

3. Would it be possible to broadcast stereophonic programs on AM radio? Why?
4. Why is intermodulation distortion generally more objectionable to the ear than an equal amount of harmonic distortion?
5. Why would Dolby Digital give more realistic sound than Dolby ProLogic?

EXERCISES

1. Compare the voltage outputs from two microphones: Microphone A has a sensitivity -60 dB re 1 V per 1 Pa and microphone B has a sensitivity of -66 dB compared to the same reference.
2. Calculate the actual voltage output of microphone A at a sound pressure of 1 Pa. To what sound pressure level does this correspond (see Chapter 6)?
3. If two $8\text{-}\Omega$ speakers are connected to the same output terminals of the amplifier, what net impedance will be "seen" by the amplifier?
4. An amplifier has a damping factor of 30 when used with an $8\text{-}\Omega$ speaker. What will the damping factor be if used with a $16\text{-}\Omega$ speaker?
5. An $8\text{-}\Omega$ loudspeaker is capable of handling 30 W of audio power. How much current will flow when it is receiving 30 W of power? What size of fuse or circuit breaker should be used to protect it from overload?
6. What is the wavelength at the center of the FM band? (See Exercise 1 in Chapter 3.) How long should the horizontal loop of an FM antenna be, if it is to be one-half of a wavelength (in order to be resonant)?

EXPERIMENTS FOR HOME, LABORATORY, AND CLASSROOM DEMONSTRATIONS

Home and Classroom Demonstrations

1. *Microphone styles* Acquire different types of microphones (including dynamic, condenser, and electret microphones) in various configurations to show the variety of microphones available in the market. Each microphone should be discussed in terms of methods of transduction, functional design, acoustical and electrical characteristics, and application in the studio. As an alternative, a catalogue (or the Internet) containing a variety of microphones could be used to select a microphone for a specific application based on its characteristics.
2. *Proximity effect* Connect a cardioid microphone to an amplifier and loudspeaker. Demonstrate the increased bass response as the source (talker or singer) gets closer to the microphone. Discuss why pop singers seem to swallow their microphones. Repeat the experiment with an omnidirectional (omni) microphone and note the difference.
3. *Sensitivity measurements* Connect a pink noise source to an amplifier and loudspeaker. Adjust the volume control to read 90 dB on a sound level meter at a given distance from the source. Measure the voltage (with a high-impedance voltmeter) on the output of a microphone at the position of the sound level meter.
4. *Phantom power* Connect a condenser microphone to a mixer with switchable phantom power. Turn the phantom

power off and determine the output level as you sing into the microphone. Turn the phantom power on and note the difference.

5. *Preamp clipping* Apply a sine wave to the input of a preamplifier and view both the input and the output signals on an oscilloscope. Note that the output signal is also a sine wave, but with an increased amplitude. Determine the gain of the amplifier by comparing the output amplitude with the input amplitude. Increase the amplitude of the signal generator until the output waveform is no longer a sine wave. The new waveform is produced by clipping the signal from overloading the input. To hear the harmonic distortion from clipping, connect a power amplifier and loudspeaker to the output of the preamp. You should hear the change of timbre of the tone as soon as it starts clipping.

6. *Series and parallel connection of loudspeakers* Connect an 8-ohm loudspeaker to the output of an amplifier and listen to some piece of music. Without changing the settings on the amplifier attach two 8-ohm loudspeakers in series to the output and listen to the same music. Now attach the two 8-ohm loudspeakers in parallel and listen again. Compare what you heard in the three cases.

Laboratory Experiments

Audio tests and measurements: an introduction (Experiment 27 in *Acoustics Laboratory Experiments*)

Amplifier performance: frequency response and harmonic distortion (Experiment 28 in *Acoustics Laboratory Experiments*)

Amplifier performance: hum, noise, and intermodulation distortion (Experiment 29 in *Acoustics Laboratory Experiments*)

Microphones (Experiment 32 in *Acoustics Laboratory Experiments*)

CHAPTER

27

Analog Electronic Music

As we saw in the last chapter, an understanding of electricity and electronics technology, combined with an understanding of the scientific principles of acoustics and music, allows the creation of modern music technology in its many forms. Advances in technology usually create opportunities to do things that were not previously possible (which is why they are called advances). What actually gets done, however, depends not only on technological opportunities but also on the concepts in the minds of those who exploit them. Instead of having the fixed physical structures of traditional musical instruments, electronic instruments can synthesize or capture virtually any sound. Musical concepts in the minds of the founders of electronic music forever changed notions of what musical instruments and music itself are, or could be, giving rise to musical explorations of entirely new kinds, much to the delight (and chagrin!) of listeners ever since. The initial implementation of electronic music technology in analog form provided both the crucible and the conceptual basis for the more powerful digital forms to come later. In this chapter we will examine these basic concepts from the standpoint of how electronics technology impacted fundamental concepts of music and musical instruments starting around the middle of the twentieth century.

In this chapter, you should learn:

- About the basic elements of electronic music;
- About the conceptual and technological basis of *musique concrète*;
- About the conceptual and technological basis of *electronische musik*;
- About analog synthesizers;
- About digital control of analog synthesizers.

27.1 ■ THE ELEMENTS OF ELECTRONIC MUSIC

Electronic music can be described in many ways, but from a technological standpoint it is useful to define a set of basic functional elements from which most electronic music techniques can be discussed and understood. Figure 27.1 shows such a set of basic elements, many of which have been described elsewhere in this book.

Microphones

In electronic music terms, a *microphone* is really a form of sensor, but it is so important that it deserves a special category of its own. A microphone transforms sound pressure

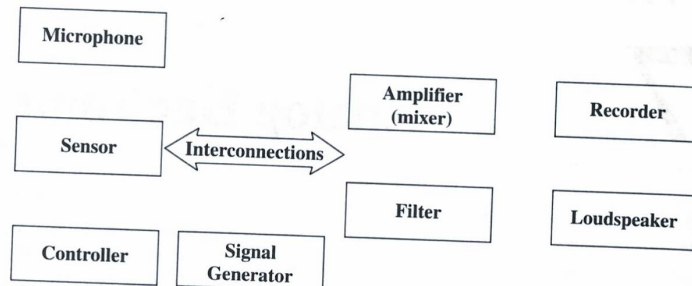


FIGURE 27.1
The basic elements
of electronic music.

variations at a particular point in space into analogous variations in some electrical quantity such as voltage (see Chapter 20). As we shall see, it is possible to create electronic music in which microphones play anywhere from an essential role to no role at all, just as it is possible to create visual art that may or may not involve the use of a camera.

Sensors

The *sensor* element refers to transducers, such as knobs, dials, switches, keys, pedals, and other controls that might be manipulated by a musician in order to convey musical information. They generally sense the position, velocity, or acceleration of physical manipulations or in some cases sound waves, heartbeats, electrical currents in the brain, temperature or altitude changes in the atmosphere, etc. Traditional musical instruments obviously involve a variety of ways of “sensing” the actions of a performer in ways that affect the sounds produced.

Controllers

A *controller* might be a sensor as well, if the output of a sensor is used to control the actions or signals used by other elementary devices. More sophisticated controllers are possible, however, from pinned rotating cylinders such as those described by the *Banū Mūsā* to stored program computers executing sophisticated algorithms for musical decision making.

Signal Generators

A *signal generator* is an artificial source of signals that might be used for control purposes or as an oscillatory waveform that represents one or more (or all) components of a sound. It is a fundamental alternative to the microphone in the sense that it is a possible source of sound signals. Electronic music systems might, of course, contain both microphones and signal generators.

Interconnections

In order to utilize more than one of the elements described here, it is necessary to interconnect them in various ways. The way in which a given set of elements is interconnected can completely change the significance of any given element. Because the number of elements in a given electronic music system may be large, important strategies have been devised to allow flexible and even dynamic ways of achieving interconnections.

Amplifiers and Mixers

Amplifiers are useful both as sources of electrical power that drive electronic signals along interconnection paths and for driving loudspeakers. Amplifiers and related devices are essential parts of many electronic devices that deal with sound signals. Amplifiers are also essential components of *mixers*, which are essentially amplifiers with multiple, individually controllable inputs.

