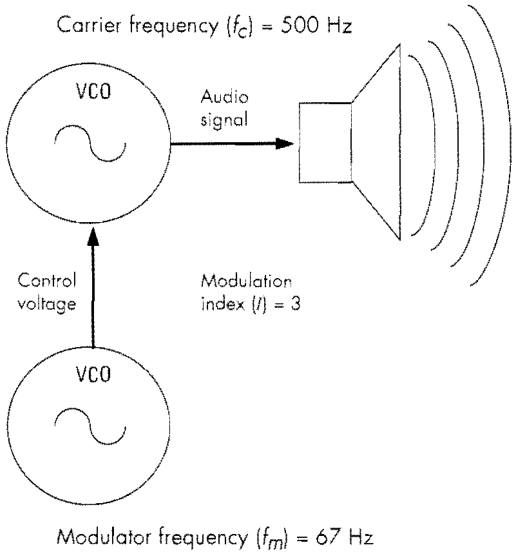
FURTHER DETAILS OF FREQUENCY MODULATION

With AM there is only one upper sideband and one lower sideband per carrier frequency, but with FM there can be many more upper and lower sidebands per carrier. The approximate number of upper sidebands (and also lower sidebands) is found by first dividing the range of the deviation of the carrier frequency (Δf_c) by the frequency of the modulator (f_m). This value ($\Delta f_c/f_m$) is called the **modulation index** (I). Generally, the number of upper sidebands (and the number of lower sidebands) is two more than the value of the modulation index.

If the amplitude of the modulator is increased, then the amount of change in the frequency of the audio oscillator (that is, the deviation of the carrier frequency) is also increased. The modulation index in this case has a greater value, indicating that more sidebands are likely to be audible. If the *frequency* of the modulator is increased (while its amplitude remains the same), then the modulation index has a lower value, signifying that fewer sidebands will be heard.

Figure 8.28 illustrates a collection of sidebands created by the modulation of a 500-Hz audio signal by a 67-Hz control signal. The modulation index in this example is 3, meaning that one may expect to hear five upper sidebands and five lower sidebands. The waveform produced by both oscillators is a sine. If the carrier has a more complex waveform, then five orders of sidebands will also be created on both sides of each overtone (an even more complex result than that illustrated in Figure 8.24 with the amplitude modulation of a sawtooth). If the *modulator* has a more complex waveform, then each of its overtones will create five sidebands on each side of the carrier (likewise an even more complex result than that illustrated in Figure 8.25 by the simple amplitude modulation of a sine wave by a sawtooth wave).

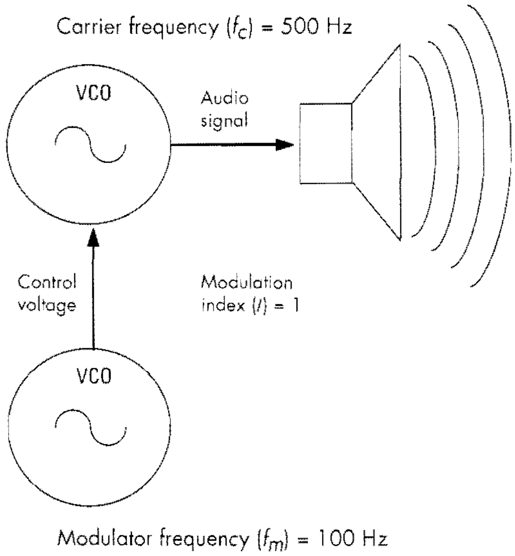
If the carrier and modulator frequencies are related by a simple ratio, then a constellation of sidebands is created that looks suspiciously similar to an overtone series (see Figure 8.29). Thus, the result of the additive synthesis of a timbre using several VCOs (as in Figure 8.14)



Audible frequencies in the output:

835 Hz ($f_c + 5f_m$) = the 5th-order upper sideband
768 Hz ($f_c + 4f_m$) = the 4th-order upper sideband
701 Hz ($f_c + 3f_m$) = the 3rd-order upper sideband
634 Hz ($f_c + 2f_m$) = the 2nd-order upper sideband
567 Hz ($f_c + f_m$) = the 1st-order upper sideband
500 Hz (f_c) = the carrier frequency
433 Hz ($f_c - f_m$) = the 1st-order lower sideband
366 Hz ($f_c - 2f_m$) = the 2nd-order lower sideband
299 Hz ($f_c - 3f_m$) = the 3rd-order lower sideband
232 Hz ($f_c - 4f_m$) = the 4th-order lower sideband
165 Hz ($f_c - 5f_m$) = the 5th-order lower sideband

FIGURE 8.28
The sidebands that can be created from just two sine waves by using frequency modulation techniques. The value of the modulation index in this example is 3. This means that there can be a total of five upper sidebands and five lower sidebands.



