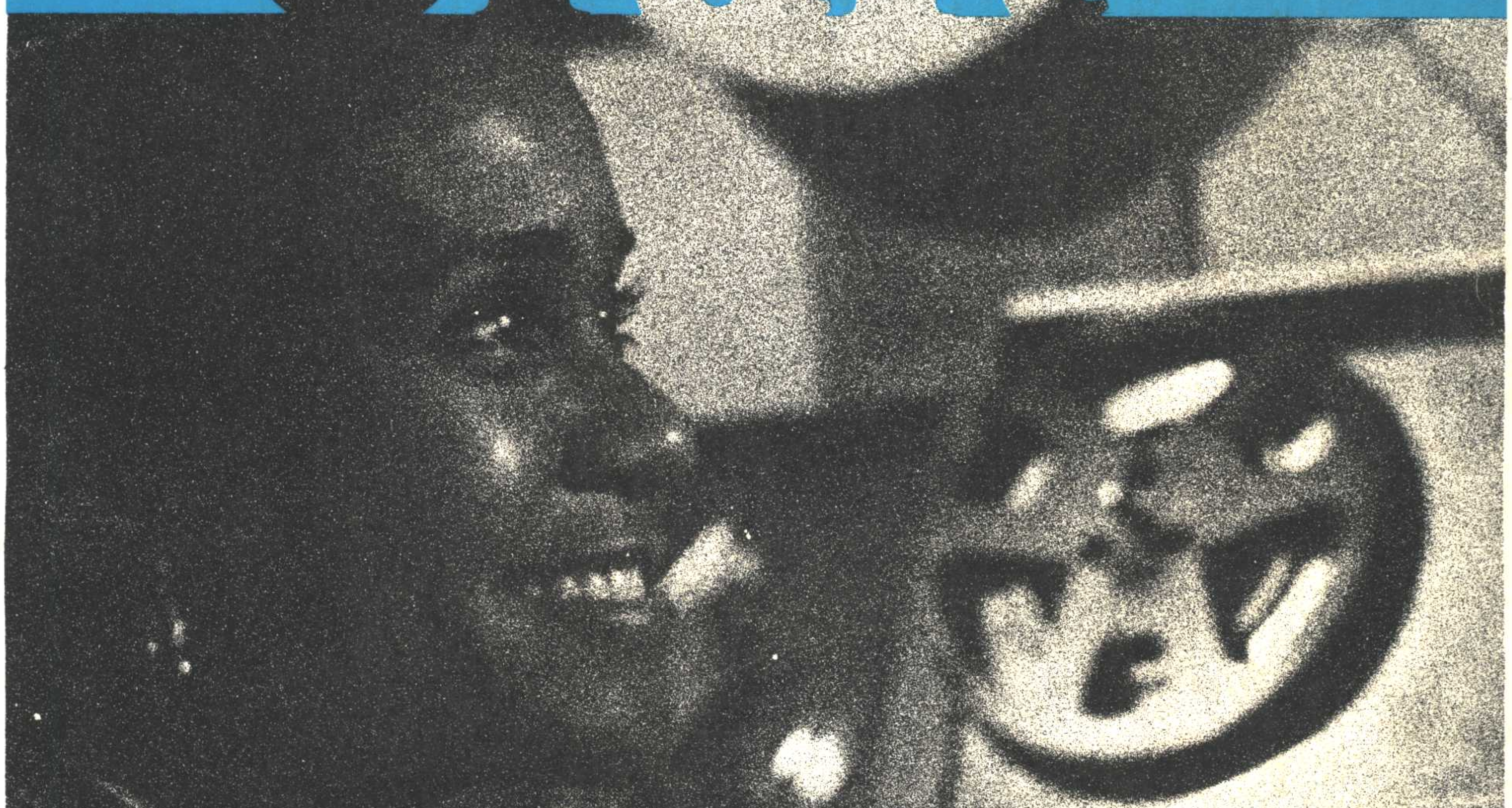
Audio Craft



AN INTRODUCTION TO THE TOOLS AND TECHNIQUES OF AUDIO PRODUCTION

Audio Craft

ABOUT NFCB

The National Federation of Community Broadcasters is a membership organization of some 60 community-based broadcast groups, and over 100 affiliated public radio stations and producers. NFCB represents its members in public policy development at the national level, provides a wide range of practical services, and distributes programs to all noncommercial stations.

