

THE SCIENCE OF
SOUND
THIRD EDITION



Rossing
Moore & Wheeler

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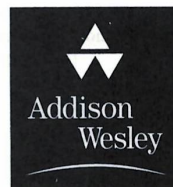
THE SCIENCE OF SOUND

THIRD EDITION

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San Francisco Boston New York
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Acquisitions Editor: Adam Black
Project Editor: Nancy Gee
Cover Designer: Blakeley Kim
Marketing Manager: Christy Lawrence
Managing Editor: Joan Marsh
Manufacturing Buyer: Vivian McDougal
Project Coordination: Integre Technical Publishing Co., Inc.

Library of Congress Cataloging-in-Publication Data

Rossing, Thomas D., 1929–

The science of sound.—3rd ed. / Thomas D. Rossing, Paul Wheeler, and Richard Moore.

p. cm.

Includes bibliographical references.

ISBN 0-8053-8565-7

1. Sound. 2. Music—Acoustics and physics. I. Wheeler, Paul (Paul A.) II. Moore, F. Richard.

III. Title.

QC225.15 .R67 2002

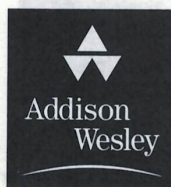
534—dc21

2001053994

ISBN 0-8053-8565-7

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CHAPTER

20

Microphones, Amplifiers, and Tuners

Chapter 19 focused on loudspeakers, which are output transducers. We now turn our attention to input transducers (microphones), which convert acoustic waves into electrical signals. Microphones are essential components in sound reinforcement and sound-recording systems. Once an electrical signal is obtained from a microphone, the signal can be amplified, modified, mixed with other signals, and then returned to listeners (sound-reinforcement system) or stored for future playback (sound-recording systems).

A home entertainment system selects previously recorded sounds from several media (compact disc, tape, etc.) along with broadcast sound (radio), which is amplified and delivered to two or more loudspeakers. A tuner is the component in a home entertainment system that receives broadcast signals (tunes them in) and passes them on to the amplifier. Often the tuner is built into the same box with the amplifier.

This chapter discusses microphones, amplifiers, and tuners used in sound recording, sound reinforcement, and home entertainment.

In this chapter you will learn:

- About the physical characteristics of microphones;
- About the electrical characteristics of microphones;
- About microphone sensitivity;
- About the characteristics of audio amplifiers;
- About distortion in audio amplifiers;
- About the properties of AM/FM tuners;
- About surround-sound receivers.

20.1 ■ MICROPHONES AS TRANSDUCERS

A microphone is a transducer that produces an electrical signal when actuated by sound waves. Microphones may be designed to respond to variations in air pressure due to the sound wave or to variations in particle velocity as the sound wave propagates. Most microphones in common use are *pressure* microphones.

A pressure microphone has a thin diaphragm, which moves back and forth with the rapid pressure changes in a sound wave. The diaphragm is connected to some type of electrical generator, which may be a piezoelectric crystal (crystal microphone), a moving coil (dynamic or magnetic microphone), or a variable capacitor (condenser microphone). Each of these microphone types will be discussed briefly.

One can also classify microphones according to their ability to pick up sound arriving from different directions. Microphones may be omnidirectional (that is, nearly the same sensitivity in all directions), unidirectional (that is, considerably more sensitive in one direction), or bidirectional in their response pattern. The directionality of a microphone is usually determined by the acoustical design of its housing.

Crystal Microphones

Crystal microphones (see Fig. 20.1) use piezoelectric crystals that generate a voltage when deformed by a mechanical pressure. Rochelle salt crystals generate a large electric signal and are used in some inexpensive microphones. Better-quality crystal microphones generally use a ceramic material, however, which is much less sensitive to humidity, temperature, and mechanical shock. Ceramics commonly used include barium titanate, lead titanate, and lead zirconate. The piezoelectric effect is described in the following box.

Crystal microphones have a comparatively large electrical output voltage, which makes them convenient for use in portable sound equipment and tape recorders. On the other hand, the high-frequency response of most crystal microphones is less "flat" than that of condenser microphones due in part to the greater mass that must move. Therefore, they are not often used in the professional recording of sound.

The *piezoelectric* (pronounced "pah-ee-zo-electric") effect was discovered in 1880 by the brothers Pierre and Jacques Curie, who observed that certain crystals acquire electric charge when they are compressed or distorted. The amount of charge that appears is proportional to the distortion and disappears when the force is removed. If leads are attached to the crystal at the appropriate places, an electrical output signal is obtained.

The piezoelectric effect is reversible; if an electrical voltage is applied to a piezoelectric crystal, it changes its shape. This has led to the use of piezoelectric crystals as ultrasonic generators and even as loudspeakers for high-frequency sound. Quartz, tourmaline, topaz, and Rochelle salt are examples of natural piezoelectric crystals.

FIGURE 20.1

A crystal microphone. Sound pressure on the diaphragm causes a deformation of the crystal, generating an electrical signal.

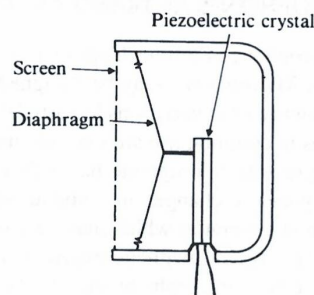


FIGURE 20.2

A dynamic (moving coil) microphone. Sound pressure on the diaphragm causes the voice coil to move in the magnetic field.

FIGURE 20.3

A simple capacitor (condenser) microphone. It consists of two parallel conducting plates connected to a battery. One plate acquires a positive charge, the other a negative charge. The amount of charge changes as the spacing between the plates changes.

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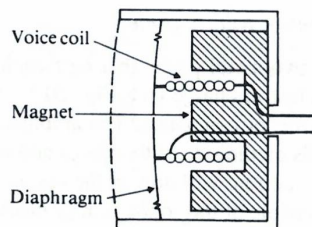
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FIGURE 20.2
A dynamic (moving coil) microphone. Sound pressure on the diaphragm causes the voice coil to move in a magnetic field.



Dynamic Microphones

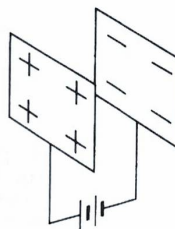
In a *dynamic* or magnetic microphone, an electrical signal is generated by the motion of a conductor in a magnetic field. This is an example of *electromagnetic induction*, or the dynamo principle, discussed in Section 19.3. In the most common type of dynamic microphone, sound pressure on a diaphragm causes the attached voice coil to move in the field of a magnet, as shown in Fig. 20.2. Movement of the voice coil thus generates an output voltage.