Filters

A *filter* element in a restricted sense is a device that modifies the amplitude or phase of a signal in a frequency-dependent manner. More generally, filters can modify or process signals in any manner whatsoever. A linear filter, for example, outputs only frequencies that are present at its input. A nonlinear filter, however, can output frequencies that are not present at its input. Filters can be used not only to modify signals but to measure or analyze them in various ways.

Recorders

A *recorder* is any device that stores a signal in such a way that it can be later retrieved. A recorder is a form of memory.

Loudspeakers

A *loudspeaker* is a transducer that renders the acoustic information in a signal as vibrations in the atmosphere that can be perceived as sound by one or more listeners (see Chapter 19). It can be anything from a simple earphone to a coordinated array of transducers intended to generate a coherent pressure wave field containing spatial information about complex sounds.

27.2 ■ *MUSIQUE CONCRÈTE*: THE CONCEPTUAL BASIS

In 1948, Pierre Schaeffer defined a new kind of music by assembling a succession of recorded sounds, the first case consisting of a rhythmic succession of locomotive steam and wheel sounds punctuated by steam whistles. He later wrote in his *A la Recherche d'une Musique Concrète* (*Musique Concrète Research* 1952):

This determination to compose with materials taken from an existing collection of experimental sounds, I name *musique concrète* to mark well the place in which we find ourselves, no longer dependent upon preconceived sound abstractions, but now using fragments of sound existing concretely as sound objects defined and whole

The fundamental idea of *musique concrète* is thus to replace the (abstract) sounds of traditional musical instruments with the (concrete) sound of nature. The first example was Schaeffer's musical composition *Etude aux Chemins de Fer* (*Railroad Study*). There are a few notable examples of the incorporation of natural sounds into music before 1948, such as Ottorino Respighi's incorporation of recorded nightingales in his orchestral piece

Pines of Rome (1924) and George Antheil's use of airplane engines on stage in his *Ballet Mécanique* (1926). Before the advance of more sophisticated sound-recording technology, however, such examples were limited to more-or-less verbatim quotations of natural sounds.

The earliest experiments of Pierre Schaeffer and his colleague Pierre Henry occurred at the *Radiodiffusion Française* (French Radio) in Paris. Sounds were at first recorded onto a rotating disk directly with a vibrating lathe. Such recordings could be edited only by playing back multiple disks at the same time and switching among them with a mixer. Although editing sounds in this manner is possible, it certainly is not quick or easy. Prospects for *musique concrète* were furthered greatly with the advance of magnetic recording. Magnetic tape recorders, in particular, allowed sounds to be edited simply by cutting the tape with a razor blade and splicing the cut pieces back together in various ways. This is, in fact, the way that much electronic music was assembled until the advance of digital sound editing and mixing in the 1980s.

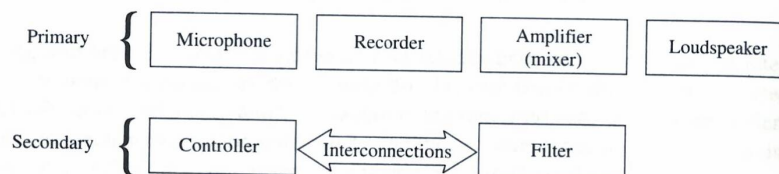
Magnetic tape recording allowed recorded sounds not only to be quoted literally but also to be repeated, or looped, simply by splicing the end of a piece of magnetic tape back to its beginning, forming a *tape loop*. Looping tape-recorded sounds in this manner could, for example, impart to them a rhythmic quality that they may not have possessed in their original form. Not only could any sound be repeated, but its tape playback speed could be changed, thus accelerating or decelerating the rhythm (with corresponding changes in pitch). By playing the tape backward, the sound could also be reversed in time, yielding an entirely new perspective on what is otherwise a natural sound. Looping techniques in modern samplers owe both their name and origin to such tape-based techniques.

Pierre Schaeffer followed his *Railroad Study* with other examples of *musique concrète*, including *Etude aux Tourniquets* (sounds of toy spinning tops and percussion instruments), *Etude Violette*, *Etude Noire* and *Diapason Concertino* (piano sounds), and *Etude Pathétique* (sounds of saucepans, canal boats, voice, harmonica, and piano). Recordings of traditional musical instruments are sometimes used in *musique concrète*, but these instruments are generally serving in their capacity as sources of sound objects as well as occasional references to traditional music.

27.3 ■ MUSIQUE CONCRÈTE: THE TECHNOLOGICAL BASIS

In terms of the elements of electronic music described in Fig. 27.1, we see the four key elements of *musique concrète*: the microphone, the recorder, the mixer, and the loudspeaker. Of course, these elements need to be interconnected, amplifiers (and possibly filters) are needed for the speaker(s), and so on, but these are incidental to the concept of *musique*

FIGURE 27.2
The basic elements
of *musique*
concrète.



concrète (see Fig. 27.2). Other issues include the need to play back the recorded sounds in various ways. Opportunities for different kinds of playback vary according to the details of the recorder. Disk recorders might allow random starting points and variable speeds of playback. Tape recorders allow variable speeds, directions, and looping of playback, though they do not allow random starting points as well as disk recordings. Tape recordings, however, can be copied, cut into various pieces, and reassembled in various ways, effectively allowing flexible temporal control of playback.

27.4 ■ ELEKTRONISCHE MUSIK: THE CONCEPTUAL BASIS

As *musique concrète* was being developed in France, a group of scientists, engineers, and musicians in Germany began working on another extension of traditional music that became known as *elektronische musik*, or electronic music. Just as *musique concrète* was based on modified and reordered recordings of natural sounds, *elektronische musik* was based on the notion that virtually any sound could theoretically be constructed by adding together appropriate mixtures of sinusoids and other artificially synthesized sounds.

The initial impetus for *elektronische musik* came in 1948 when Homer Dudley, of Bell Telephone Laboratories in Murray Hill, New Jersey, visited Werner Meyer-Eppler, then the director of the Institute of Phonetics at the University of Bonn. Dudley was renowned for his work on a device called the *vocoder* (for *voice coder*), which used filters to both analyze and synthesize the sounds of human speech. The vocoder was the basis for a similar device—also due to Dudley—featured at the 1939 World's Fair in New York City. Called the *voder* (or *voice operation demonstrator*), a highly skilled operator “performed” speech by pressing on keys while simultaneously operating pedals, knee-operated sliders, and a variety of other control devices. Fair visitors were amazed that the operator could make Dudley's *voder* say understandable things in languages she didn't speak and even to sing in limited ways.

The result of these meetings were demonstrations and collaborations resulting in a series of lectures given in Darmstadt by Meyer-Eppler and Robert Beyer during the summer of 1950 entitled “The World of Sound of Electronic Music.” As a result of this and related activity, the technical staff at *Westdeutscher Rundfunk* (WDR, for West German Radio) in Cologne committed resources to the idea of establishing a studio to investigate the musical ramifications of such ideas. The first musical result of the Cologne studio for *elektronische musik* was a composition completed in 1952 called *Musica su Due Dimensioni* for flute, percussion, and electronically generated tape by composer Bruno Maderna, who worked in collaboration with Meyer-Eppler.

The first director of the Cologne studio for *elektronische musik* was a composer named Herbert Eimert, who was very much a proponent of a compositional technique called *serialism*. In the early 1920s the Viennese composer, music theorist, and painter Arnold Schoenberg published his method of composing music with 12 tones. Schoenberg showed that basing an entire musical composition on a fixed sequence of the 12 different equal-tempered pitch classes can lead to a conceptual unification of musical materials not unlike the traditional system, which relates all pitches to a single key-note, called tonality. Because it was an alternative to the traditional system of tonality, Schoenberg's approach has sometimes been called *atonality*, though this is a linguistic misnomer.