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INTRODUCTION

AUDIOCRAFT: AN INTRODUCTION TO THE TOOLS AND TECHNIQUES OF AUDIO PRODUCTION has been developed from the experience of America's community radio stations. Located in all kinds of communities, from rural areas to major cities, and controlled by organizations of men and women reflecting diverse backgrounds, these stations share a commitment to extensive public participation in both governance and programming.

AUDIOCRAFT is a handbook for people seeking a basic "literacy" in the electronic medium of audio production. It is written primarily with radio production in mind, but it should also be of great benefit to people working in video, film and audio/visual presentations.

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The unique and important contribution of community radio has been to open the airwaves to thousands of people who are (or were) not radio producers by profession. The cultural and political life of dozens of communities is richer today because these stations have provided the means through which citizens can speak to one another and share their concerns, their convictions and their creativity.

This innovative approach to direct public participation in the electronic media has come about through a two-stage process. The first step, taken in the late 1960's, involved going beyond the existing role models in both commercial and public radio – the relatively narrow range of music, cultural material and political views, the slick but usually simplistic use of audio technique, and the near exclusive involvement of white men.

The early community stations placed their emphasis on "alternative" broadcasting. Buoyed with tremendous enthusiasm and idealism, these stations produced an exhilarating explosion of music, poetry, children's programming, documentaries and "free-form" radio, the likes of which had never been heard before. They shocked and inspired and challenged their audiences, continually pushing at the conventions and limits of the medium.

For the most part, these early community broadcasters were quite new to radio. The freedom and experimentation of the stations attracted all manner of artists and activists to try their hand. A few signed on as full-time staff, generally at miniscule salaries. The vast majority of this new generation of programmers worked as volunteers.

In time, the community stations came to see that this massive involvement of volunteer programmers -- which many had believed would decline as the stations' financial fortunes improved — was one of community radio's greatest assets. The volunteer programmers brought to the air a continuing diversity of viewpoint, taste and experience. They were a strong link to the ebb and flow of community life.

Unfortunately, energy, creativity and passion, alone, do not suffice for good radio. As the stations (and their audiences) became more sophisticated and demanding, it became increasingly clear that gaps in basic skills needed to use the medium of radio with competence were undermining this important effort.

The community stations recognized that their missions of participation, access, experimentation and effective community service meant little if no one listened, if those who were participating were not really *communicating*. Out of this recognition came the important second stage of community participation: an ambitious and ongoing commitment to training.

Today, at virtually all community stations, there is a regular program of production training, designed to assure that the many volunteers who come to use the station can do so with the fullest power and effect that radio has to offer.

This commitment to training has not been an easy one to meet. Since the beginning of the medium, most audio producers have learned their craft as apprentices, a time-consuming process that depends heavily on an able and willing teacher. Both the learning and the teaching have suffered from the lack of a single, comprehensive text that guides the novice producer from early concepts to finished work. The problem has been especially serious for community broadcasters, with the large number of people involved and the special emphasis on extensive local production.

AUDIOCRAFT is meant to be such a guide and, we hope, another important step in realizing true community access and participation in radio. It is written so that it can be understood by those without science and engineering backgrounds. It will be useful to performers, announcers, producers and others who use the medium of sound. It is designed both for use in organized courses and for self-teaching.

As the electronic media play an ever larger role in our national life, we hope that AUDIOCRAFT will be one resource that enables a far broader range of people — whether as full-time professionals or as skilled "citizen producers" — to use these powerful tools.