Audible frequencies in the output:

800 Hz ($f_c + 3f_m$) = the 3rd-order upper sideband
700 Hz ($f_c + 2f_m$) = the 2nd-order upper sideband
600 Hz ($f_c + f_m$) = the 1st-order upper sideband
500 Hz (f_c) = the carrier frequency
400 Hz ($f_c - f_m$) = the 1st-order lower sideband
300 Hz ($f_c - 2f_m$) = the 2nd-order lower sideband
200 Hz ($f_c - 3f_m$) = the 3rd-order lower sideband
100 Hz = implied fundamental

Note: If the modulation index is increased to a value of 2, then the "fundamental" frequency of 100 Hz will actually be present in the signal, as the 4th-order lower sideband.

FIGURE 8.29
A harmonic series of sidebands can be created if the carrier frequency and the modulating frequency are related by a simple ratio.

can be approximated fairly closely with the use of only two oscillators—one to produce a carrier frequency, and one to produce a modulator frequency. Because of the instability of the tuning of analog oscillators, this is not an easy technique to implement. However, FM synthesis has become an extremely important technique of digital sound synthesis (and will be described more completely in the next chapter).

ÉTUDE 8.5

Tune the frequency of a VCO to a pitch in the octave above middle C. Select a sine waveform. Patch the output of the VCO to the audio input of a VCA. Set the dial for base amplitude on the VCA to approximately the midpoint. Patch the audio output of the VCA to an output line of the synthesizer.

Tune the frequency of a second VCO to a pitch an octave below that of the first VCO, and select a sine waveform. Patch the output of this second VCO to the audio input of a second VCA. Set the base amplitude of this second VCA to zero. Patch the output of an envelope generator to the control voltage input of the second VCA. Patch a trigger source to the envelope generator, and set the envelope for long attack and decay times.

Patch the output of the second VCA to the control voltage input of the first VCO. The sustained tone being produced by the first VCO will then blossom into many colors whenever an envelope is applied to the second VCA. The passage of the signal through the second VCA permits modulation of the frequency of the first VCO by the frequency of the signal originating from the second VCO.

GUIDE FOR LISTENING
THE WILD BULL, BY MORTON SUBOTNICK

Among the most highly regarded compositions realized on an analog synthesizer system are those of Morton Subotnick. Born in Los Angeles in 1933, Subotnick joined Pauline Oliveros, Terry Riley, Ramon Sender, and others around 1962 in establishing the San Francisco Tape Music Center. By 1965 Don Buchla, an engineer, had become involved at the center. In particular, he collaborated with composers Subotnick and Sender in the development of a solid-state, modular, analog synthesis system. This was subsequently introduced commercially, in 1966, as the Buchla Modular Electronic Music System.[a]

Unlike the Moog synthesizer and other instruments that were introduced contemporaneously, the Buchla system did not include a traditional keyboard as a source of control voltages. Instead, it had a set of 16 touch-sensitive plates that could be used primarily to generate trigger voltages. However, the most important source of control voltages in the system was the control voltage sequencer, the first of its kind to be included on a synthesizer. On receiving a trigger from one of the touch-sensitive plates, or a similar source, the sequencer could begin to generate a repeated pattern of events, or with more sophisticated programming and occasional manual control, it could create a complex and richly varied series of events. Sounds triggered by the sequencer could be shaped with a variety of pitch contours, amplitude envelopes, and filter contours through the use of several envelope generators. As an instrument without a keyboard, the

[a] Buchla continues to develop marvelous instrument systems—most recently, a powerful alternate MIDI controller called Lightning™.

Buchla system encouraged composers to explore more boldly and to rely less on familiar patterns and techniques. It was indeed a terrific "toybox" for composers.

Shortly after the introduction of the Buchla synthesizer, a classical-music record label, Nonesuch Records, commissioned Subotnick to compose a piece for the instrument. This composition, entitled *Silver Apples of the Moon,* was released in 1967 (Nonesuch H-71174) and was expressly intended for the record medium—a piece to be heard on a stereo system at school or at home rather than for concert presentation or broadcast. *Silver Apples* was a great success, and a second piece was subsequently commissioned by Nonesuch and released in 1968 (Nonesuch H-71208). This piece was *The Wild Bull,* with a title inspired by a Sumerian poem from about 1700 B.C. Like the earlier Nonesuch release, this work is a masterful excursion through rich and exotic worlds of sound evoked by the Buchla instrument.

Side 1 begins with lengthy sounds of descending pitch and inharmonic spectrum that pan gently from one side of the stereo field to the other. The texture is very sparse here, with much silence. Unlike the compressed compositions of Stockhausen and Babbitt (described in Chapter 10), the compositions of Subotnick convey a much more expansive sense of time: they unfold rather gradually, and the total duration of this piece is about 28 minutes.

At around 1:29, the first sequenced pattern makes a clear appearance as a background textural element. Generally, the stratification of texture is fairly clear throughout the work. From 2:22 to 4:44, the warblings of an oscillator controlled by a low-frequency square wave become an important element of the texture. At 4:44, a sequence with a strongly defined rhythmic pattern and a variety of percussion and metallic timbres commences. The machine seems to acquire a life of its own as it proceeds to generate a succession of patterns. This passage could perhaps be described as the stirrings of a demented drum kit accompanied by a surreal bass soloist. The pace accelerates subtly and the texture gets busier until there is an abrupt change of texture at around 8:23. From here to the end of side 1, the texture consists predominantly of more delicate timbres, including some ethereal contours created by the play of resonating low-pass filters.