Schoenberg had two very influential students named Alban Berg and Anton Webern. Webern, in particular, extended Schoenberg's ordering principles beyond pitch rows to fixed sequences for virtually every musical characteristic, including loudness (musical dynamic), duration (rhythmic value), articulation (*staccato*, *legato*, *pizzicato*, etc.), and tone color (*timbre*). This total-ordering approach to music composition became known as serialism and attracted many influential composers in Europe (such as Boulez, Stockhausen, Nono, Dallapiccola, and Ligeti) and America (such as Stravinsky, Babbitt, Carter, Wuorinen, Davidovsky, and Martino).

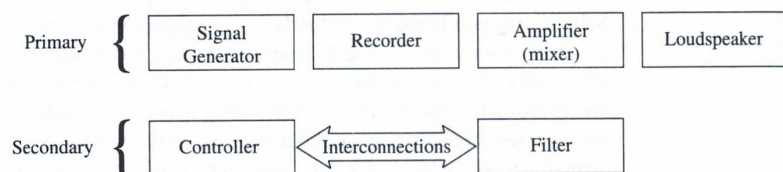
In effect, the amplitudes, frequencies, and phases of sinusoids (as well as any other sound parameters that composers could think in terms of) became grist not only for the mill of *elektronische musik* but for serialist composers as well. Eimert (1959) wrote: "It is certain that no means of musical control could have been established over the electronic material had it not been for the revolutionary thought of Anton Webern."

27.5 ■ ELEKTRONISCHE MUSIK: THE TECHNOLOGICAL BASIS

In terms of the elements of electronic music described in Fig. 27.1, the key elements of *elektronische musik* were the signal generator, the recorder, the mixer, and the loudspeaker. As in the case of *musique concrète*, these elements need to be interconnected, amplifiers (and possibly filters) are needed for the speaker(s), and so on (see Fig. 27.3). In some ways, and especially because of equipment limitations, the need to mix several sounds together and the need to play back the recorded sounds in various ways are essential to the practice of *elektronische musik* if not necessarily its theory. The equipment complement at the original studio in Cologne, for example, consisted of a sine-wave generator, a white-noise generator, two electronic keyboard devices, a mixer, a four-track tape recorder, and a monophonic tape recorder. Composers worked by recording a sinusoid on one channel of tape, then another (possibly with a different loudness contour, or "envelope") on a second channel, then a third, and then a fourth. The duration of the sound was determined by cutting the tape, which traveled at 15 or 30 in/s. These might be mixed together and rerecorded on the monophonic recorder, then transferred back on channel one of the four-track recorder, etc. When all the bits and pieces of a complete composition were prepared, they could all be spliced together into a final result.

Although theoretically sound, *elektronische musik* was extremely tedious to produce in this manner. The advantages of constructing sounds on such a microscopic level were often offset by the difficulty of the procedure, the opportunities for errors, and limitations caused by the buildup of tape hiss and other noises during multilayered mixing operations.

FIGURE 27.3
The basic elements
of *elektronische*
musik.



27.6 ■ TAPE MUSIC

It soon became apparent to practitioners of both *musique concrète* and *elektronische musik* that the philosophical differences in sound source (tape-recorded natural sounds versus electronic signal generators) were less important than the manipulative opportunities offered by the new techniques of electronic music. Karlheinz Stockhausen, one of the principle proponents of *elektronische musik*, as a result of his work in Schaeffer's studio in 1952–1953, later observed: "Wherein lies the difference between instrumental sounds, between any audible events: violin, piano, the vowel a, the consonant sh, the wind?" One result of such observations was Stockhausen's landmark composition *Gesang der Jünglinge*, which is based on sounds recorded from the speech of a young boy blended with electronically produced tones. The result is experienced by the listener as a kind of sound-continuum, in which the natural and the artificial together provide a sound palette much richer than either does alone.

Other composers in other places soon began to carry out experiments using electronic music. Notably, Toshiro Mayazumi, who had worked with Schaeffer in Paris, and others started an electronic music studio in 1953 at NHK (Nippon Housō Kyōkai, or Japan Broadcasting Corporation) in Tokyo. This effort resulted in Mayazumi's *Etude I* in 1955 as well as later works by composers Minao Shibata, Makoto Maroi, Toshi Ichihyanagi, and Joji Yuasa.

In 1952, composers Vladimir Ussachevsky and Otto Luening presented works of their own at a concert organized by famed conductor Leopold Stokowski at the Museum of Modern Art in New York City. This concert of electronic music was the first of its kind in the United States, and it led to a demonstration and interview on the *NBC Today* show by Ussachevsky and Luening. As a result of the pioneering work of Ussachevsky and Luening, the Rockefeller Foundation funded the first studio for electronic music in the United States at Columbia University in 1955.

Joining forces with composer Milton Babbitt of Princeton University, Ussachevsky and Luening founded the Columbia-Princeton Electronic Music Center. This center comprised several tape studios and, notably, the first significant electronic music synthesizer.

27.7 ■ EARLY ELECTRONIC MUSIC SYNTHESIZERS

As electronic music studios grew in size and number, several problems became increasingly evident. The equipment they contained tended to be very expensive and required a great deal of technical knowledge to use and maintain. The early electronic music studios were similar in their requirements to radio stations and recording studios, where they were therefore often located. Institutional-sized budgets and technical staffs were needed to keep them going.

As the amount of equipment increases, the problem of interconnections increases combinatorially. Straightforward patch cords soon become tangled forests when more than a few devices are interconnected, especially when many sound channels are needed simultaneously. Finally, when many devices are used, each typically requires from a few to many parameters to be set, such as the frequency, waveshape, and amplitude of each signal generator. It is frequently desirable to make changes in interconnections and parameters in a

time-dependent manner, depending on the requirements of the musical composition being created.

By integrating and coordinating the operation of most or all the elements of electronic music (see Fig. 27.1), synthesizers address these problems of size, cost, and flexible, dynamic interconnections and parameter control.

27.8 ■ THE MARK II ELECTRONIC MUSIC SYNTHESIZER

In 1955, the RCA Electronic Music Synthesizer (see Fig. 27.4) was designed and built by Harry Olson and Herbert Belar at RCA's Sarnoff Laboratories in Princeton, New Jersey. A more advanced version—the Mark II—was installed in 1959 at the 125th Street studio of the Columbia-Princeton Electronic Music Center in New York at a cost of \$10,000, funded by a grant from the Rockefeller Foundation. Its principle users were composers, who described it in the following terms.

(Charles Wourinen) ... a 750-vacuum-tube affair in which information was encoded by a fiendish combination of 4-bit binary switches, banks on the walls and console of this nearly room-sized machine, and two particularly clever paper drives, each of which would encompass two channels of information. Holes were punched in the paper that then passed over a metal roller to which contact was made by a set of brushes. The brushes were arranged so that they would lie over the holes that passed beneath, making continuous contact. Time was represented on the machine as the number of holes at a certain rate of the paper drive. Pitch

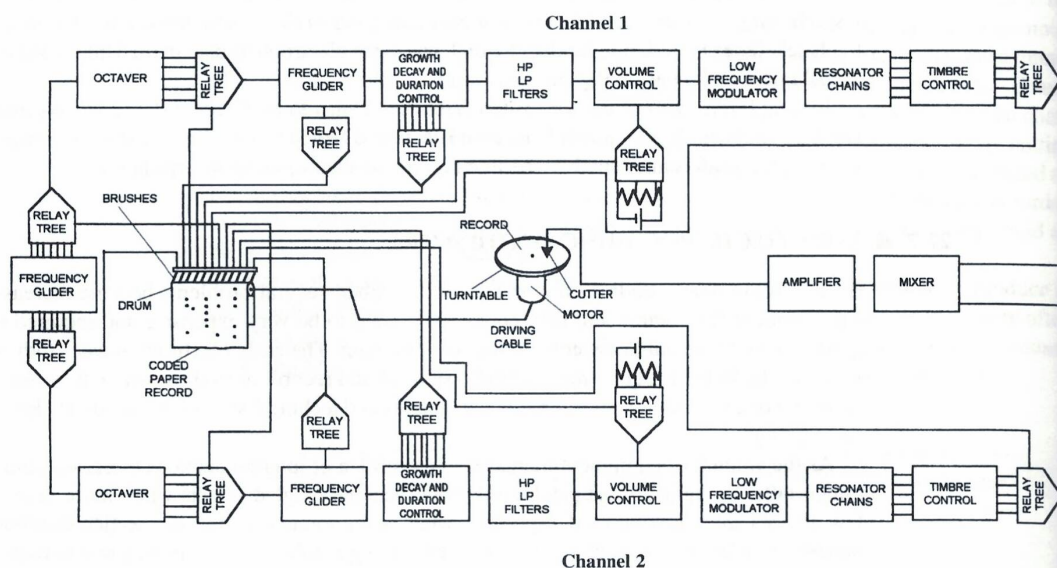


FIGURE 27.4 Schematic of the RCA electronic music synthesizer designed by Olson and Belar (ca. 1955).

and other information were represented in binary numbers by a combination of preset switches and banks of holes in the paper (Boulanger 1984).