The electrical voltage generated by a dynamic microphone is quite small, but the impedance of the voice coil is also small. Because of the low source impedance, the power generated by a dynamic microphone is not necessarily small, even though the voltage output is small. The low source impedance is a distinct advantage when the microphone is located a substantial distance from the amplifier, as will be discussed later (see Section 20.3). If a higher output voltage is desired, a transformer can be used, preferably at the amplifier end of the transmission line. It is essential that the mass of the moving coil be kept very small in order to yield good response at high frequency.

A dynamic microphone is similar in principle to a small loudspeaker, and, indeed, a loudspeaker will serve as a low-quality microphone. This is frequently done in two-way intercommunication units. The mass of a loudspeaker cone plus voice coil is too large to give satisfactory response at high frequency, however, and so its fidelity as a microphone is generally low.

Dynamic microphones are rugged, have a broad frequency response, and are able to withstand the high sound levels that occur in popular music. Thus they are widely used in both live performance and recording.

FIGURE 20.3
A simple capacitor (condenser) consisting of two conducting plates connected to a battery. One plate acquires a positive charge, the other a negative charge. The amount of charge changes as the spacing between the plates changes.



Condenser Microphones

When two metal plates that face each other are connected to a battery, they acquire and store electric charge (see Fig. 20.3). Such an arrangement of charged plates is called a *capacitor*, or *condenser*. The amount of electric charge that can be stored in a condenser depends on the size of the plates and on their spacing. Thus, if one of the plates moves, an electric current will flow in the circuit as charge arrives and departs from the plate. This is the principle of the condenser, or electrostatic, microphone.

In a condenser microphone, one plate is usually the thin movable diaphragm, and the other plate is the fixed backing plate, as shown in Fig. 20.4. As the diaphragm is moved by the pressure of a sound wave, a small current flows in the circuit.

A condenser microphone has a very high source impedance, and for this reason, a preamplifier is usually incorporated into the microphone itself. The diaphragm can be made very light; therefore, the condenser microphone is capable of excellent response at high frequencies. A major disadvantage is the need for a relatively high voltage source to maintain electrical charge (bias) on the plates of the capacitor.

Electret-Condenser Microphones

A type of microphone that retains most of the advantages of the condenser microphone but eliminates the need for a high-voltage bias supply is the *foil-electret*, or *electret-condenser*, microphone, shown in Fig. 20.5. The diaphragm consists of a plastic foil less than one-thousandth of an inch thick with an even thinner layer of metal attached. The foil has been given a permanent electrical charge by a combination of heat and high voltage or by electron bombardment during manufacture.

FIGURE 20.4

A condenser microphone. The diaphragm constitutes one plate of a capacitor and it moves with sound pressure.

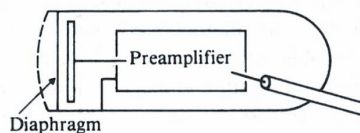


FIGURE 20.5

An electret condenser microphone. A thin metalized plastic diaphragm is tightly stretched across a perforated backing plate. The holes in the back plate couple to an air cavity.

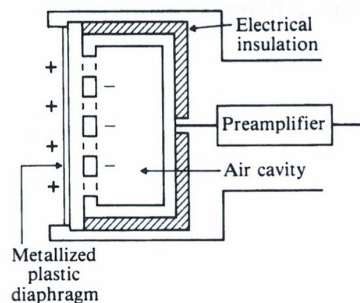
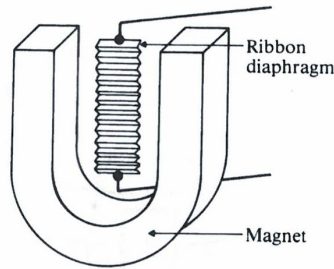


FIGURE 20.5
A ribbon microphone. A lightweight diaphragm is attached to a coil of wire, thus generating an electric current.

FIGURE 20.6

A ribbon microphone. A lightweight ribbon diaphragm moves in a magnetic field, thus generating an electrical signal.



Electret-condenser microphones often incorporate a built-in preamplifier that is usually a field-effect transistor. Because of their high input impedance, field-effect transistors are a better match for condenser microphones than the junction transistors we discussed in Chapter 18 (see Appendix A.9).

Ribbon Microphones

A *ribbon* microphone is a magnetic microphone in which the lightweight ribbon diaphragm is also the moving conductor (see Fig. 20.6). The ribbon responds to the acoustic velocity (the speed at which air particles are moving) rather than the pressure of the sound wave. A ribbon microphone responds readily to sound waves arriving from the front or back but is insensitive to sound arriving from the sides. Thus, it has a bidirectional response.

Ribbon microphones were popular in the early days of radio broadcasting, because their bidirectional nature allowed performers to stand on opposite sides facing each other. They are sensitive to moving air currents, however, and therefore are not satisfactory for outdoor use. Their use is now mainly restricted to special situations, such as a vocal soloist who wants to sing very close to the microphone and minimize the danger of “popping” when plosive consonants (p, t, d, etc.) are sung.

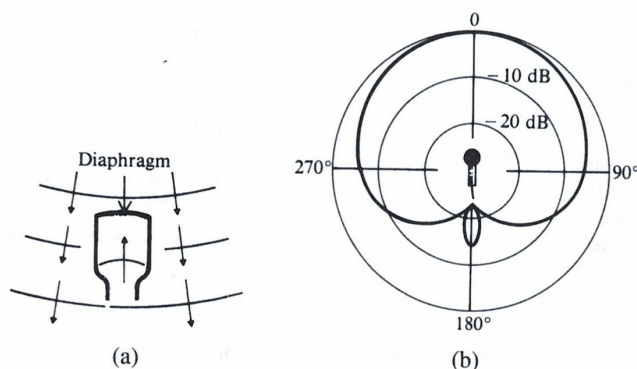
20.2 ■ MICROPHONE PICKUP PATTERNS

The physical construction of a microphone determines its directivity, or pickup pattern. We saw an example in the ribbon microphone, which is bidirectional. Another common pattern is the omnidirectional (omni) microphone, which is sensitive to sound coming from all directions. Omnidirectional microphones are often used for recording and stage plays to pick up sound from performers spread across a wider area.