Many of the same techniques are in evidence on side 2 of the recording. The timbres here seem to be somewhat harsher, however, and very "metallic." The spectrum of the sounds is often inharmonic, or if harmonic, the high, closely spaced partials are extraordinarily prominent and therefore noticeably dissonant. The listener familiar with the ways in which rich spectra can be sculpted with filters may want to listen with particular closeness to the virtuosic use of these modules here by Subotnick.

The sequencer is also quite busy on side 2. A notable sequence, consisting of repeated notes descending three steps and then rising in pitch, first appears at 5:49 on side 2. At 6:55, the pattern also appears in inversion (rising three steps, then descending in pitch). This may be the result of applying the technique of control voltage inversion to the output of the sequencer. This inverted form recurs at 12:20, and the original form of the pattern makes a brief subsequent reappearance at 12:48, shortly before the end of the work. A listener will want to study closely the ways in which this and other sequenced series of events are altered upon successive repetitions. These patterns provide just one illustration of the fact that, as an expressive instrument at the hands of a gifted composer, the Buchla synthesizer has rarely been matched as a source of sonic wonders.

A **sequential controller,** also known as a **sequencer,** is a module that can produce a programmed series of discrete control voltages. Connecting the main output of the sequencer to a control voltage input on a VCO makes it possible to produce a repeatable series of pitches. A series of timbres is created if the output of the sequencer is patched to the control voltage input of a VCF. Each control voltage in the series is referred to as a step, or stage, in the sequence. The number of stages in a sequence is typically 8 or 12, but can range from 2 to 256.

The voltage produced at each stage is usually preset by means of a dial. A 12-stage sequencer would typically feature a row of 12 such dials. Some sequencers that can produce three or more control voltages simultaneously might have several rows of dials, called banks.

A pulse generator functions as the sequencer clock, and advances the sequencer from one stage to the next. If a control voltage is applied to the control voltage input of the clock, the duration of each stage can be adjusted automatically. Thus, if a sequence of control voltages is applied to the input of the clock, then rhythmic patterns that are varied and repeatable can be established.

As even this brief description might suggest, a sequential controller can be a complex device, and there are significant differences in design among the many models that have been built and sold. If you should encounter one on an analog synthesizer, you should plan to spend more than a few moments reading the appropriate sections of the operator's manual so that you can make the most effective use of this module.

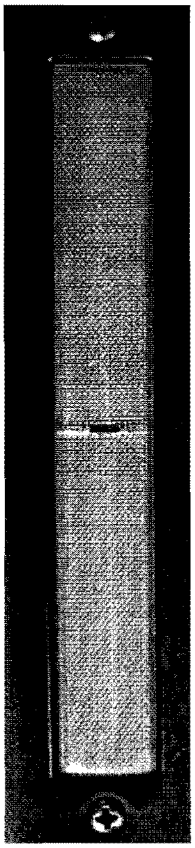


FIGURE 8.30
A ribbon controller.

A variant of the sequential controller is a controller known as an **arpeggiator.** The purpose of this module is to generate control voltages that can be used to produce an arpeggio—a rolling chord, much like that produced by a harp (*arpa,* in Italian). When the arpeggiator is engaged, it will scan the keyboard to identify which keys are being pressed. It will then produce their corresponding control voltages in a sequential fashion. There will probably be a switch on the arpeggiator to determine if this pattern of sequenced voltages is to be confined to its original range or extended to the other octaves covered by the keyboard. Another switch will determine if the pattern begins from the lowest voltage and proceeds to the highest (an ascending arpeggio), if it begins at the top and goes to the bottom (a descending arpeggio), or if it will move up and then back down. Other features may be available as well. Again, it is wise to refer to the operator's manual for a complete description of the capabilities of the arpeggiator on a particular synthesizer.

A fine assortment of even more esoteric controllers has been developed over the years. One of these, called a **ribbon controller** (see Figure 8.30), produces a control voltage that is proportionate to the point along its length at which the ribbon is pressed. Moving the finger up or down the ribbon generates a continuously changing control voltage. This can be patched to a filter to create a smoothly changing timbre; to a VCA to

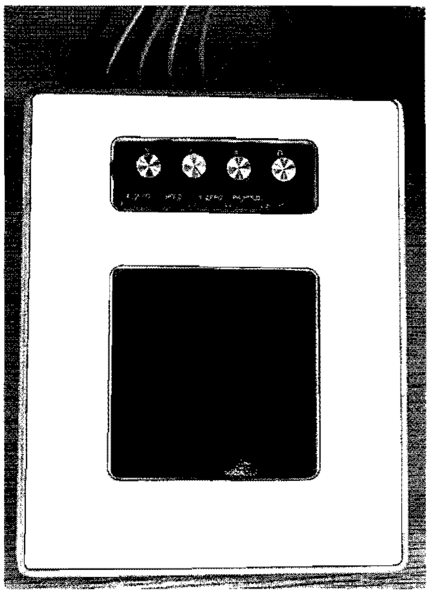


FIGURE 8.31
A touch-sensitive plate

achieve an even, gradual crescendo or diminuendo; or to a VCO to bend the pitch.