Washington, D.C.
June, 1982

Thomas J. Thomas
Theresa R. Clifford

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PART ONE

SOME BASIC THEORY

• SOUND • ELECTRICITY AND SOUND • THE INS AND OUTS OF AUDIO CONNECTIONS

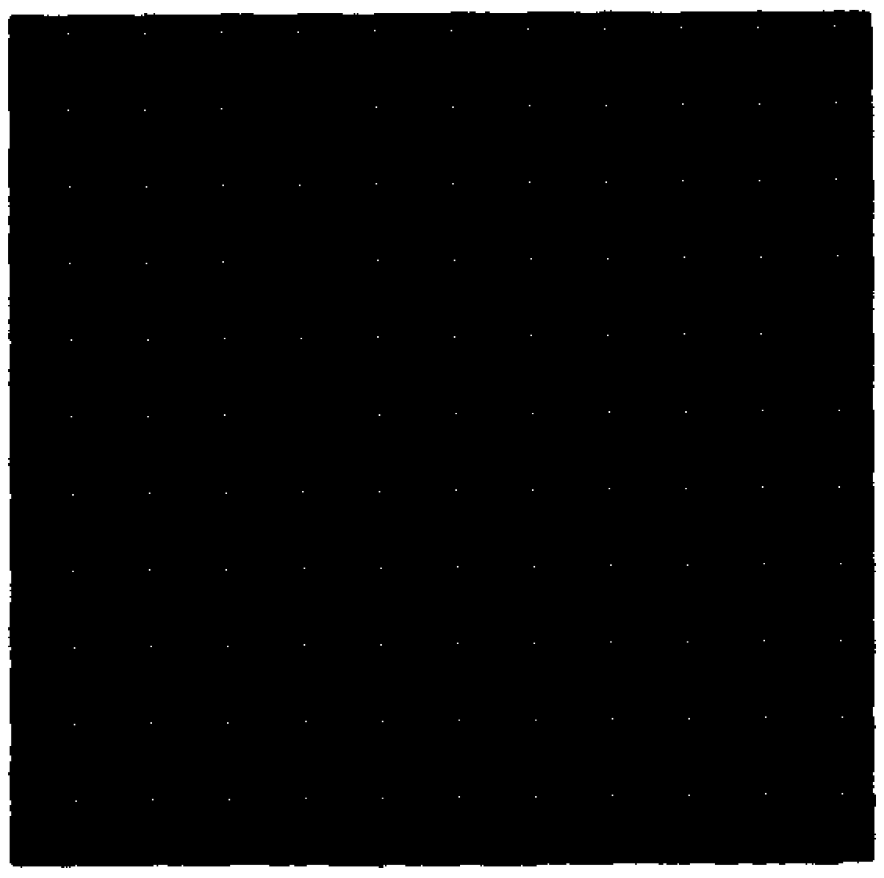
This book is mainly about the techniques used to record and process sounds, so it is important to have some idea of how people perceive sound in the first place. A few comments will show you why and, we hope, cause you to think about, read more about, and experiment with sounds and how they are interpreted.

- There is no time when, if we listen, we won't hear some sound! (The "natural" sounds within any given environment are called *ambience*.)

- Most of us are so accustomed to our eyes and ears working together that we don't often think about them as separate and very different senses. For example, when inexperienced radio reporters are sent to record a press conference, they often stand at the back of the room and hold their microphones in the air in order to pick up the voices of the speakers in front.

  What these reporters don't realize is that what they see unavoidably affects what they hear. The sight of the speaker's lip movements, facial expressions and body movements all add meaning to the words being spoken. The ability to locate the speaker in a particular part of the room also helps the reporter to ignore sounds coming from other parts of the room.

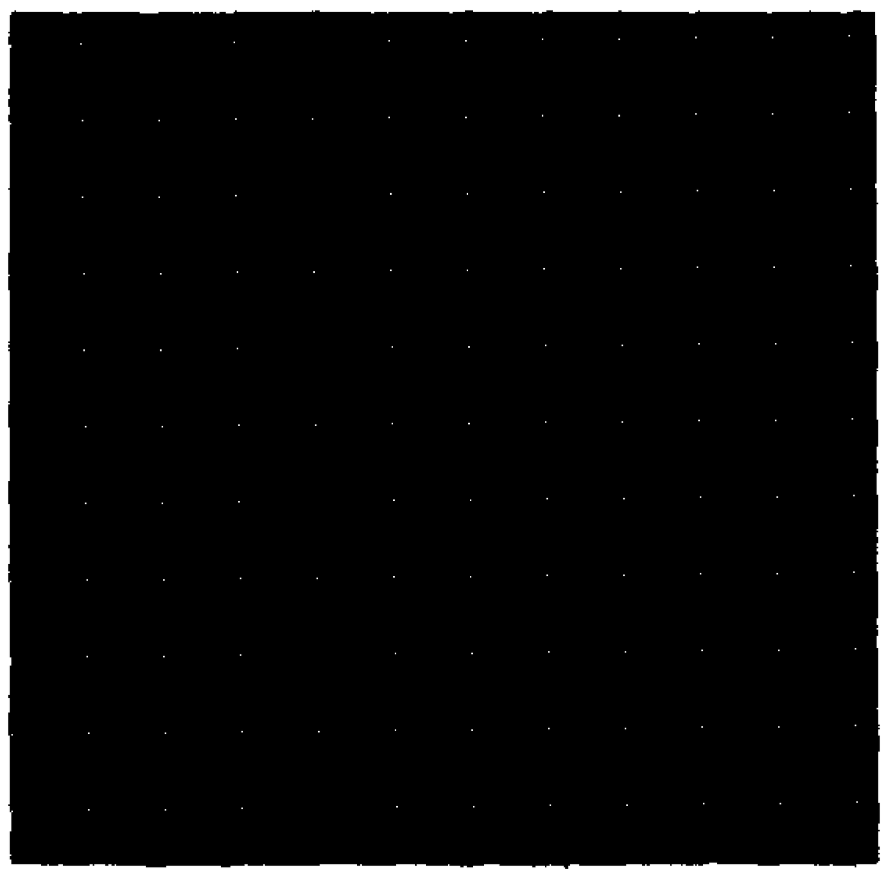
  The radio listener doesn't have these advantages, and this makes the clarity of the voice recording essential. The closer the microphone is to the person speaking, the easier it will be for the radio listener to clearly hear what is being said.