In many applications, it is desirable to have a microphone with maximum sensitivity in one direction only (unidirectional). One such application is in sound systems in which acoustic feedback may lead to oscillation if the microphone picks up sound from the loudspeaker.

The most popular unidirectional microphone is the *cardioid* microphone, which has a heart-shaped pickup pattern (see Fig. 20.7b). A cardioid microphone can be made by adding a hole in the rear of the case of a sealed omnidirectional microphone. When a sound wave arrives from the rear, the pressure maxima and minima reach both front and

FIGURE 20.7
A simple cardioid
microphone (a) and
its pickup
pattern (b).



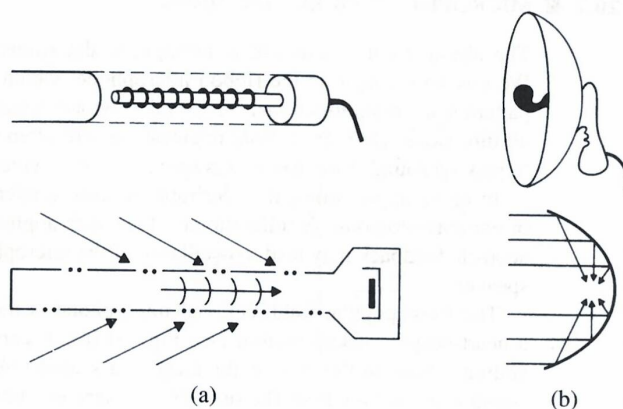
back of the diaphragm at the same time, and its deflection is thus minimal. A sound wave arriving from the front produces normal deflection, however.

It is relatively difficult to obtain a cardioid response pattern over a wide range of frequencies; thus, many cardioid microphones have an uneven frequency response, especially a roll-off at bass frequencies. Off-axis response, in particular, will be colored by a variation in the directional response with frequency. This is offset, however, by the unidirectional characteristic of the cardioid microphone, which reduces the pickup of background noise.

For even greater directivity, line microphones or reflector microphones can be used. The line microphone shown in Fig. 20.8(a) (sometimes called a "shotgun" microphone) has a tube with side holes through which sound can enter. Sound waves approaching from nearly head-on will leak into the tube at each side hole, building up into a traveling wave within. Sound waves from other directions, however, will enter the tube in varying phases through the side holes, and these contributions will tend to cancel each other.

A reflector microphone consists of a microphone element at the focus of a parabolic reflector, as shown in Fig. 20.8(b). Parabolic reflectors are also used in spotlights and tele-

FIGURE 20.8
Directional
microphones:
(a) line
microphone;
(b) reflector
microphone.



scopes; they are shaped so that all the sound (or light) arriving from straight ahead is reflected toward the focus. A reflector microphone has even greater directivity than a line microphone does.

20.3 ■ MICROPHONE IMPEDANCE

Microphones are generally classified as having *high impedance* or *low impedance* (referring to the electrical impedance of the output of the microphone). Crystal, condenser, and electret-condenser microphones have high output impedances from 50,000 ohms and up. Dynamic microphones, on the other hand, have low impedances, typically from 50 to 600 Ω .

If a high-impedance microphone is used with a long microphone cable, high-frequency signals will be attenuated (a 20-ft cable, for example, may result in a 6-dB loss at 10,000 Hz). This loss may be prevented by using a line-matching transformer or by locating a preamplifier near the microphone. Most commercial condenser microphones include a built-in transformer, resulting in a low output impedance. Low-impedance microphones can be used with long cables without much loss of signal at high frequency. Low-impedance (600- Ω) cables are frequently used in sound systems.

20.4 ■ MICROPHONE SENSITIVITY

There are basically two ways to express the sensitivity of a microphone: in terms of voltage output or in terms of power output for a given sound pressure. Sensitivity is measured by placing the microphone in a 94-dB sound field (sound pressure = 1 Pa) and measuring the open-circuit voltage on the output. The voltage sensitivity, S_V , is expressed in decibels (dBV):

$$S_V(\text{dBV}) = 20 \log V/p, \quad (20.1)$$

where V is the open-circuit output voltage and p is the sound pressure in pascals. (Note that 0 dBV = 1V/1 Pa). A sound field of 74 dB (1 microbar = 0.1 Pa) was used in the past to measure microphone sensitivity. If you encounter an old sensitivity rating (given in volts/microbar), add 20 dB to determine the actual voltage sensitivity.

Output-power sensitivity, S_P (in dBm, where m means that 0 dB is at 1 mW), is used in professional sound reinforcement. It is calculated from the voltage sensitivity and the microphone impedance:

$$S_P(\text{dBm}) = S_V - 10 \log Z + 24, \quad (20.2)$$

where Z is the electrical output impedance of the microphone. An alternative power sensitivity rating G_m , recommended by the Electrical Industries Association (EIA), gives the output power level for a 0-dB sound field as

$$G_m(\text{dBm}) = S_P - 94. \quad (20.3)$$

Most microphone manufacturers provide specifications for the open-circuit voltage sensitivity and the output impedance. Some manufacturers include the output-power sensitiv-

ity, whereas only a few manufacturers give the EIA sensitivity. To compare the sensitivities of two microphones, sensitivity ratings found in the specification may be converted to open-circuit voltage sensitivities and then compared.

Of primary interest to the microphone user is the output voltage that can be expected at a given sound level. The open-circuit output voltage is expressed by

$$20 \log V = S_V + L_P - 94, \quad (20.4)$$

where V is the open-circuit voltage, S_V is the open-circuit voltage sensitivity, and L_P is the sound pressure level. The following example illustrates how to compute output voltage.

EXAMPLE 20.1 A microphone has a sensitivity listed as 1.2 mV/Pa and an output impedance of 200 Ω .

- (a) What is the open-circuit voltage sensitivity in dBV?
- (b) What is the power sensitivity in dBm?
- (c) What is the EIA sensitivity in dBm?

Solution

- (a) $S_V(\text{dBV}) = 20 \log V/p = 20 \log(1.2\text{mV}/1 \text{ Pa}) = -58 \text{ dBV}$
- (b) $S_P(\text{dBm}) = S_V - 10 \log Z + 24 = -58 - 10 \log 200 + 24 = -57 \text{ dBm}$
- (c) $G_m(\text{dBm}) = S_P - 94 = -57 - 94 = -151 \text{ dBm}$

EXAMPLE 20.2 A microphone has a sensitivity listed as -56.6 dB ($0 \text{ dB} = 1 \text{ mW/Pa}$) and a 600- Ω output impedance.