An **X-Y controller** (also known as a **joystick**) is operated by moving a stick left and right, and front and back. Two control voltages are generated—one that is proportional to the position of the stick to the left or right (the X axis), and the other determined by the position of the stick to the front or back (the Y axis). Each of these voltages, like that of the ribbon controller, is continuously variable and can be put to many of the same uses as that of the ribbon controller.

A **touch-sensitive plate** (see Figure 8.31) can produce up to four independent control voltages. One is determined by the position, on a continuum from left to right, at which the plate is touched. Another is determined by the position, on a continuum from front to back, of the point of contact on the plate. Yet another is proportional to the pressure applied by the finger to the plate. And yet another is proportional to the surface area of the plate that is covered by the contact.

An **envelope follower** produces a control voltage with a contour that is proportional to the intensity of an external signal patched into the module. Typically, an external source of audio signals, such as a microphone or tape player, provides the original signal. The control voltage that results from following the fluctuations in the intensity of this original signal—in effect, the envelope of the original signal—can then be patched to the control input of a VCA or VCF. An electronic sound being processed by the VCA or VCF can thus be shaped by envelopes modeled after those of the external sounds (see Figure 8.32).

A **sample-and-hold unit** (or **S/H controller**) also requires an input signal. This input signal is measured, or sampled, from time to time. The level of the input voltage at the moment of sampling is then "held" until the next sample is taken. Thus, a continuously variable input voltage can be con-

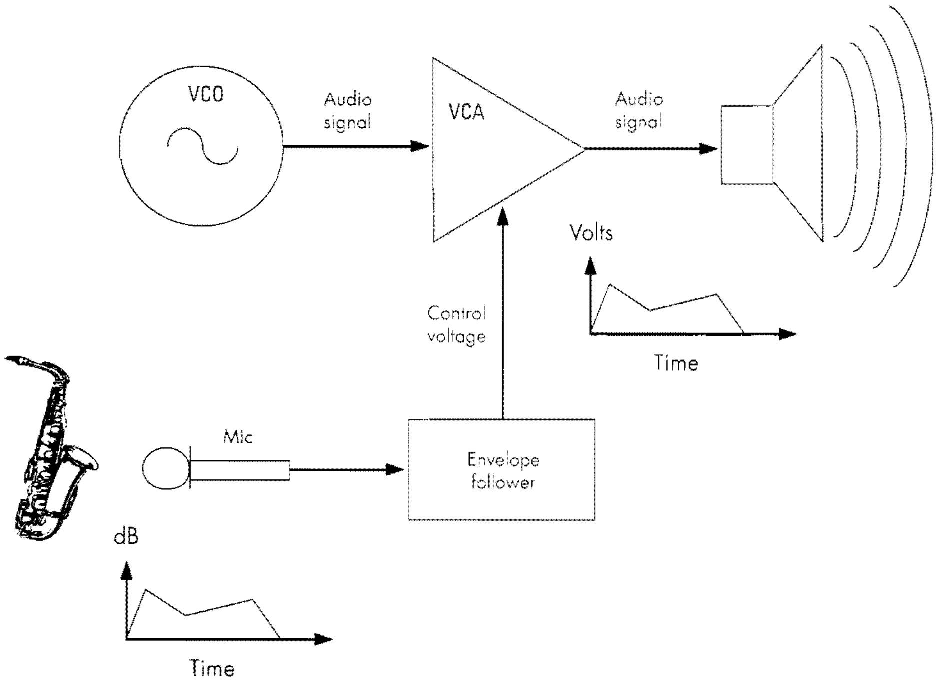


FIGURE 8.32
One potential application of an envelope follower. Here the amplitude envelope of the saxophone tone is converted into a control voltage envelope. This can then be applied to the VCA to shape the amplitude of the audio signal produced by the VCO.