(Milton Babbitt) The machine was extremely difficult to operate. First of all, it had a paper drive, and getting the paper through the machine and punching the holes was difficult. We were punching in binary. The machine was totally zero, nothing predetermined, and any number we punched could refer to any dimension of the machine. There was an immense number of analog oscillators, but the analog sound equipment was constantly causing problems. I couldn't think of anything that you couldn't get, but other composers gave up—it was a matter of patience There were many people who would look at this machine and say, "It's a computer." But it never computed anything. It was basically just a complex switching device to an enormous and complicated analog studio hooked into a tape machine. And yet for me it was so wonderful because I could specify something and hear it instantly (Chadabe 1997).

Although the RCA synthesizers are more of historical than technological interest today, they introduced several important concepts into electronic music. First of all, the variable-speed paper tape drives—mechanically problematic though they were—separated control of the synthesizer from the physical limitations of human performers: no longer were the actions of the synthesizer limited by ten fingers, two feet, two hands, etc. Neither did the synthesizer produce ancillary sounds such as bow scratches, key clicks, or valves rattling. A musical trill consisting of precisely 32 (or 33 or 34) alternations per second could readily be obtained, and rhythms too complex for human performance could be precisely realized. This allowed some of the first methodical investigations into aspects of human perception that had been hitherto inaccessible. For example, what is the just noticeable difference (jnd) of trill speed? How many layers of complex polyrhythms can be distinguished using similar or contrasting tone qualities?

Finally, synthesizers can provide for at least a certain measure of real-time control, allowing them to be played like traditional musical instruments.

27.9 ■ VOLTAGE CONTROL

The next important technological advance in electronic music synthesizers was the concept of *voltage control*. Robert Moog, working near New York City, and Donald Buchla, working in San Francisco, built successful voltage-controlled synthesizers in the mid-1960s (see Fig. 27.5). Since they were based on transistorized modules, the Moog and Buchla synthesizers could be made much smaller, lighter, and cheaper than the vacuum-tube-based RCA synthesizers. This allowed voltage-controlled synthesizers to be successfully developed as consumer products.

The principal of voltage control is straightforward: *applying a voltage to a control input has the same effect as manually turning a knob*. Any of the elements of electronic music can, in principle, be voltage-controlled as well as manually controlled. For example, a voltage-controlled oscillator (VCO) is one that generates one or more waveforms with a frequency that is set by either a manually turned knob, a control voltage, or some combination of both. The control voltage can be the output of an electronic circuit such as an electronic keyboard, a knob-adjusted voltage source (such as a battery attached to a potentiometer), another oscillator, or any (mixed) combination of such voltages (see Fig. 27.6).

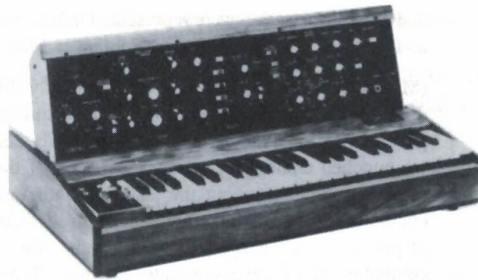
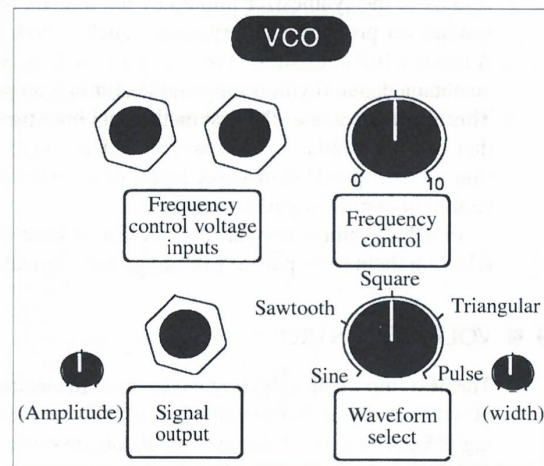


FIGURE 27.5 The Mini Moog Sonic Six (1974–1979). It included a 49-key keyboard and two VCO modules with sawtooth, triangular, and rectangular (pulse/square) outputs. Knobs adjusted such things as the width of the rectangular waveforms, portamento rates, and attack and decay times for the envelope generators. It also contained a VCF and two LFOs (low-frequency oscillators). It could produce as many as two notes at once.

FIGURE 27.6

A generic voltage-controlled oscillator (VCO) module. The frequency of the output signal is controlled by the sum of the control input voltage(s) and the setting of the frequency control knob. Other knobs set the output waveshape, the width of the pulse waveform, and the output signal amplitude.



Similarly, a voltage-controlled amplifier (VCA) is one whose gain is controlled by either a knob or a control voltage, a voltage-controlled band-pass filter (VCF) is one whose center frequency and bandwidth are both controlled by knobs or control voltages, and so on.

27.10 ■ MODULES AND PATCHES

A voltage-controlled synthesizer typically consists of several modules such as VCOs, VCAs, and VCFs, all of which are interconnectable. The strategy used for such interconnections may rely on cables or switches of various kinds—in any case, a given inter-

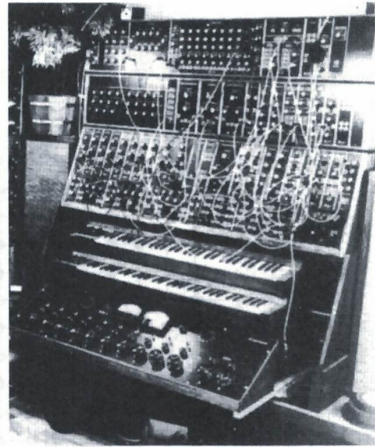


FIGURE 27.7
A forest of patch
cords interconnect
the modules of a
Moog synthesizer.

connection among modules is called a *patch* (no doubt named for the lightweight cables called *patch cords* designed specifically for such interconnections). Complex patches often required users to color-code their patch cords (see Fig. 27.7).

To combat the “forest of patch cords” problem, some analog synthesizers employ innovative interconnection methods, ranging from switches to slider pots to matrix pins (see Fig. 27.8). Although internal programming of interconnections tends to restrict the number of possible interconnections that can be made at one time, it also reduces programming time and improves reliability, making the synthesizer more practical for a live performance.

For example, by patching the output of one VCO module into the frequency control input of another, we can modulate the frequency of the second oscillator with the signal of the first (see Fig. 27.9). A more useful patch might add a modulating waveform to a source of constant voltage: the constant voltage would then set the center frequency, around which frequency modulation occurs. This type of frequency modulation (FM) could be used to produce a vibrato effect, with the vibrato rate and depth being determined by the frequency and amplitude of the modulating VCO.

Similarly, the signal output of one VCO may be added to the control voltage output of a keyboard in order to obtain a vibrato effect. The frequency and amplitude of the modulating VCO would then control the rate and depth of the vibrato (see Fig. 27.10).

Such patches may be combined in various ways. Figure 27.11 shows a patch that simultaneously produces both frequency and amplitude modulation using two VCOs and a VCA.