- (a) What is the voltage sensitivity in dBV?
- (b) What is the open-circuit voltage with a singer singing at 102 dB?

Solution

- (a) $S_V = S_P + 10 \log Z - 24 = -56.6 + 10 \log 600 - 24 = -52.8 \text{ dBV}$
- (b) $20 \log V = S_V + L_P - 94 = -52.8 + 102 - 94 = -44.8 \text{ dB}$
 $V = 10^{-44.8/20} = 5.75 \text{ mV}$

20.5 ■ AMPLIFIERS

Home entertainment amplifiers increase the signals of CD players, tape players, and tuners from line level (from around half a volt to several volts) to a level appropriate for loudspeakers. Even though the phonograph is becoming obsolete, modern amplifiers still include a preamplifier that boosts the low-level phonograph signal (a millivolt or two) to line

level. An amplifier may be packaged as a separate component or integrated with a tuner into a receiver.

The amplifier has controls to select the desired program source, adjust the desired signal level ("volume"), and adjust the tonal balance between high, middle, and low frequencies. The tonal balance is usually adjusted by means of bass and treble controls, although some amplifiers provide equalizers which adjust the gain in several frequency bands individually. The action of typical treble and bass tone controls is illustrated by Fig. 20.9.

Many amplifiers have a loudness control. With the loudness switch turned on, the frequency response of the amplifier will automatically change with the volume level. At high level, the response will be essentially flat, but as the volume level is reduced, the bass and treble are increased to compensate for the reduced sensitivity of the ear to low-level sounds of high and especially low frequency.

The power amplifier section is designed to supply substantial amounts of audio power to the loudspeakers with as little distortion as possible. Because large amounts of power are involved, efficiency becomes an important consideration. The large output transistors are supplied DC power from the power supply; some of it is converted into audio output, the rest into waste heat. Because this heat must be carried away by heat "sinks," designers try to minimize its production.

One way to minimize waste heat in the output transistors is to have them conduct electricity only when an audio signal is present, and then only as much as necessary. A circuit so designed is called a class-B amplifier, as opposed to a class-A amplifier in which the transistors conduct nearly all the time. Unfortunately, a class-B amplifier generates substantially more distortion than does a class-A amplifier. The solution arrived at in most power amplifiers is to operate in an intermediate condition known as *class-AB*.

Waste heat is usually carried away from the output transistors by attaching them to large aluminum or copper heat sinks, which are good conductors of heat. The heat sinks in turn are cooled by air motion, or convection. For this reason, it is important to comply with the manufacturer's instructions regarding the free circulation of air past the output circuitry. Transistors, which normally last for years, can be destroyed in a wink by overheating or overloading electrically!

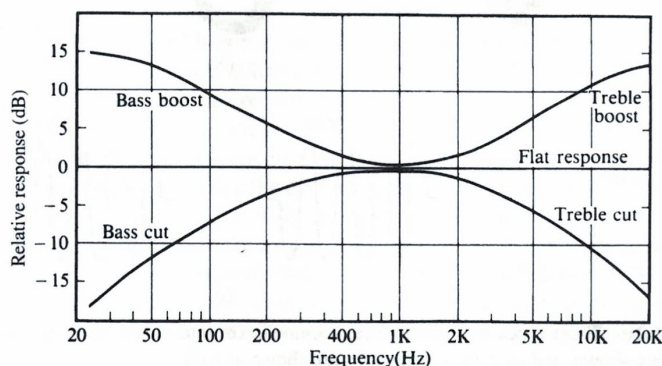


FIGURE 20.9
Action of typical
bass and treble
controls.

20.6 ■ DISTORTION

Distortion refers to any signal not in the original recording or program source that appears in the reproduced sound. Distortion can be generated in any component in the audio system. Once generated, it will most likely be reproduced by the remaining components along with the desired sound.

The three types of distortion that are most significant in a high-fidelity system are harmonic distortion, intermodulation distortion, and transient distortion. Harmonic and intermodulation distortion usually result from a nonlinear transfer characteristic in some component. Transient distortion results from insufficient damping.

Harmonic distortion results in the creation of harmonics of the fundamental frequency that are not present in the original signal. "Clipping" of waveform peaks, which results in substantial harmonic distortion, can occur because of overloading some component. Small amounts of harmonic distortion often escape notice, because musical tones already contain several harmonics. Harmonic distortion is expressed as a percentage of the total signal.

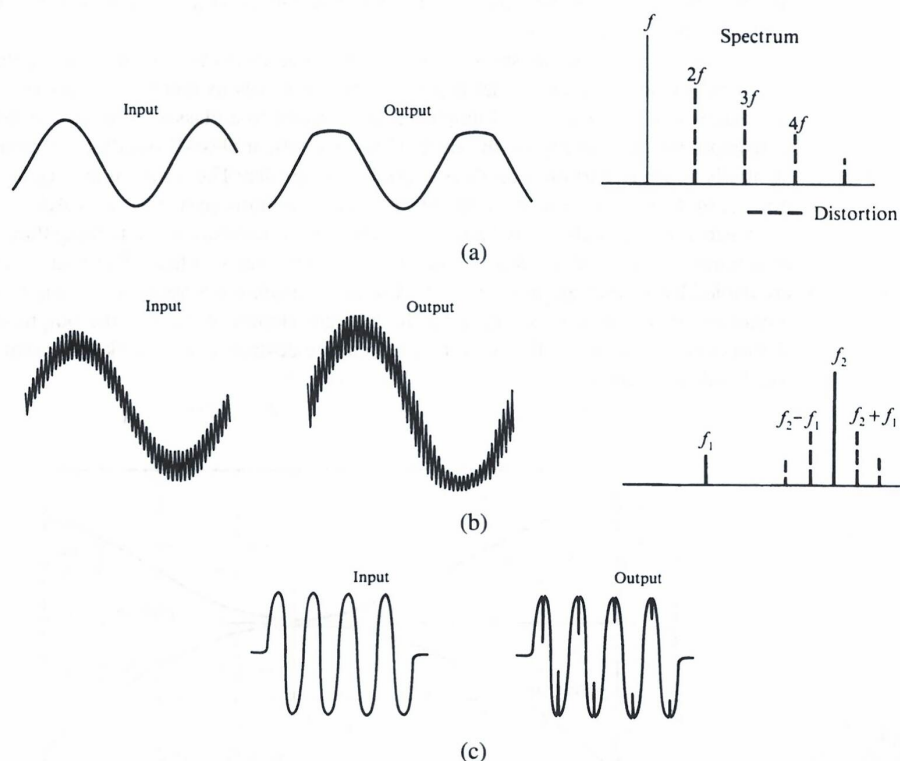


FIGURE 20.10 Three types of distortion: (a) harmonic; (b) intermodulation; (c) transient. In each case, input and output waveforms are shown, and spectra of (a) and (b) are shown as well.