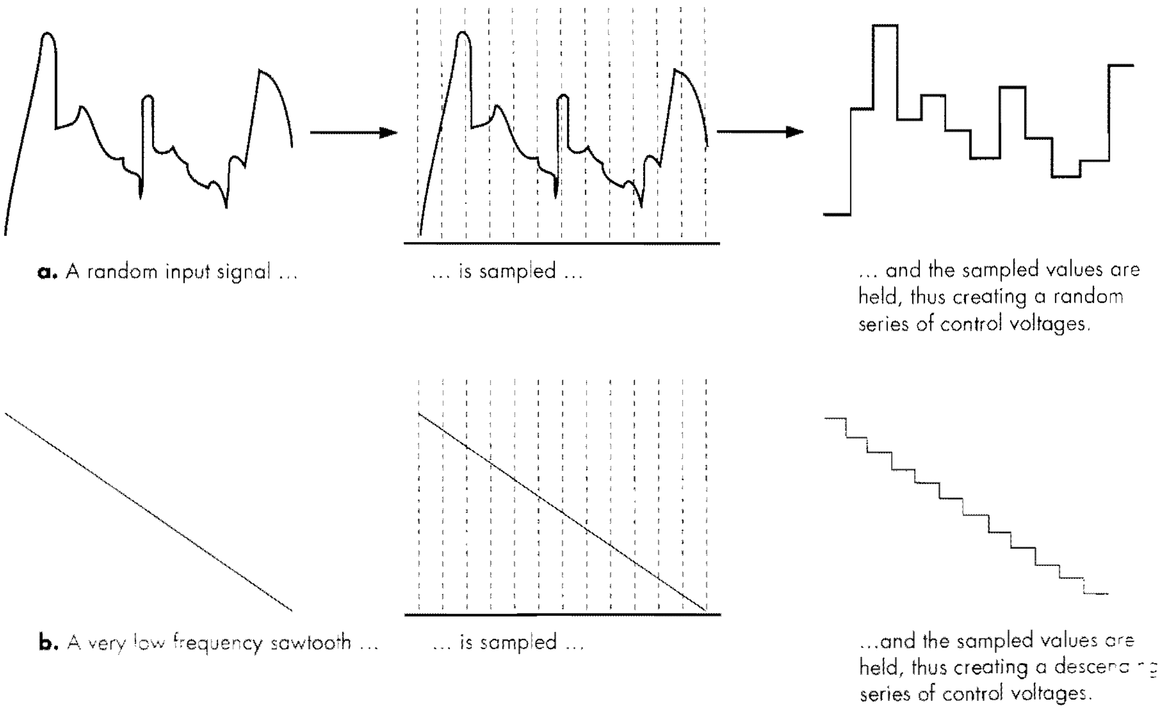


FIGURE 8.33
Two examples of series of control voltages that can be created by a sample/hold unit.

sample is taken. Thus, a continuously variable input voltage can be converted into a series of discrete voltage levels (see Figure 8.33). If the input signal is random, as is the case with white or pink noise (illustrated in Figure 8.33a), then the output signal will be a series of random voltages that can be patched to the control input of a VCO, for example, to produce a random series of pitches. If the input signal is a periodic waveform of very low frequency, then the output signal will be a patterned series of voltages (as in Figure 8.33b). Some sample-and-hold units provide for voltage control of the sampling rate, so that durational patterns can also be created.

THE DEMISE OF ANALOG SYNTHESIZERS

At first, nearly all analog synthesizers were studio instruments (with the notable exception of the Synket). Though certainly not as bulky as the RCA synthesizer of the 1950s, they were nonetheless rather large compared to most musical instruments and rather difficult to pack and transport for a performance tour. They were also complicated to set up and operate on stage. The modular design facilitated great flexibility in the production of sounds, but required the proper connection of numerous patch cords and the correct setting of dozens of dials and switches. Only the bravest of electroacoustic musicians would attempt these things while in the presence (and under the pressure) of an eager and expectant audience.

By the early 1970s, however, manufacturers began to introduce analog synthesizers that were specifically designed for live performance. Not only were they more compact, but they were also somewhat less complicated to operate. Many of the most common connections among modules were wired internally, and other connections could be made by pushing switches on the front panel. Thus, it became possible for a performer to set up the patch for a sound while onstage. The gain in relative ease of operation meant somewhat less flexibility in the design of sounds, but this loss was considered an acceptable trade-off by electroacoustic musicians who aspired to get onstage.

The Mini-Moog, introduced in 1970, was one of the first performance synthesizers. Its controls included two wheels at the left end of the keyboard for bending the pitch and for regulating vibrato. These controller wheels have since become standard equipment on performance synthesizers. Other performance synthesizers introduced at about this time were the ARP 2600, the ARP Odyssey, and the Synthi AKS.

While the elimination of most patch cords certainly made the task of live performance much easier, it was still necessary to change manually the settings of the dials and switches when moving from one kind of sound to another. By the late 1970s, however, so-called programmable synthesizers were developed that made it possible to change a patch at the press of one or two buttons. The settings of the dials and switches could be scanned and digitally encoded for storage until needed later. Entire banks of control

THE VOCODER

In the 1930s, researchers at the Bell Telephone Laboratories, searching for a more efficient way to transmit telephone messages, developed a device called a **vocoder** (for VOice enCODER). It was found to be of limited use to the telephone company, but in the 1950s the composers in the studios for electronic music in Cologne and New York became interested in the device as a way to transform sounds.

The basic function of a vocoder is to model the spectral characteristics of one sound after those of another sound. Or, to put it another way, the spectral characteristics of a sound are superimposed from a different sound. Figure 8.34 is a diagram of a network of analog circuits designed to accomplish this. The sound to be modeled is provided as an input to a bank of band-pass filters, each covering perhaps only one-third of an octave within the range of audible frequencies. An envelope follower circuit for each frequency band then detects the amplitude envelope of the output from each of the filters. In this way, the amplitude changes within each frequency band of the input sound are analyzed.