- Taking a picture and recording a sound are very different kinds of operations, partly because of the difference in the way light and sound are reflected. Making a good picture is in many ways much easier than making a good recording of a sound, because a visual image is *not* reflected by its surroundings nearly so much as a sound is.

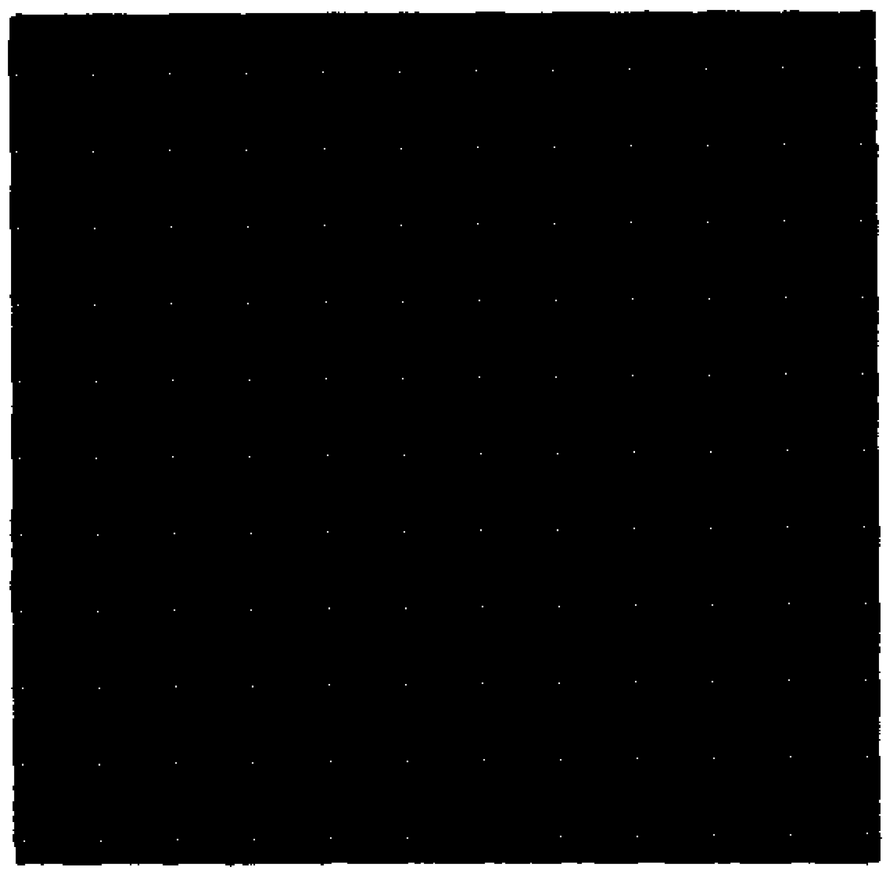
  Trying to make a clear recording of somebody speaking indoors is a little like trying to get a good picture of somebody who is standing in a roomful of mirrors. The only way to get a good picture is to move the camera very close to the person. Likewise, it's usually necessary to place a mike within a couple of feet of the desired sound source to obtain a clear recording, with a minimum of unwanted sound.

- The printed word allows the reader to go back and re-read anything that might have been confusing the first time. The radio listener has no such option. In radio, the art of repeating ourselves, creatively, allows us to convey information as clearly as possible.



- The silences between words or between notes of music are as valuable in conveying meaning as the sounds themselves; each is meaningless without the other.

- Radio, unlike television, requires the listener to use imagination.