Another important module on most voltage-controlled synthesizers is the *ADSR envelope generator*. The ADSR (for attack-decay-sustain-release) envelope generator typically produces a control voltage that is intended to be applied to the control input of a VCA. Typical ADSR modules combine four knobs with four control inputs for setting each of the four segments of the envelope signal (see Fig. 27.12). In addition, a “trigger” input detects when a control voltage exceeds a preset value, thus initiating—or triggering—the

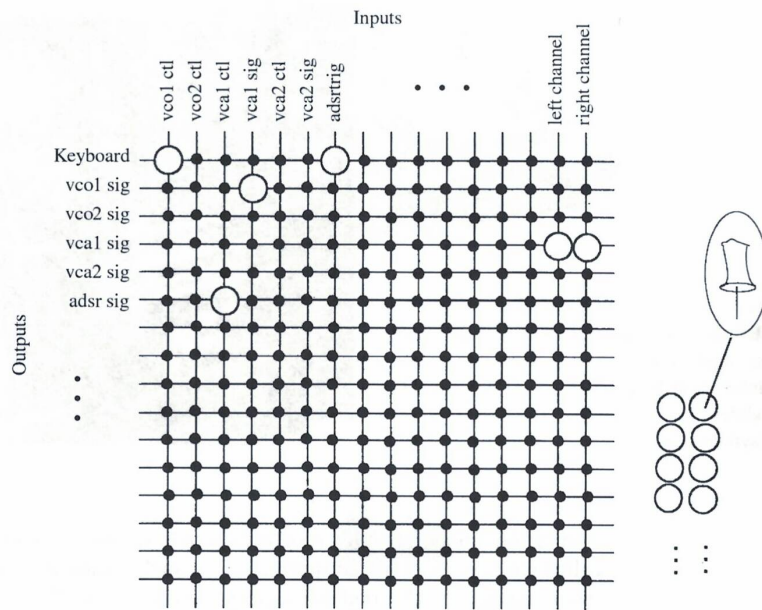


FIGURE 27.8 A pin matrix can be shown to connect an arbitrary number of available outputs (rows) to an arbitrary collection of signal and control inputs (columns). Placing a patch pin at a matrix intersection established electrical connection between the output row and input column. The patch shown connects the output of a keyboard to both a VCO and to the trigger input of an ADSR envelope generator. When a key is pressed, the triggered envelope uses a VCA to modulate the amplitude of the signal produced by the VCO. The result is placed into both output channels.

FIGURE 27.9

The signal output of a VCO 1 is patched into the frequency control input of VCO 2. Assuming both are set for sine waveforms, and that the frequency and amplitude of VCO 1 are small, the waveforms produced will resemble those shown.

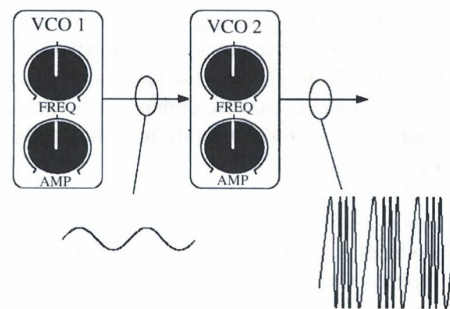
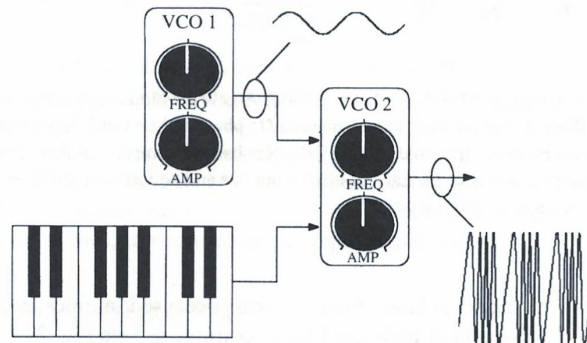


FIGURE 27.10

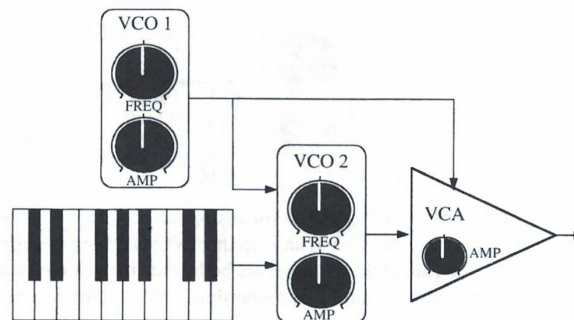
A vibrato patch: the frequency and amplitude of VCO 1 control the rate and depth of a vibrato around the pitch determined by the keyboard. Note that both control inputs and the frequency knob setting of VCO 2 are added together, further allowing this knob to be used as a pitch bend.



Voltage-controlled filters (VCFs) may be low-pass, high-pass, band-pass, or band-reject filters with parameters such as cutoff frequency, center frequency, and bandwidth controlled by knobs and/or control voltages. The principle use of a VCF is to control the *timbre* of a sound, which is often a complex waveform (such as a sawtooth waveform) from a VCO. A simple (and very effective) patch involving a sawtooth VCO, an ADSR envelope generator, and a VCA may be used to produce what on many modern synthe-

FIGURE 27.11

An AM/FM patch: VCO 1 controls the rate and depth of both the vibrato imparted by VCO 2 and the tremolo imparted by the VCA. The keyboard is patched into a second frequency-control input of VCO 2.



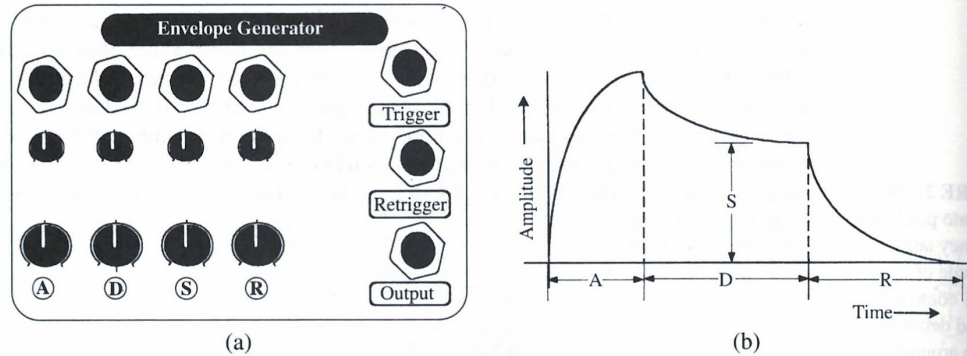


FIGURE 27.12 A typical ADSR envelope module. A new envelop is generated when the trigger input goes above a preset value. It passes through the attack (*A*) and decay (*D*) phases, then holds the sustain (*S*) level until the trigger goes below the preset level, which starts the final release (*R*). Knobs and control voltages determine how much time the *A*, *D*, and *R* segments take as well as the *S* level. Each control input has an attenuator; a retrigger input causes the envelope to immediately start again at the beginning of the attack phase.

sizers is labeled *analog brass* because it mimics an early analog synthesizer imitating the sound of traditional brass instruments (see Fig. 27.13). This patch is based on the finding of computer music pioneer Jean-Claude Risset that the bandwidth of a brasslike tone is proportional to its amplitude. It is the basis for a tone quality found commonly on later synthesizers and called *analog brass*.

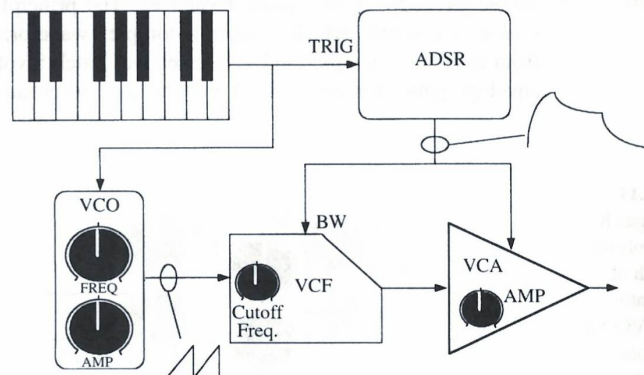


FIGURE 27.13 A patch for analog brass. A harmonic-rich (sawtooth) waveform is passed through both a filter (VCF) and an amplifier (VCA). Using voltage control, the bandwidth (BW) of the filter and the gain of the amplifier are both made to track the same ADSR envelope, causing the bandwidth of the output to shrink in proportion to the amplitude envelope, imparting a brass-like quality to the sound.