The sound upon which this spectral information is to be superimposed is provided as an input to a second bank of band-pass filters, covering the same frequency bands as those in the analysis filter bank. Each band of frequencies in the target sound is then passed to a voltage-controlled amplifier. The control voltage for each VCA is provided by the corresponding envelope follower in the analysis bank. In this way, the amplitude envelopes from each frequency band of the model sound are used to shape the amplitudes of each band of frequencies in the target sound. The output of each VCA is then passed to a mixer and the target sound is reassembled, now with new spectral patterns.

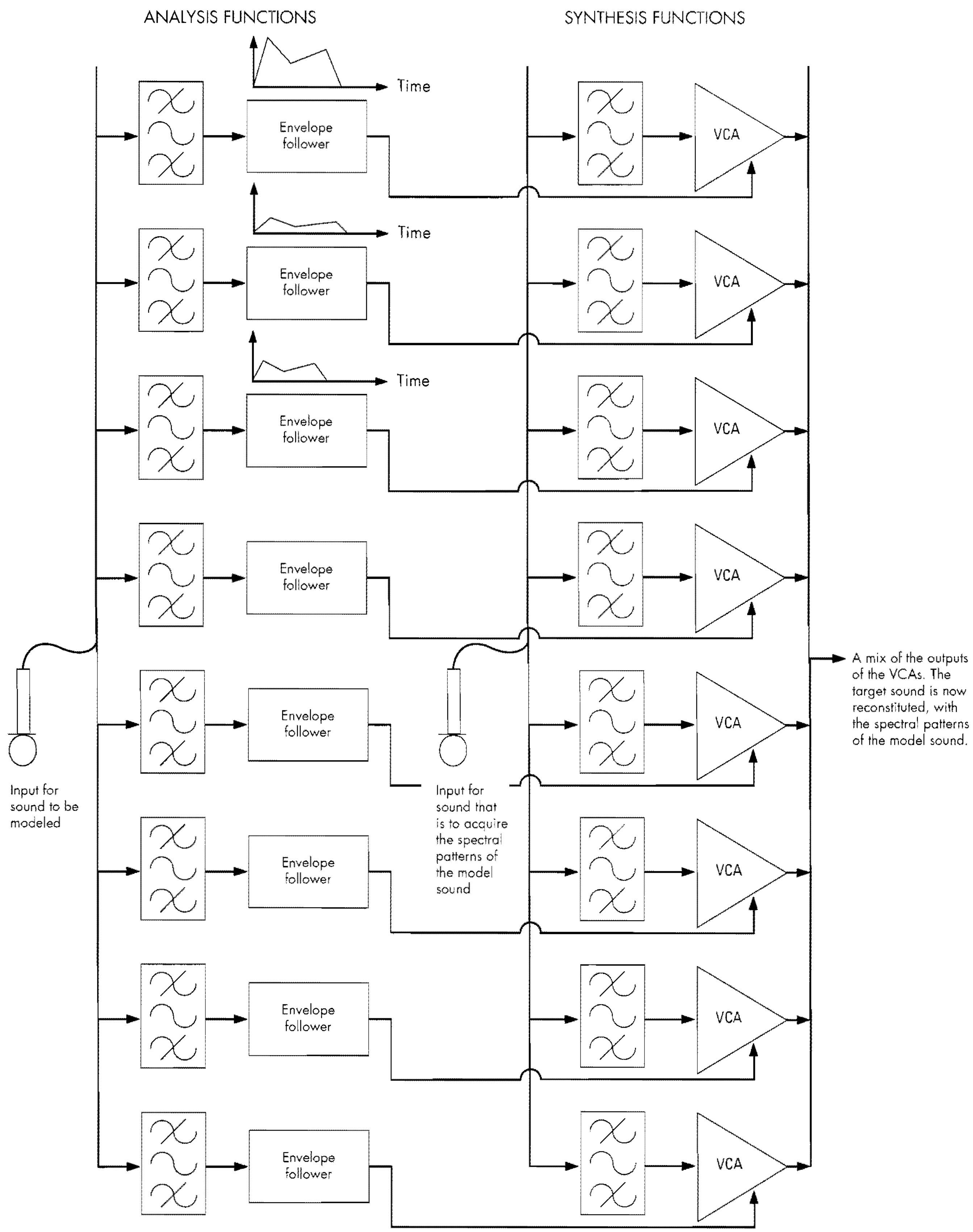
For the technique to provide satisfactory results, the second sound should have at least as rich a spectrum as the first sound. Noisy sounds, or sounds with sawtooth waveforms, are typically effective. Common examples of vocoding include robotic voices in grade-B science fiction films, or talking windstorms and other such special effects in similar films. Many artists, such as Laurie Anderson, have made much use of the vocoder in very effective ways in their music.