1 SOUND

The basic attributes of a sound, by which it is described or compared with another sound, are: its *frequency* (pitch), its *duration* (length in time), its *volume* (loudness), and its *timbre* (tone color – the relative strength of all the component frequencies in a sound). Further, all natural sounds vary in loudness over time — this is called *dynamics*. The range of volumes of any natural sound, from softest to loudest, is thus referred to as its *dynamic range*.

For example, imagine a conversation involving several people. Some of the people have louder voices than others (volume). Every voice has a basic pitch (bass, treble, or somewhere in between), but each person's voice sounds unique, due to the combinations of various frequencies within a voice (timbre). The conversation lasts for a certain period, as does each word and sentence (duration), and it varies in volume as it progresses (dynamic range). We'll see that each of these characteristics presents specific problems and possibilities.

As sound technicians, we have to consider more than just the source of sound. All sounds happen in an environment. There are likely to be several sources of sound in a given environment – some that we want to record, and some that we don't. Also, all of the surfaces in the environment will affect the way we perceive those sounds. If you and I are having a conversation in a room, much of what we hear of each other's voice will have bounced off the walls, ceiling and floor, rather than having come directly from mouth to ear.

How about the equipment we use to record and process sounds? We can evaluate any piece of audio equipment in three basic ways:

- *Frequency Response* — How completely does this device respond to and reproduce all of the frequencies of sound fed into it? (In other words, how accurately is the timbre of the sounds reproduced?)
- *Signal-to-Noise Ratio* — How much noise will this device add to the signal (sound) being fed into it?
- *Distortion* — How accurately will this device reproduce the signal that is being fed into it without altering or deforming it? (For example, when you turn up the volume of your stereo system so loud that you hear crackling sounds superimposed over the music, you are causing the system to "distort.")

Through this book you'll learn how to recognize and deal with problems in each of these areas.

FREQUENCY AND HOW IT IS MEASURED

All sounds are caused by something vibrating. The rate of the fundamental vibration determines the pitch (frequency) of the sound the vibration makes – the faster the vibration, the higher the pitch. A guitar string, for instance, that vibrates back and forth fifty times a second makes a sound that is said to have a frequency of fifty *cycles per second*. Recently, instead of "cycles per second," the word *Hertz* (or "Hz") is used, after a German physicist who made important discoveries about the nature of sound. (Since the metric system is used in science, we refer to a thousand cycles as *kilo-Hertz*, or "kHz.")

Any single-frequency sound is called a *tone*. "Pure" tones do not occur naturally, and can only be generated artificially. All *natural* sounds are *complex*, i.e., they contain *many* frequencies of sound that are heard simultaneously. The relative strength of the various frequencies composing a sound determine the timbre (tone color) of that sound. That is why a piano, a baritone sax, and a singer can all produce the "same" note but still sound different.

The range of frequencies people can hear is from 16 Hz to 20,000 Hz (referred to as the *audio range*). This range is commonly subdivided into *low* (bass), *midrange* (mids), and *high* (treble) frequencies:

- LOW = any frequency below approximately 400 Hz
- MIDRANGE = approximately 400 Hz to 3,500 Hz
- HIGH = any frequency above approximately 3,500 Hz

The frequency response of the ear varies from person to person and is affected by age and the amount of prolonged exposure to very loud sounds. Hearing capability is also affected by such factors as the physical health of the ear and the amount of wax present.

The highest frequencies are usually the first to go in hearing loss. By the age of twenty, many people can no longer hear frequencies above 15 kHz. This is one of the reasons why tastes vary as to the amount of high frequency signal desirable in music. People who are in the habit of turning up the treble (higher frequencies) when listening to recordings or broadcasts may be doing this to compensate for lack of high frequency response in their ears, of which they may be unaware.

THE DECIBEL – COMPARING THE VOLUMES OF SOUNDS

If we are making a recording of a piece of music, and I notice that the saxophone, for example, isn't being recorded as loudly as it should be in relation to the other instruments, I need a way to tell you (as the person operating the controls) exactly how much louder I think it should be. If I say, "I think it should be a little louder," your idea of what "little" means may be significantly different from mine. This need to be precise when comparing volumes of different sounds is soon apparent to all of us who work with the medium.

Human ears can hear a wide range of volumes of sound. The loudest sounds we can hear without pain are about 100,000,000,000 times as powerful as the softest sounds we are able to detect. Working with such large numbers in comparing volumes is awkward, so a logarithmic system of comparison using smaller numbers was devised. The basic unit of this system of comparison is the *decibel* (abbreviated "dB"). We compare different levels of volume by saying that one sound is so many dB more or less than some other sound. A difference of one dB is so slight as to be barely audible. But because the scale is logarithmic, a difference of six dB (for example) is substantial, and a difference of twenty dB is huge.