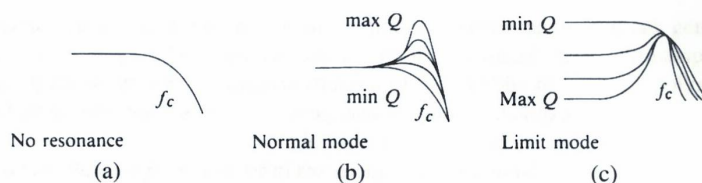


FIGURE 27.14 Characteristics of a voltage-controlled low-pass filter with cutoff frequency f_c , without resonance (a), with resonance (b), and with resonance in *limit mode* (c). Because these are filter responses, the horizontal axis is frequency and the vertical axis is amplitude.

Some VCFs include a feature that adjusts the resonance, or Q factor, of the filter at or near its cutoff frequency (Q is essentially center, or cutoff, frequency divided by bandwidth: thus a high- Q filter has relatively narrow bandwidth). In some VCFs, increasing Q increases the gain of the filter at its resonance frequency, whereas in others, the resonance gain remains the same and the gain at other frequencies is proportionally decreased, which is sometimes known as *limit mode* (see Fig. 27.14). A VCF with a high Q can “ring,” i.e., it can oscillate on its own, sometimes with no input signal at all.

27.11 ■ CONTROL VOLTAGE CHARACTERISTICS

Depending on the synthesizer, control voltages may or may not be interchangeable with waveform signals. In addition, the modules may respond to control voltages either linearly or exponentially (see Fig. 27.15). These variations of control methods represent the choices of the synthesizer designer, and there are advantages and disadvantages to whichever choices are made.

The principal advantage of exponential control voltages is that the responses of VCOs and VCAs follow Fechner’s hypothesis (see Section 5.6) that sensation is proportional to the logarithm of stimulus. With exponential response, a VCO, for example, responds with a certain number of semitones per volt. Thus, equal increases in control voltage result in

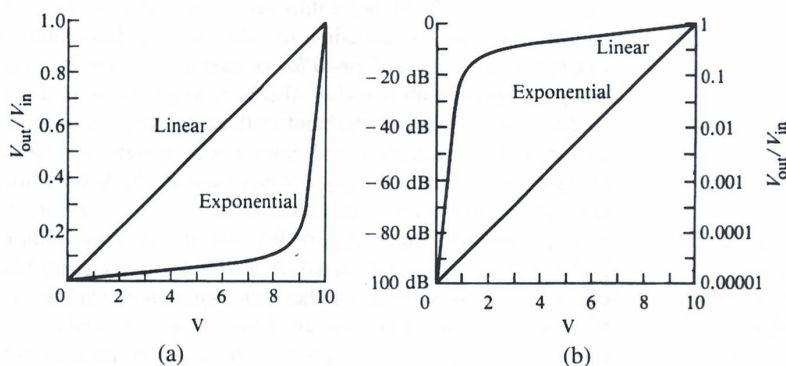


FIGURE 27.15 Linear and exponential control characteristics for a voltage-controlled amplifier (VCA) shown on (a) a linear graph and (b) a semilogarithmic graph.

equal ratios of frequency increases, i.e., equal musical intervals. Thus, if a VCO increases its frequency by one octave per volt, adding $\frac{1}{12}$ V will increase the pitch by a semitone, and adding 1 V to a control signal will transpose all pitches up by one octave. Such an exponential control voltage is sometimes more convenient for musical applications because it relates directly to musical pitch.

Similarly, equal increases in control voltage result in equal ratios of increase in amplifier gain for an exponentially responding VCA, i.e., equal increases on a decibel scale. Such a VCA might vary its gain by 10 dB/V, for example, which readily translates into 1 dB per tenth of a volt, etc.

Because waveform signals are inherently linear, the primary disadvantage of exponential control voltage response is that signals and control voltages typically may not be mixed. Adding the output of a low-frequency VCO with a sinusoidal waveform and amplitude of $\frac{1}{12}$ V to a constant voltage will produce a vibrato of ± 1 semitone if the controlled VCO responds exponentially at 1 V per octave. If true frequency modulation is desired, it may be necessary to pass one or more signals through a conversion module designed to translate between linear and exponential scales.

27.12 ■ OTHER MODULES

In addition to those already mentioned, voltage-controlled synthesizers may contain other modules that perform other functions. Most synthesizers contain noise sources such as white-noise or pink-noise modules. A *white-noise generator* produces a randomly fluctuating signal in which all frequencies are present in more or less equal proportion. By analogy with white light, then, which contains equal proportions of all visible frequencies of light, a white-noise generator produces a sound that is rather high pitched due to the uneven response of the ear to different frequencies. Especially at low amplitudes, white noise sounds to the ear like the inverse of the equal-loudness curves (see Fig. 6.4).

A *pink-noise generator* produces a randomly fluctuating signal that sounds more like all frequencies to the ear. This is typically achieved by filtering white noise so that low frequencies are emphasized. Instead of having equal energy at all frequencies, as does white noise, pink noise typically has equal energy per octave. Because octaves occupy increasing amounts of frequency bandwidth as they go up in pitch, the energy density (per unit frequency) of pink noise thus decreases with increasing frequency.

Many synthesizers contain a *reverberation module* similar to the spring unit described in Chapter 26. An *envelope-follower module* measures the average amplitude of a signal and generates a control voltage that is proportional to it. Thus the gain of a VCA (or the frequency of a VCO, or the bandwidth of a VCF) can be made to “follow” the amplitude envelope of an arbitrary signal. Such a control signal can then impart some of the quality of the input sound to virtually any other sound (this is the source of many “talking” effects that can be superimposed on almost any sound).

A particularly interesting module called a *ring modulator* multiplies two signals together to form its output signal. A *modulator* is a circuit that modifies something about one signal in response to another. An *amplitude modulator*, for instance, uses one signal to control—or modulate—the amplitude of another signal (we will also discuss *frequency modulation* in some detail later on). An amplifier, for example, multiplies its input signal

by a positive value called the *gain* of the amplifier. Because the signal can generally be either positive or negative but the gain is always positive, we can think of an amplifier as a *two-quadrant* amplitude modulator, because it must accommodate only the following two cases of multiplication.

1. Input signal is +, gain is +, output is +.
2. Input signal is -, gain is +, output is -.

Because a ring modulator multiplies two *arbitrary* signals, either one of which can be positive or negative at any time, it must accomplish *four-quadrant* amplitude modulation according to the following:

1. Input 1 is +, input 2 is +, output is +.
2. Input 1 is +, input 2 is -, output is -.
3. Input 1 is -, input 2 is -, output is +.
4. Input 1 is -, input 2 is +, output is -.

The ring modulator gets its name from the fact that the analog circuit that accomplishes four-quadrant multiplication is traditionally drawn with its four multiplying parts lying on a circle.

Multiplying one signal by another in this manner produces many interesting results. Essentially, the output signal consists of frequencies located at all possible sums and differences of frequencies taken from the two input signals. As a simple example, if input 1 is a complex waveform containing frequencies at 100 Hz and 200 Hz, and input 2 is a sinusoid at 10 Hz, the output would consist of a signal containing the four frequencies 100 ± 10 Hz and 200 ± 10 Hz. More complex inputs can produce dazzlingly complex outputs that generally sound distorted in ways ranging from pleasant to disturbing.

As explained in Section 8.4, alternating constructive and destructive interference between sinusoids at 100 Hz and 102 Hz will give rise to a beat frequency of 2 Hz. Figure 8.6 also shows that the frequency of the resulting waveform is the average of the two frequencies, in this case, 101 Hz. The beating waveform can be thought of either as resulting from adding the 100 Hz and 102 Hz sinusoids or from multiplying the 101 Hz sinusoid by one of 1 Hz (which way we hear it is determined by the size of the frequency difference).

Another way to understand the equivalence of adding and multiplying sinusoids is to observe that this is not just a physical or perceptual phenomenon but an inherent property of the sinusoids themselves. We learned in Section 2.1 that sinusoids represent basic vibration patterns called simple harmonic motion. A sinusoid always consists of just one frequency—any other vibration pattern is the result of multiple frequencies.