The functions of a vocoder circuit can now be emulated digitally, and a vocoder function is found on many commercially available effects devices. Such digital signal processors, described in Chapter 4, are esteemed as especially versatile and useful devices in the studio.

FIGURE 8.34 (facing page)
A schematic diagram of an analog vocoder.

settings could be stored in this way. Any of the patches required during a performance could then be recalled, virtually in an instant. The first performance synthesizer to implement this technology was the Sequential Circuits Prophet-5, introduced in 1978. Other popular programmable synthesizers were the Korg Polysix (1982), shown in Figure 8.35, and the Roland Jupiter 8.

Analog synthesizers have other limitations that have not been so easy to address. For example, subtractive synthesis is a relatively crude way to specify and shape a timbre. The technique of biting away chunks of the



ANALYSIS FUNCTIONS
SYNTHESIS FUNCTIONS
Time
Time
Time
Envelope follower
VCA
Input for sound to be modeled
Input for sound that is to acquire the spectral patterns of the model sound
A mix of the outputs of the VCAs. The target sound is now reconstituted, with the spectral patterns of the model sound.

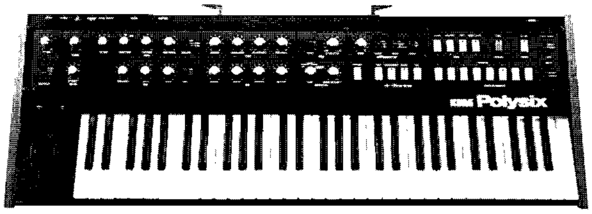


FIGURE 8.35
The Korg Polysix synthesizer. (Courtesy of Korg USA, Inc.)

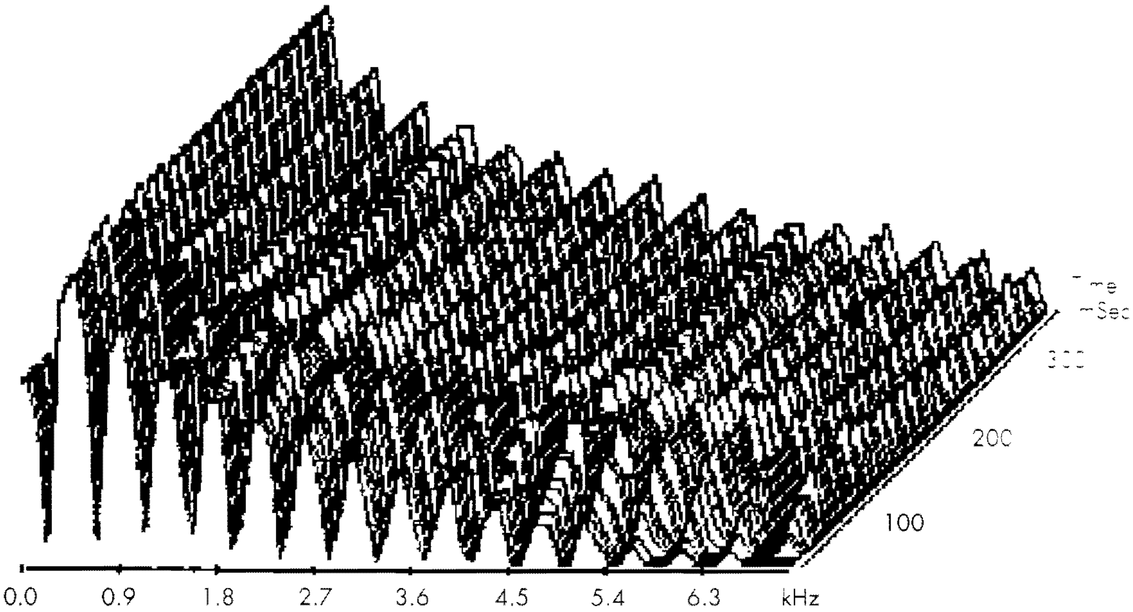
spectra of fixed waveforms by using relatively imprecise filters does not permit a great range of subtlety.

Amplitude envelopes tend to be similarly crude. Rather than an individual envelope for each frequency component of a timbre, as research has indicated to be the case with natural sounds, the VCA superimposes a single envelope over the whole sound, as illustrated by the perspective plots in Figure 8.36 (contrast these to the perspective plots in Figures 7.8 and 7.9).