Intermodulation distortion may result when tones of two or more different frequencies are reproduced at the same time (almost always the case in music). A nonlinear characteristic in some component results in the creation of sum and difference frequencies. These new frequencies are harmonically and musically unrelated to the desired tones, and are therefore more noticeable and objectionable than harmonic distortion products. Intermodulation distortion is also expressed as a percentage of the total signal.

Transient distortion occurs when some component cannot respond quickly enough to a rapidly changing signal. In a loudspeaker, it may occur because of insufficient damping of a mechanical resonance. The loudspeaker depends on the amplifier to damp its motion, and an amplifier with a damping factor of at least 20 is desirable. (The *damping factor* expresses the ratio of speaker impedance to "internal resistance" in the amplifier output.)

Transient intermodulation distortion (TIM) can occur in an amplifier because time delay in the feedback loop causes momentary overloading of some circuit element. Thus the percussive sound of a cymbal is distorted, although the steady sound of a violin is not.

Figure 20.10 illustrates the three types of distortion just discussed. If the input signal is a pure tone of frequency f , harmonic distortion will create distortion products with frequencies $2f$, $3f$, etc. Intermodulation distortion may appear as amplitude modulation (AM) or frequency modulation (FM). AM distortion, shown in Fig. 20.10(b), produces distortion products at the sum and difference frequencies. (FM distortion, not shown, produces additional frequencies as well.) One type of transient distortion is shown in Fig. 20.10(c).

20.7 ■ AMPLIFIER POWER AND DISTORTION

The most familiar specification of amplifiers is the output power; it is also one of the most misleading. In an electronics catalog from the early 1970s, one can find an assortment of expressions such as music power, IHF power, peak power, or rms power. In 1974, the Federal Trade Commission (FTC) ruled that the continuous average power per channel, the power frequency response, and the total harmonic distortion at full rated power must be stated "clearly, conspicuously, and more prominently than other representations." Furthermore, the test conditions for measuring power were clearly defined, including a preconditioning at one-third of rated output (the heating of the output transistors is usually greater at one-third of rated output than at full power). A few manufacturers still use the incorrect term rms power to refer to continuous average power, however.

How important is power? It is not easy to answer this question, because room conditions, loudspeaker efficiencies, and listening habits differ widely. A typical average listening level of 65 dB in a room may require from 0.01 to 0.03 mW of acoustic power. Even with a low-efficiency (1%) loudspeaker, this requires only 0.001 to 0.003 W of electrical power. However, at an orchestra concert, you would hear peaks to 105 dB or even more. To reproduce such peaks, 40 dB above the average level, would require a 10,000-fold increase in power! This would require 10 to 30 W of power from the amplifier.

Table 20.1 is a rough guide to the amplifier power requirements recommended in a publication by quality equipment manufacturers (Institute of High Fidelity 1974). The figures appear to pertain to living rooms with large amounts of absorption. In most dormitory rooms, 10 W per channel even into low-efficiency speakers will reproduce peaks well in excess of 100 dB.

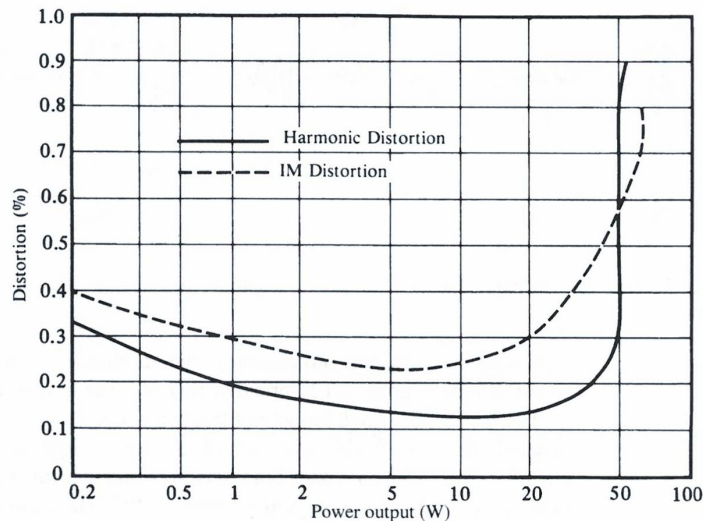
TABLE 20.1 A rough guide to power-amplifier requirements for various speaker efficiencies and room sizes

Amplifier power (continuous watts per channel)	Highest sound pressure level (in dB) possible for a room of the indicated volume (in ft ³)											
	Low-efficiency systems				Medium-efficiency systems				High-efficiency systems			
	2000 ft ³	3000 ft ³	4000 ft ³		2000 ft ³	3000 ft ³	4000 ft ³		2000 ft ³	3000 ft ³	4000 ft ³	
10	94 dB	92 dB	91 dB		97 dB	95 dB	93 dB		102 dB	101 dB	100 dB	
20	97 dB	95 dB	94 dB		100 dB	98 dB	96 dB		105 dB	104 dB	103 dB	
35	99.5 dB	97.5 dB	96.5 dB		102.5 dB	101.5 dB	98 dB		107 dB	106 dB	105 dB	
50	101 dB	99 dB	98 dB		104 dB	102 dB	100 dB		109 dB	108 dB	107 dB	
75	103 dB	101 dB	100 dB		105 dB	103.5 dB	101.5 dB		110.5 dB	109.5 dB	108.5 dB	
100	104 dB	102 dB	101 dB		107 dB	105 dB	103 dB		112 dB	111 dB	110 dB	
125	105 dB	103 dB	102 dB		108 dB	106 dB	104 dB		113 dB	112 dB	111 dB	

Notes: Ninety to ninety-five decibels of typical sound pressure level would be about as loud as what you would hear when sitting in the balcony seats at a live concert. One hundred decibels is fairly loud, about what you could expect at midorchestra seating in a concert hall. One hundred ten decibels is reached by an orchestra playing full blast, with the listener in the front rows of the hall. One hundred twenty-six decibels is considered to be the threshold of physical pain. (From the Institute of High Fidelity, © 1974. Reprinted by permission.)