A sinusoid is any curve that is identical in shape to the mathematical sine or cosine functions (the sine and cosine functions themselves have the same basic shape, only different starting points). It is possible to demonstrate many interesting things about sinusoids by using their basic mathematical properties (see Appendix A.10). By combining the basic definitions of sine and cosine with the Pythagorean theorem,

for example, we can show that

$$\cos A + \cos B = 2 \cos \left(\frac{A+B}{2} \right) \cos \left(\frac{A-B}{2} \right) \quad (27.1)$$

where A and B are any angles. Such relationships are called *trigonometric identities*, because they show two different ways to express the same thing. If we let $A = 2\pi f_A t$ and $B = 2\pi f_B t$, $\cos A$ and $\cos B$ can equally well represent two sinusoids at frequencies f_A and f_B . The left side of Eq. (27.1) then represents the sum of the two sinusoids. The right side correspondingly represents one sinusoid at the average frequency, $(A+B)/2$, multiplied by another at one-half of the difference frequency, $(A-B)/2$. Beats occur at twice the latter frequency because a sinusoidal waveform passes through both its positive and negative peaks during each cycle, resulting in two places of maximum amplitude in the product waveform. There is also a factor of 2 multiplying the right side of the equation because the peak amplitude of the left side is twice that of a single sinusoid whenever both peaks add in the same direction (up or down).

Another trigonometric identity that says the same thing in a different way is

$$\cos A \cos B = \frac{1}{2} [\cos(A-B) + \cos(A+B)] \quad (27.2)$$

27.13 ■ OTHER CONTROLLERS

In addition to the many types of voltage-controlled modules found in analog synthesizers, Donald Buchla invented an analog automation device called a *sequencer* that allowed a series of control voltages to be specified and then played back in sequence. Initially an attempt to reduce the labor involved in splicing tapes, the sequencer allowed virtually any melody, loudness sequence, or other series of sounds to be played back automatically. The main restriction was that the length of the sequence could not exceed the number of stages in the sequencer (originally 8 or 16 in a Buchla synthesizer). Clever application of the sequencer makes it possible to automate the playing of a nonrepeating series of sounds as well (listen, for example, to composer Morton Subotnick's landmark composition *Silver Apples of the Moon*, which was realized on a sequencer-equipped Buchla synthesizer in 1967).

27.14 ■ DIGITAL CONTROL AND THE GROOVE SYSTEM

As analog voltage-controlled synthesizers grew in capabilities and complexity (see Fig. 27.16), the problems of rapidly manipulating the interconnections of increasing numbers of signal and control modules grew as well. By the late 1960s, digital computers had advanced to the point where they could be used to generate some or all of the signals needed to control an analog synthesizer, which gave rise to the *hybrid synthesizer*. In a hybrid synthesizer all the waveforms were typically analog, but some or all of the control signals are generated using a digital computer.

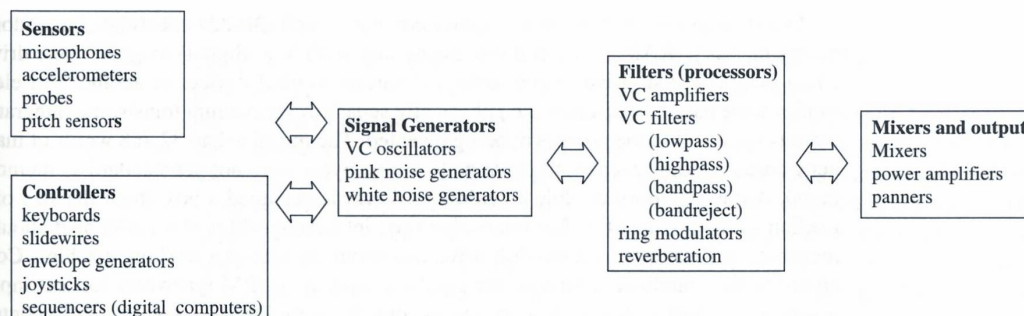
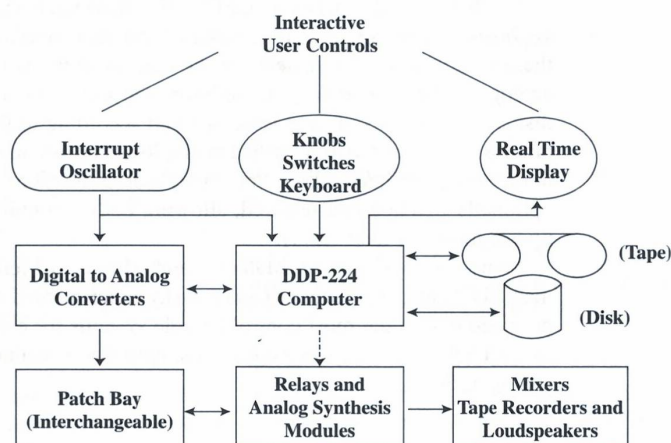


FIGURE 27.16 The “assembly line” for sound in an analog electronic music synthesizer. Compare the modules to the basic elements of electronic music mentioned in Fig. 27.1.

Max Mathews and F. R. Moore created such a system in the late 1960s at Bell Laboratories. It was called *GROOVE*, for Generated Realtime Operations On Voltage-controlled Equipment. It consisted of a Honeywell DDP-224 computer attached to a bank of 14 DACs, as well as a bank of computer-controlled relays and a variety of analog synthesizer modules (see Fig. 27.17). Some of the analog modules were commercially produced by Moog, ARP, and other companies; many were custom designs. An interchangeable “patch bay” allowed users to rapidly reconfigure interconnections between the DACs and the analog modules, as well as among the modules themselves. Each SPDT (single-pole, double-throw) relay appeared on the patch bay as well, allowing some of the interconnections to be controlled directly and dynamically by the computer (which ones depended on how a user configured the patch panel in use).

FIGURE 27.17 Schematic diagram of the *GROOVE* system, ca. 1969. In response to user inputs, including interactive controls and a computer program, the computer updates digitally stored time functions, a bank of DACs, and SPDT relays at a rate determined by an interrupt oscillator. A user-supplied patch bay determines how these signals affect an extensive bank of analog synthesis equipment.



As a state-of-the-art, late 1960s-vintage computer, the DDP-224 consisted of a random-access memory (RAM), a central processing unit (CPU), a (digital) magnetic tape drive, and a magnetic disk drive in one room and various control devices in another. All electronics were based on *discrete* (i.e., physically separate) germanium transistors; integrated circuits were still in the process of being invented. The machine had 32,768 words of magnetic core memory (each word contained 24 bits; bytes were not yet standard as memory units). A core memory module containing 4K words occupied a box about the size of a medium-sized refrigerator. The removable (i.e., interchangeable) disk packs held about 1 megabyte of data each, and the disk drive was about the size of a washing machine. Connected to the computer were a vector graphics display, an IBM typewriter for text input, several knobs and switches whose positions could be sensed by the computer, and a control panel consisting of blinking lights showing the current state of all machine registers. Any bit in any register could be set or reset by pressing on the corresponding control panel light, thus defining the starting point for low-level programming.

The GROOVE software system consisted of a set of programs written in FORTRAN and assembly language that ran on the DDP-224 computer in real time, the speed of which was determined by an electronic oscillator the computer could sense through *interrupt* signals. Each time an interrupt signal occurred (about 100–200 times a second), the computer would execute a *foreground* program that necessarily had to accomplish everything in less than about 5 or 10 ms. Whenever the foreground program was not running, a *background* program stored in the computer memory would run.

The background task of GROOVE was to update the display viewed by the user, which showed time marching from left to right across the screen. The vertical axis was divided into several regions, like staves in a musical score. Each region displayed the graph of one function of time, whose significance was determined by the user. One function of time might control the frequency of a VCO, another the gain of a VCA, another the bandwidth of a VCF, etc., depending on the patch used. Values of each time function changed at the interrupt oscillator rate—about 100 times per second (this rate could be manually adjusted by the user).