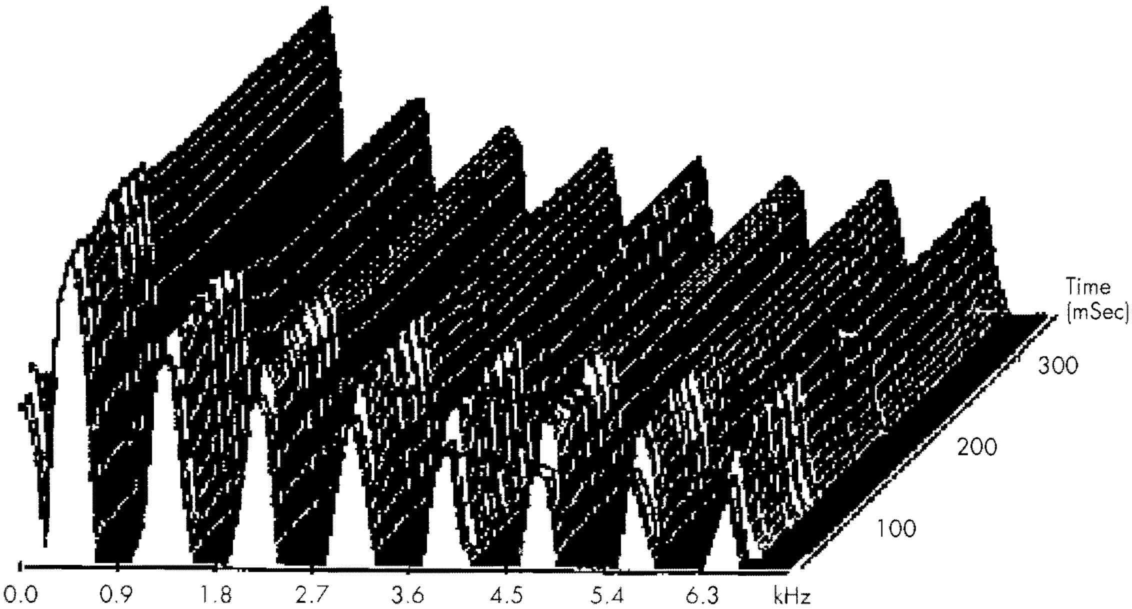
Furthermore, analog electrical circuits tended to be rather unstable over the duration of a performance or a studio session. This is particularly noticeable as VCOs drift in and out of tune. Finally, there was the problem that synthesizers made by different manufacturers were almost totally incompatible with one another. Without a satisfactory way to connect synthesizers of different manufacturers, there was no practical way to integrate the strengths of one synthesizer with the strengths of the other synthesizers in the studio or onstage. Setups involving several instruments could be very difficult to manage.

By the end of 1983 there were two significant developments that would eventually lead to the near extinction of analog synthesizers. Yamaha introduced the first commercial digital synthesizer, the DX7. At the same time, a group of manufacturers announced their agreement on a set of digital messages and the corresponding hardware requirements that would enable control information to be exchanged among synthesizers. This, of course, was the introduction of MIDI, the Musical Instrument Digital Interface.

By the time of this writing, the techniques of analog synthesis have been almost entirely supplanted by those of digital synthesis. Even so, it is important that everyone who is interested in electronic music find an opportunity to spend at least some time with an analog synthesizer. Many of the concepts and paradigms that were introduced with analog synthesis continue to be important. Perhaps of the greatest importance, however, is the sense of extended musical possibilities that these machines fostered. The modular design of analog synthesizers virtually invited experimentation with sound and form. Modern, digital synthesizers seem to be much more tightly locked into the "normal" 12-equal-tone-per-octave paradigm. Some of the most fascinating sounds produced by analog synthesizers—the grand sweep of a resonant low-pass filter, or the truly random series of pitches produced by a VCO controlled by a sample-and-hold unit—have yet to be satisfactorily recreated by even the most modern digital machine. This belies the notion (to be discussed further in Chapter 12) that progress is an inseparable companion to technological change.



a. A perspective plot of a sound with a sawtooth waveform. Note the highly uniform distribution of partial frequencies and the flatness of their envelopes relative to those of acoustic instruments (compare Figures 7.8 and 7.9).



b. A perspective plot of a sound with a square waveform. Again note the very regular spacing of the partials and the extreme uniformity of their envelopes. Incidentally, observe also that in the square-wave timbre, every other overtone is absent (compared to the sawtooth-wave timbre shown in (a).

FIGURE 8.36
The uniformity of spectral characteristics of simple, electronically synthesized tones. A major challenge of synthesis is to create tones with much less regimented patterns of activity. (Courtesy of Digidesign, Inc.)

IMPORTANT TERMS

digital synthesizer
analog synthesizer
elektronische Musik
control voltage
modules
patch
voltage-controlled oscillator (VCO)
triangle wave
pulse wave
duty cycle
square wave
noise generator
voltage-controlled amplifier (VCA)
voltage-controlled filter (VCF)
low-pass filter
cutoff frequency
filter slope, or roll-off
Q
high-pass filter
band-pass filter
bandwidth
band-reject filter, or notch filter
subtractive synthesis
additive synthesis
envelope generator
attack
initial decay
sustain
release, or final decay
ADSR
trigger
gate
monophonic
polyphonic
keyboard tracking
control voltage attenuator
control voltage inverter
quarter tone
low-frequency oscillator (LFO)
sidebands
carrier
modulator
amplitude modulation, or AM
balanced modulator, or ring modulator
frequency modulation, or FM
modulation index
sequential controller, or sequencer
arpeggiator
ribbon controller
X-Y controller, or joystick
touch-sensitive plate
envelope follower
sample-and-hold unit, or S/H controller
vocoder

FOR FURTHER READING

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