Differences of volume are commonly measured on a *volume unit (VU) meter.* You do not need a thorough understanding of logarithms to read a VU meter. Anyone who can interpret a typical VU meter knows enough about comparing volumes to produce first-rate audio programs.

KEY TERMS TO REMEMBER

audio range
complex sound
cycles per second
decibel (dB)
distortion
duration
dynamic range
dynamics
frequency
frequency response
Hertz (Hz)
kiloHertz (kHz)
signal-to-noise ratio
timbre
tone
tone color
volume
VU (volume unit) meter

2
ELECTRICITY AND SOUND

In order to send sounds over long distances, amplify them, store them on tape, or broadcast them, they must first be converted to electricity. To better understand this conversion, let's examine one of the most common devices which performs this function: the *microphone*.

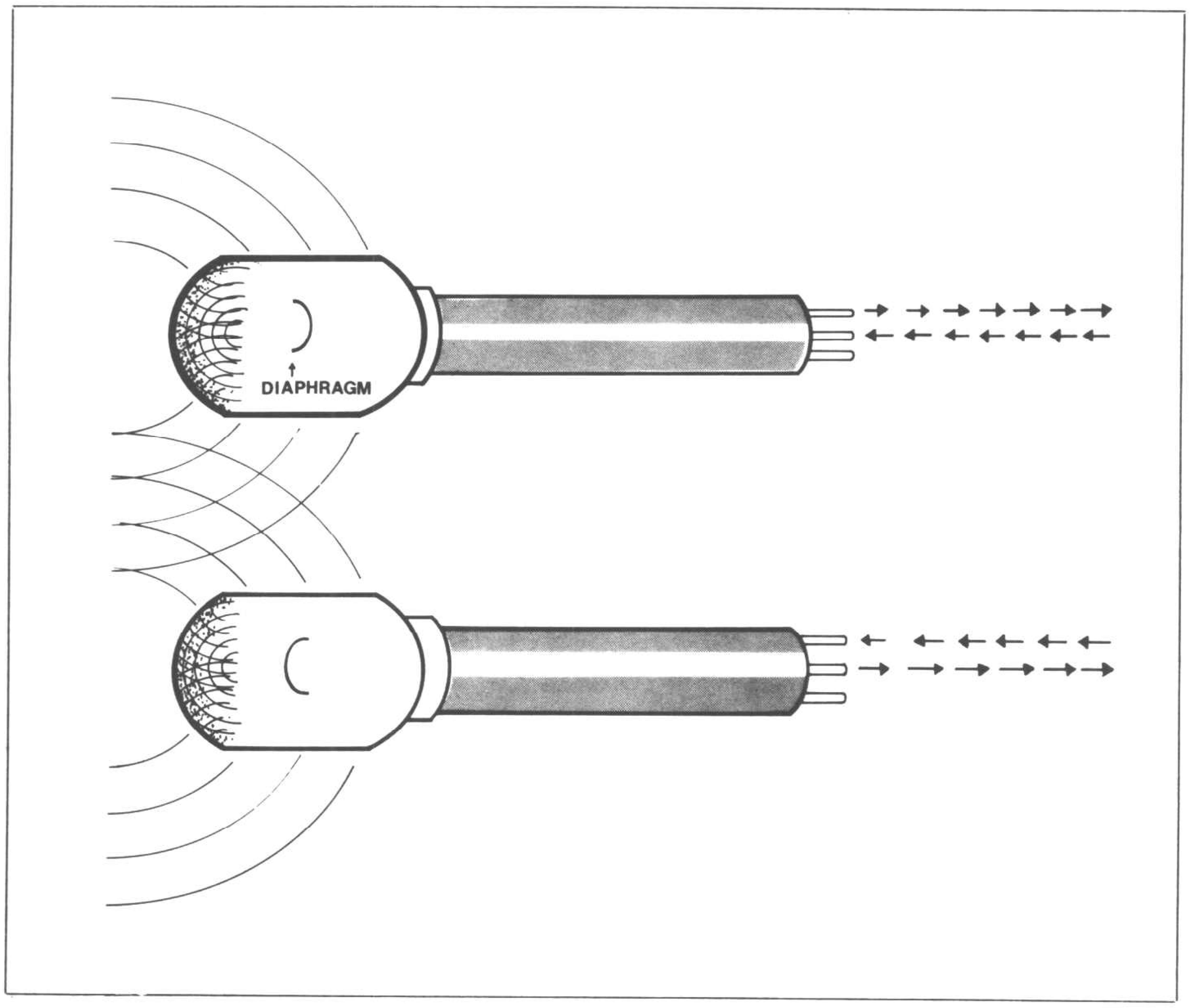
All microphones are electrical generators, which produce very tiny amounts of electricity. There are several types of microphones used in professional sound work. Each has a different way of generating electricity from sound. We'll use the most common type – the dynamic microphone – as an example.