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FIGURE 20.11
Typical amplifier
distortion curves.
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the Electronic
Industries
Association.)



Salespeople have been known to recommend the purchase of high-power amplifiers because the large amount of reserve power will supposedly assure less distortion. This is not normally the case. In most modern amplifiers, harmonic and intermodulation distortion remain low until one approaches the rated output power. Rarely is the amplifier the main source of distortion in a high-fidelity system, in any case.

The harmonic and intermodulation distortion of a typical amplifier are shown in Fig. 20.11. Intermodulation is generally measured by the application of two signals with widely spaced frequencies (e.g., 60 Hz and 7000 Hz) in a 4:1 ratio of amplitudes. In both cases, distortion is expressed as a percent of the output signal.

20.8 ■ AM/FM TUNERS

Radio stations are assigned a frequency in one of two bands: 540–1600 kHz ($5.4\text{--}16 \times 10^5$ Hz) and 88–108 MHz ($8.8\text{--}10.8 \times 10^7$ Hz). Broadcasts in the upper band use *frequency modulation* (FM), whereas broadcasts in the lower band use *amplitude modulation* (AM). At the lower frequency of the AM band, signals can be transmitted over long distances, but because the AM band is crowded, stations are allowed to transmit audio frequencies up to 5000 Hz only. This severely limits the fidelity of sound reproduction. Furthermore, AM signals are subject to interference from electrical noise, both natural (atmospheric electricity) and human-made (car ignition, motors, etc.). On the other hand, FM stations are allowed to transmit a much greater range of audio frequency, but the high-frequency waves in the FM band are normally limited in range to “line of sight” or slightly more. Within metropolitan areas, however, FM radio has become the major source of high-fidelity program material.

Figure 20.12 illustrates the difference between amplitude modulation and frequency modulation. In both cases, audio information is superimposed on a radio carrier wave of

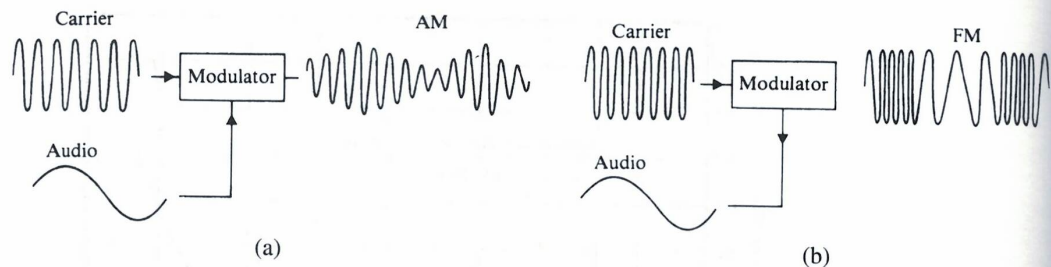


FIGURE 20.12 A schematic diagram showing (a) amplitude modulation (AM); (b) frequency modulation (FM).

high frequency. In AM broadcasting, the amplitude is varied (modulated) in accordance with the audio signal; in FM broadcasting, it is the frequency that varies.

The AM band, which has a total frequency range of 1060 kHz, is divided into broadcast channels that are 10 kHz wide, which is wide enough for the 5000-Hz audio signal plus channel separation, so it does not interfere with adjacent channels. On the FM band, on the other hand, with a total frequency range of 20 MHz, channels can be 200 kHz wide. This bandwidth, 20 times greater than that of AM stations, allows not only a wider range of audio frequency but a wider dynamic range as well; furthermore, it permits the broadcasting of stereophonic sound.

The tuner selects the AM or FM signal from the desired station, amplifies it, and then *demodulates*, or detects, it (that is, extracts the audio signal). (The technical details of how this is accomplished will not be described.) Many modern AM/FM tuners are finely engineered instruments, capable of quality sound reproduction. Furthermore, the performance specifications are much more accurate and complete than for other components in an audio system. A few of the more important specifications are as follows.

1. *Sensitivity* describes the ability of the tuner to pick up weak signals. The sensitivity is usually expressed as the electrical signal (in microvolts) at the antenna, which will result in an audio signal 30 dB greater than the noise level or, more recently, in dBf for 50 dB quieting.
2. *Signal-to-noise (S/N) ratio* expresses the ratio of audio signal to noise for a strong input signal. As the input signal increases, both noise and distortion usually decrease, as shown in Fig. 20.13. The S/N ratio for stereo reception is generally somewhat less than for mono reception.
3. *Total harmonic distortion (THD)*, also shown in Fig. 20.13, compares (in percent) the harmonic distortion generated in the tuner to the undistorted signal.
4. *Selectivity* measures the ability of the tuner to reject stations on nearby channels. It is usually expressed in decibels of attenuation for a station two channels (400 kHz) away.
5. *Capture ratio* expresses the ability of a receiver to reject the weaker of two signals on the same channel. The capture ratio (in decibels) tells how much smaller the one signal must be to be rejected; a capture ratio below 3 dB is desirable.
6. *Stereo separation*, the separation between stereo channels, should be at least 30 dB at all audio frequencies.

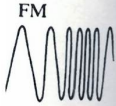
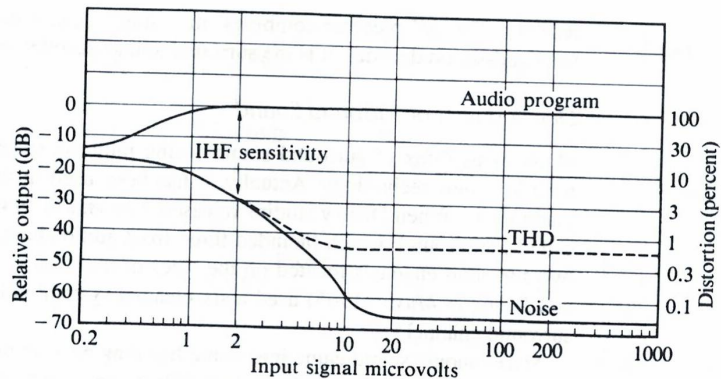


FIGURE 20.13
Typical noise and distortion in an FM tuner. (From the Institute of High Fidelity, © 1974. Reprinted by permission of the Electronic Industries Association.)