The foreground program of GROOVE sensed the positions of all inputs, such as knobs, keyboards, joysticks, etc., and executed the user-supplied program that determined how these data were used to determine the outputs to the DACs (which in turn controlled the analog synthesis modules). In addition, the user-supplied program determined how the displayed functions of time were updated according to the input data and various signal-processing tactics the user wished to employ, such as combining keyboard data with a knob to create a pitch bend effect. Previously stored functions of time were read from disk and optionally updated and restored, allowing both a nonbinding rehearsal and a permanent update mode for later replay.

Many composers from Mathews and Moore to Pierre Boulez, Laurie Spiegel, Emmanuel Ghent, and Vladimir Ussachevsky experimented with the GROOVE system, which provided one of the most control flexible systems for electronic music ever devised. But, as the DDP-224 computer exemplified, electronics technology had begun to change from analog to digital.

27.15 ■ SUMMARY

Electronics has transformed music into a very general art of sound. Electronic music itself grew out of advances in electronics and sound-recording technology combined with acoustical principles and new musical concepts such as serialism. One major school of thought was *musique concrète*, which was based on creating a new kind of music that relied on combining and manipulating natural sounds instead of traditional musical instruments. A second major school of thought was *elektronische musik*, in which arbitrarily complex sounds could be synthesized from simple, basic components. Both these approaches allowed exploration of sounds that had never before been accessible to musicians and listeners. Work in both of these increasingly nondistinct areas was originally hampered by available technology, which made the manipulation of sounds extremely expensive and labor intensive. Early experiments in electronic music nevertheless provided an augmented conceptual basis for later developments in electronic music that were based on increasingly inexpensive and powerful solid state electronics (i.e., transistors). The first primary result was the analog synthesizer, which capitalized on modular design principles, the principle of voltage control, innovative interconnection schemes such as patch cords and pin matrices, increasingly sophisticated control devices such as ADSR envelope generators and sequencers, and, ultimately, control by a digital computer.

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GLOSSARY

ADSR envelope generator *A*, *D*, *S*, and *R* refer to the parameters of an envelope: *A* = attack time; *D* = (initial) decay time; *S* = sustain level; *R* = (final) release time.

amplitude envelope The manner in which peak or average amplitude varies with time; the envelope is comprised of the attack and decay of a tone, among other things.

elektronische musik A type of music based on the control of many electronically generated signals that together form a musical result.

envelope follower A circuit that generates a control voltage that duplicates the amplitude envelope of a signal taken from an acoustical musical instrument or other sound source.

exponential A function or characteristic that varies as e^x . In a typical exponential control characteristic, an increase of 1 V doubles some parameter such as frequency.

hybrid synthesizer Use of a digital computer to control the analog circuitry in a synthesizer; also a synthesizer that combines analog and digital generators or modules.

linear A function or characteristic that varies in direct proportion to a control signal. In a typical linear control characteristic, an increase of 1 V adds a fixed amount to some parameter such as frequency.

modulator A device that controls (modulates) a characteristic of one signal (amplitude, frequency, etc.) with another signal.

musique concrète A type of music based on manipulating the “concrete” sounds of nature (rather than the “abstract” sounds of traditional musical instruments).

panning Applying a set of output signals to a series of loudspeakers sequentially so that the sound seems to move in space.

patch (Verb) to interconnect; or (noun) a set of interconnections that causes a synthesizer to produce certain types of sounds.

pink noise Low-pass-filtered random noise for which the energy contained in each octave band is the same.

pitch bend A control that lowers or raises an otherwise fixed pitch (such as one specified by a note on a keyboard).

Q A filter parameter that specifies the sharpness of a resonance (technically, ratio of energy stored to energy dissipated per cycle).

ring modulator A circuit that multiplies two input signals, thereby forming an output signal containing the sums and differences of all pairs of frequencies present in the two input signals.

sequencer A circuit that switches through a predetermined sequence of control signals.

synthesizer An instrument that creates complex sounds by generating, altering, and combining various electrical waveforms, typically by means of voltage-controlled modules.

VCA (voltage-controlled amplifier) An amplifier whose gain varies linearly or exponentially in proportion to a control voltage.

VCF (voltage-controlled filter) A filter whose cutoff frequency (or center frequency and bandwidth) varies linearly or exponentially in proportion to one or more control voltages.

VCO (voltage-controlled oscillator) An oscillator whose frequency varies linearly or exponentially in proportion to one or more control voltages.

REVIEW QUESTIONS

1. What is the basic concept of *musique concrète*?
2. How were sounds first recorded in *musique concrète*?
3. Name three advantages of magnetic tape recording for sound manipulation.
4. What was the fundamental concept of *elektronische musik*?
5. What is a vocoder?
6. How were the original concepts of *elektronische musik* linked to *serialism*?
7. Why were *musique concrète* and *elektronische musik* originally so tedious to produce?
8. Where was the first major electronic music synthesizer located?
9. How did the RCA Mark II avoid the limitations of human performance?
10. What is the basic principle of voltage control?
11. What is a VCO? A VCA? A VCF? An LFO? An ADSR generator?
12. What is a patch?
13. What is a sequencer?
14. How does frequency modulation work?
15. How does one synthesize analog brass?
16. What are the advantages and disadvantages of exponential control voltages?
17. Which sound has a higher pitch, white noise or pink noise? Why?
18. Describe the operation of a ring modulator.
19. What is a hybrid synthesizer?
20. What did functions of time control in the GROOVE system?

QUESTIONS FOR THOUGHT AND DISCUSSION

1. What are the principal advantages and disadvantages of *musique concrète* and *elektronische musik*?
2. In what ways do synthesizers replace traditional musical instruments or performers?
3. What kinds of music, if any, can be made electronically that cannot be made with traditional instruments?
4. About how much of the music heard today is produced electronically?
5. What is the purpose of the limit mode shown in Fig. 27.14?
6. What effect does turning the “width” knob have on the resulting sound for the VCO shown in Fig. 27.6?

EXERCISES

1. The control characteristic of a certain VCO is stated as 1 V per octave.
 - (a) Is this a linear or exponential characteristic?
 - (b) If a control voltage of 3 V causes it to oscillate at $C_3(f = 131 \text{ Hz})$, what voltage will be required to give $G_3(f = 196 \text{ Hz})$?
2. If one input to a ring modulator contained two frequency components at $f_1 = 200 \text{ Hz}$ and $f_2 = 350 \text{ Hz}$ and the other input was a sinusoid at $f = 50 \text{ Hz}$, what frequencies would be present at the output? What would you expect the pitch of the resulting sound to be?
3. Make a drawing similar to Fig. 27.13 showing what patch is represented by the pin matrix shown in Fig. 27.8.
4. Make a drawing (to scale) of an ADSR envelope with a maximum amplitude of 2 V and the following parameters: $A = 50 \text{ ms}$, $D = 20 \text{ ms}$, $S = 1.5 \text{ V}$, $R = 100 \text{ ms}$.

EXPERIMENTS FOR HOME, LABORATORY, AND CLASSROOM DEMONSTRATIONS

Home and Classroom Demonstrations

1. *Musique concrète* Using a portable tape recorder collect a variety of sounds from your environment. Using a home or laboratory computer, edit and combine these sounds into an interesting collage of arbitrary length.
2. *Elektronische Musik* Tape record sounds produced by electronic oscillators and noise generators, filtering them before or after recording as desired. Using a home or laboratory computer, edit and combine these sounds into an interesting collage of arbitrary length.
3. *Electronic music* Using your local library and/or other sources, find as many recorded examples of electronic music made before about 1955 as you can. Also find as many examples made within the past five years as you can. Listen to these examples and compare their features.
4. *Analog music synthesizers* Obtain the oldest working analog music synthesizer you can and experiment with its capabilities.

Laboratory Experiments

Electronic Music Synthesizers (Experiment 38 in *Acoustics Laboratory Experiments*).



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—Dan Boye, Davidson College

"... remains true to the interdisciplinary nature of acoustics... the book provides a wonderful outline for my course and... an invaluable resource for both me and my students."

—Anderson Mills, University of Texas, Austin

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