The *dynamic microphone* contains a magnet and a *diaphragm*, a thin disc which vibrates back and forth as sound waves strike its surface and which in turn vibrates a tiny coil of wire which is an electrical *conductor* (i.e., electrical current will flow through it). The magnet, like all magnets, is surrounded by a magnetic field. (The patterns that iron filings form around a magnet illustrate the lines of force which make up a magnet's field.)

Any time an electrical conductor is moved through a magnetic field, an electric current is generated by the simple fact of their relative movement, and flows through the conductor. The diaphragm in a microphone is designed to vibrate at the same rate (*frequency*) and with the same relative *intensity* (*volume*) as the sound wave which strikes it. The rate and intensity of flow of the electrical current generated by the movement of the diaphragm and coil in the magnetic field is equivalent to the rate and intensity of the sound wave which caused the diaphragm to vibrate.

Thus, when you speak into a microphone, the sound waves of your voice strike the diaphragm, causing it to vibrate. The pitch (frequency) of your voice's sound waves cause the vibration to be at a certain rate (frequency): the higher the pitch of your voice, the higher the frequency. The volume of your voice determines the intensity of the vibration.

Since the coil is an electrical conductor, its vibration (back and forth movement) in the magnetic field causes a current to flow. Again, the rate (number of vibrations per second) determines the frequency of the electrical signal. The intensity determines the "force," or volume! And because the diaphragm moves alternately back and forth, the electrical current which it generates likewise moves back ("negatively," or –) and forth ("positively," or +), and for that reason is called *alternating current*, or "AC." Thus, the microphone has created an *electrical analog* of an *acoustic* event!



HOW A MICROPHONE GENERATES ELECTRICITY *Gail Chase*

Here we see the diaphragm in a mike being vibrated back and forth by a sound. As it bends in one direction, it causes an electrical current to flow in one direction. As it bends the opposite way, the current flows the opposite way. The frequency of this alternating current will be the same as the frequency of the vibrating sound source; thus an electrical analog of an acoustic event has been created.

The *intensity* of an electrical current is referred to as its *amplitude*, or *voltage*. As we have seen, the voltage of an electrical current is analogous to the amplitude (volume) of a sound wave.