Other tuner specifications may include *image rejection*, *i-f rejection* and *spurious-response rejection*, all of which describe the ability to reject some type of undesired signal.

20.9 ■ STEREO BROADCASTING

Much of the program material of FM stations these days is broadcast in stereo. However, it is broadcast in a compatible format, so that it can be received on either stereophonic or monophonic tuners. This is done by mixing the left (L) and right (R) signals together to create sum and difference signals, which are then broadcast.

The sum ($L + R$) is broadcast as a conventional FM signal and is received by both monophonic and stereophonic receivers. At the same time, a difference signal ($L - R$) is broadcast at a higher frequency. A monophonic FM radio, of course, receives ($L - R$) along with the regular audio signal, but it is not heard because of its high frequency. A stereophonic receiver, on the other hand, reconstructs the desired (L) and (R) signals from the ($L + R$) and ($L - R$) signals received (Fig. 20.14). Because the signal-to-noise ratio of a tuner is usually greater when operated in the mono mode, it is sometimes desirable to use this mode when listening to a weak or distant FM station.

20.10 ■ AV RECEIVERS: SURROUND SOUND

The most recent advances in home entertainment systems combine music and video in the home theater. A key component of the home theater system is the AV (audio-video)

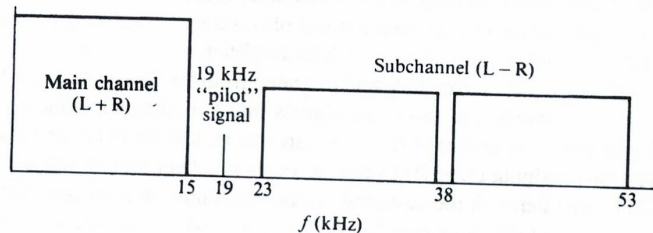


FIGURE 20.14
Composition of a stereo broadcast signal.

receiver. The AV receiver combines, in a single box, a tuner, a power amplifier, and a surround-sound decoder. It is the surround-sound decoder that allows home theater.

Development of Surround Sound

Many of us think of surround sound (using multiple speaker channels in an audio system) as a new technology. Actually, it has been used in the motion picture industry for many years. When Disney studios released *Fantasia* in 1940, they used a new technology called Fantasound, which included three front audio channels located on the screen plus two surround channels located on the sides of the theater. The movies *Oklahoma* (1955) and *Sound of Music* (1965) used a six-channel system with five screen channels and one surround channel.

Stereophonic sound came into home listening as a simplification of theater sound. The physical nature of LP records made it difficult to have more than two channels. Attempts at quadraphonic sound in the late 1960s and early 1970s failed, largely because of competing formats and resistance to placing four large speakers in the home. The stereo format dominates the home market, which has expanded to include FM radio, many tape formats, and CD audio.

Meanwhile, the movie industry expanded surround sound by using Dolby Stereo (an amplitude-phase matrix derived from quadraphonic sound technology) to place four channels of sound onto an optical sound track that had space for only two tracks. Three channels were used at the screen in addition to one surround channel. The requirement for extra power in the bass with movies such as *Star Wars* and *Close Encounters of the Third Kind* led to dedicated subwoofers in the system. Finally, with *Superman I* the surround sound was split into two separate channels, left and right. In 1987 the Society of Motion Picture and Television Engineers (SMPTE) developed a standard for the minimum number of channels needed to create the desired sensations in the theater. This standard is called 5.1, which means five amplified channels are needed (front left, front center, front right, surround left, and surround right) as well as a subwoofer channel with its own amplifier. New digital formats, such as DVD, have taken surround sound from the theater to home entertainment.

Surround-Sound Technology

A typical home theater system is shown in Fig. 20.15, which consists of an AV receiver connected to various input sources and a set of loudspeakers. Audio inputs include analog devices such as phono, tape deck, and VCR as well as digital devices such as CD, DVD, and digital TV. The output meets the 5.1 surround-sound specification by giving five separate amplified outputs plus a separate subwoofer output that is not amplified. Typical subwoofers have a built-in amplifier.

As surround sound migrated from the cinema to home theater, several encoding schemes combined up to four signals into two channels. One of the early schemes (called *Hafler stereo*) passed the stereo signals (L and R) to the left and right speakers and the different signal (L - R) to the surround speakers, thus synthesizing the ambient sound. Including delay in the surround speakers resulted in a greater "feel" of a real concert hall. In 1982 Dolby Surround Sound was released for the home theater market, which added a center

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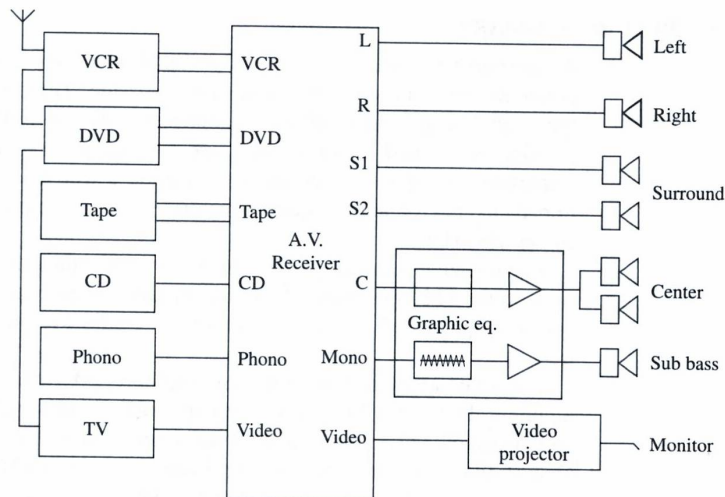


FIGURE 20.15
Typical home theater system.

channel ($L + R$). The encoding is a 4-2-4 scheme, in which four original channels (left, right, center, and surround) are encoded into two stereo tracks and then decoded back into the four original channels. The sound intended for the surround channel is recorded on top of the left and right channels but out of phase with respect to each other. The surround channel is recreated by taking the difference ($L - R$) and then adding delay. Dolby Surround ProLogic is a common method used to record movies onto VHS tape. Sound localization is improved by actively sensing the dominant direction of sound and increasing the gain to loudspeakers in that direction. This allows the sitting area to be expanded while preserving the surround effect.