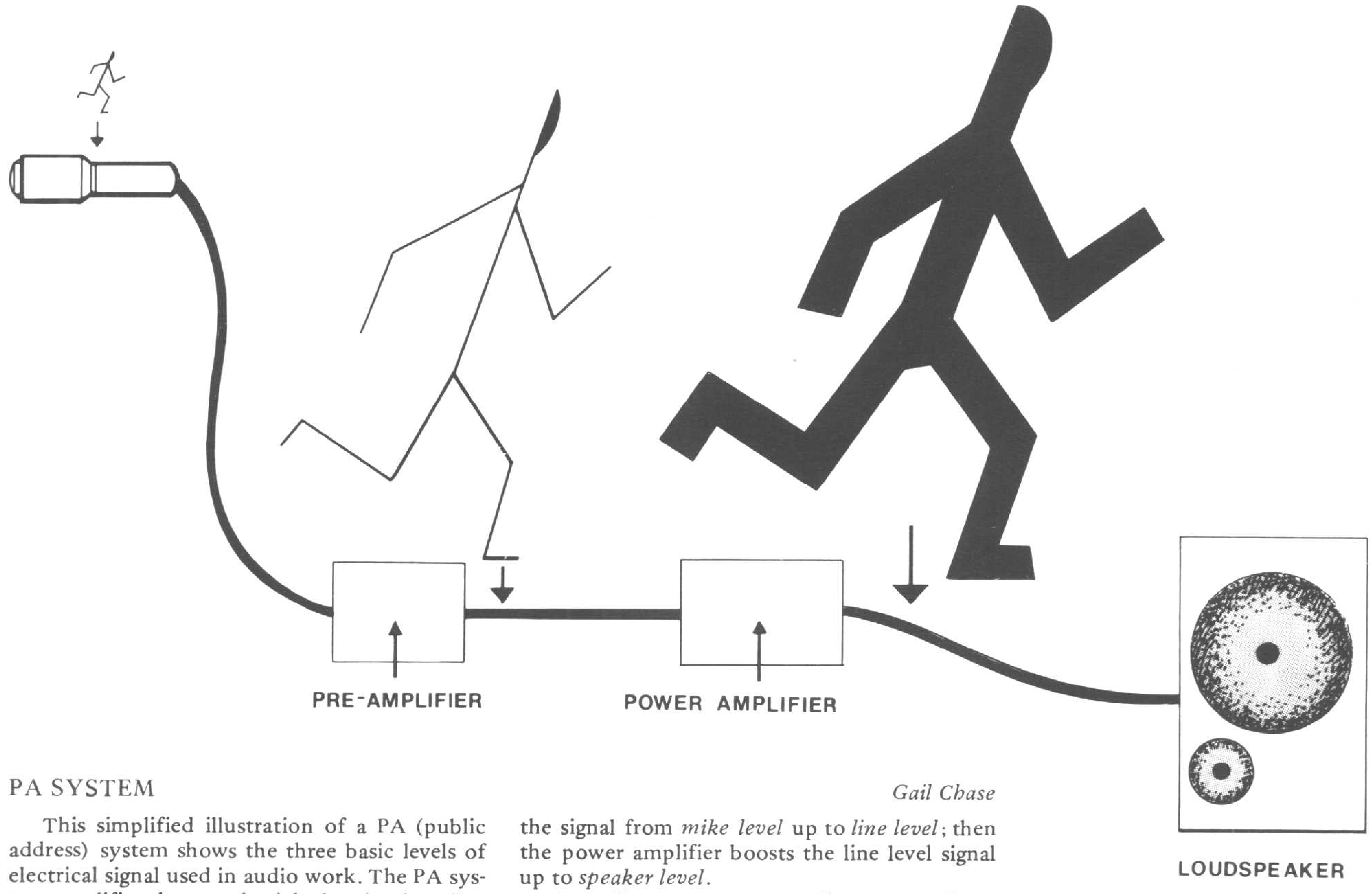
SOUND	**ELECTRICITY**
Volume = Intensity = Amplitude	**Intensity = Amplitude = Voltage**

When we "turn up" the volume of an audio device, we are simply increasing the voltage. One control that allows us to do this is called a *voltage potentiometer*, commonly referred to as a *pot*.

Now that we understand how sound waves are converted into electricity, let's examine a common audio system in which our "electrical analog" is put to use: a *public address* (PA) system. A PA system is a way of making sound louder (*amplification*). The basic components of a common PA system are: a microphone, a pre-amplifier, a power amplifier, a loudspeaker, and cables to connect each of these devices in sequence.

Because the electrical current generated by a microphone is very slight, it must be increased (amplified) significantly before most audio devices can use it. The microphone must therefore be plugged into a *pre-amplifier*. The pre-amplifier (like all amplifiers) *boosts* the voltage (amplitude) of the electrical signal. The amount of voltage generated by a microphone is known as *mike level*; the pre-amplifier boosts this mike level to a stronger signal, known as *line level*.

Once the electrical signal has been boosted by the pre-amp, it can proceed to the *power amplifier*. The power amplifier's job is to boost the line level signal to an even greater level, known as *speaker level*. It takes a very large amount of electricity (compared to that generated by a pre-amp) to make a typical loudspeaker work. The speaker level signal can then be sent from the power amplifier to the loudspeaker, where the "analog process" is reversed, and electricity is converted back to sound!



PA SYSTEM

Gail Chase

This simplified illustration of a PA (public address) system shows the three basic levels of electrical signal used in audio work. The PA system amplifies the sounds picked up by the mike, so they can be heard by many people. It takes two devices to do this: the pre-amplifier boosts the signal from *mike level* up to *line level*; then the power amplifier boosts the line level signal up to *speaker level*.

A similar process occurs whenever we listen to sounds coming from microphones, turntables or tape recorders.

What our simple diagram of a PA system describes is a *gain structure*. The word *gain* in electronics means, simply, a change in volume. Our PA system has two stages of gain. First, the mike level signal is boosted to line level, then the line level signal is boosted to speaker level. The "path" of the signal through all of these components (or through any audio device) is called a *channel*.

KEY TERMS TO REMEMBER

alternating current
amplification
amplitude
channel