Dolby Digital was introduced to meet the new standards being developed for digital television, digital video discs, and direct broadcast satellite. Dolby Digital has a full 5.1 implementation, where signals for five channels and a dedicated subwoofer are stored separately rather than combined onto two channels through a matrix. Other multichannel surround sound systems are available. Digital Theater Systems (DTS), commonly found on laser discs, is used for mixing and reproducing live performances. It is capable of 5.1 surround sound but has less compression than Dolby Digital. Sony Dynamic Digital Sound (SDDS) adds two extra screen channels for a 7.1 capacity. Lucasfilm combined with Dolby Labs to develop TDX Surround EX, which extends Dolby Digital to 7.1 by adding rear surround speakers.

Because many surround-sound formats are available (although they may be incompatible), it is important to determine which formats are decoded when you select an AV receiver. It is possible to find receivers that will decode Dolby ProLogic, THD, and Dolby Digital.

20.11 ■ SUMMARY

A microphone produces an electrical signal when actuated by sound waves; most microphones respond to variations in sound pressure. There are many different microphone types, including crystal, dynamic, condenser, electret-condenser, and ribbon. The most popular directional characteristics are omnidirectional and cardioid (unidirectional). Important microphone specifications include sensitivity and output impedance. High-impedance microphones require transformers or in situ preamplifiers if long microphone cables are to be used.

Radio stations transmit in two frequency bands; the greater bandwidth allocated to stations in the FM band makes high-fidelity stereophonic sound transmission possible. The performance of FM tuners and receivers is well described by rather elaborate specifications.

Integrated amplifiers include a preamplifier and a control section as well as a power amplifier. Power amplifiers produce large audio signals with very small amounts of noise and distortion. Distortion is of three main types: harmonic, intermodulation, and transient. Distortion in loudspeakers is more prevalent than in amplifiers, but is seldom in the manufacturer's specifications. High-efficiency, large-cone speakers tend to have less distortion at high power levels than do low-efficiency, long-throw speaker systems.

Home theater systems use six channels which include left, center, and right loudspeakers in front, two surround-sound speakers on the side, and a subwoofer with a separate amplifier.

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GLOSSARY

amplitude modulation (AM) The method of radio broadcasting in which the amplitude of a carrier wave is determined by the audio signal. The AM band extends from 540 to 1600 kHz.

AV receiver A home entertainment component including a tuner, amplifier and surround sound decoder.

carrier wave A high-frequency electromagnetic wave capable of being modulated to transmit a signal of lower frequency.

class AB A power amplifier circuit in which current flows in each transistor for slightly more than one-half of the cycle.

complementary symmetry An *npn* and a *pnp* transistor that otherwise have identical electrical characteristics; such pairs of transistors are used as audio power amplifiers.

condenser microphone A microphone in which the diaphragm serves as one plate of a small capacitor or condenser. As the diaphragm moves, the electrical charge on the condenser varies.

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REVIEW

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distortion Signals that appear in the output of a sound reproduction system that were not present in the original input signal.

dynamic microphone A microphone that generates an electrical voltage by the movement of a coil of wire in a magnetic field.

electret-condenser microphone A condenser microphone that has an electrified foil between the plates of the capacitor, thus eliminating the need for a polarizing voltage as required in an air-dielectric condenser microphone.

electromagnetic induction Generation of an electrical voltage in a wire that moves across a magnetic field.

frequency modulation (FM) The method of radio broadcasting in which the frequency of a carrier wave is modulated (altered slightly) by the audio signal. The FM band, which extends from 88 to 108 MHz, allows stations sufficient bandwidth to transmit high-fidelity stereophonic sound.

harmonic distortion The creation of harmonics (frequency multiples) of the original signal by some type of nonlinearity in the system (the most common cause is overdriving some component).

impedance The ratio of voltage to current. In the case of source impedance or output impedance, it is the current that the device can deliver; in the case of input impedance, it is the current that the device draws from the source.

intermodulation (IM) distortion The creation of sum and difference frequencies from signals of two different frequencies.

piezoelectric crystal A crystal that generates an electric voltage when it is bent or otherwise distorted in shape.

push-pull An arrangement used in power amplifiers; a positive voltage in the input causes the current to rise in one transistor, whereas a negative voltage does the same for another transistor. Push-pull amplifiers usually have an output transformer but do not require complementary transistors.

sensitivity (microphone) Voltage or power generated in a microphone at a given sound pressure level.

sidebands Sum and difference frequencies created in the modulation process.

surround sound A multichannel loudspeaker system used in home theaters.

transducer A device that converts one form of energy into another; in this chapter, we discussed the conversion from acoustic energy to electrical energy (microphone).

transient distortion Overshoot, or other unprogrammed response, that results from the inability of some component to follow a very rapid change in signal.

velocity microphone A microphone that responds to particle velocity rather than to sound pressure.

REVIEW QUESTIONS

1. Which type of microphone is very inexpensive and produces a large electrical signal?
2. Which type of microphone is sensitive to acoustic velocity rather than pressure?
3. Which type of microphone requires a high-voltage bias?
4. Which type of microphones use electromagnetic induction?
5. Which type of microphone pattern might best be used to pick up a quarterback's signals at a football game?
6. What is the advantage of using a low-impedance microphone over using a high-impedance microphone?
7. Which audio component requires the use of an audio preamplifier?
8. Which class of power amplifier is most efficient with least distortion?
9. Which type of distortion is produced by clipping the waveform peaks?
10. Which type of modulation is most commonly used for stereo broadcasts?
11. Which surround-sound decoding scheme uses the 5.1 standard?

QUESTIONS FOR THOUGHT AND DISCUSSION

1. What are the advantages and disadvantages of using microphones of low impedance as compared to those of high impedance?
2. Describe the directional pattern of a cardioid microphone. For what applications are cardioid microphones superior?

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

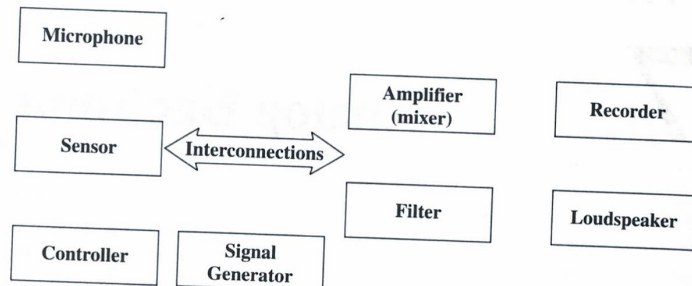


FIGURE 27.1
The basic elements
of electronic music.

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.

