

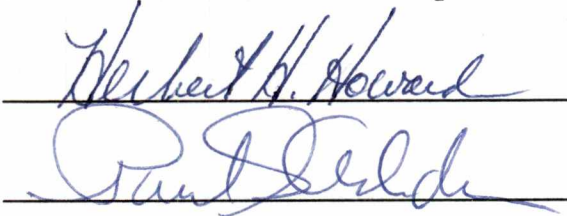
To the Graduate Council:

I am submitting herewith a thesis written by Richard D. Phillips entitled "Musical Sound Design: The Technical and Esthetic Work of the Recording Engineer." I have examined the final copy of this thesis for form and content and recommend that it be accepted in partial fulfillment of the requirements for the degree of Master of Science, with a major in Communications.



Darrel Holt, Major Professor

We have read this thesis  
and recommend its acceptance:



Accepted for the Council:



The Graduate School

MUSICAL SOUND DESIGN: THE TECHNICAL AND ESTHETIC  
WORK OF THE RECORDING ENGINEER

A Thesis  
Presented for the  
Master of Science  
Degree  
The University of Tennessee, Knoxville

Richard D. Phillips

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## ABSTRACT

The technical art and science of the field of recording engineering continues to grow and become more complex. A single text and reference source is needed to aid in the instruction in the field of recording engineering. This work should also serve as a reference for practicing recording engineers.

The production of this manuscript began in the winter of 1982 and concluded in the summer of 1983. Recording engineers were interviewed and studio sessions were observed by the author. The material was compiled and organized for logical flow and consistency.

The text material is divided into chapters dealing with the technical aspects of the engineer's work, and chapters noting esthetic considerations in the work of the recording engineer. Technical chapters cover sound principles, miking, signal flow, control consoles, monitor systems, recorders, and outboard instrumentation. Esthetic considerations are outlined in the chapters dealing with the tracking, overdub, and mixdown sessions.

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## INTRODUCTION

The mystical world of music, its production, and recording surrounds us daily, yet we understand very little of its methodology and technology. Every day most of us encounter thousands upon thousands of sounds. Some are pleasant and soothing. Some are harsh and aggravating. Some cause us to become excited, sad, cautious, sentimental, or anxious. Many simply go unnoticed. Music has long been the medium of the soul, the expression and transmission of emotion through tones, rhythms, and lyrics.

When we think of the sounds we call music, we usually think of the artist, the composer, or the instrumentalist who gave birth to the sound. But how did it arrive in the form we hear? Who put it together? Who is the unseen force that designed the sounds you enjoy as a musical selection?

"Recording engineer" has long been a nebulous term that, for most people outside the music and recording industry, means only some cog in the process of getting music from the source to the audience. In reality, the recording engineer is much more than that. The competent recording engineer is as much an artist in sound design as a performer is an artist in vocal or instrumental

music. Advances in audio technology and recording processes continue to make the art of recording engineering more complex.

### Background of the Problem

The process of recording music dates back to the original cylinder recorder developed by Thomas Edison. From wax cylinders to 78 rpm platters, mechanical recording continued to grow. Music was recorded direct. The sounds that a band and vocalists produced were mechanically captured on the record and played at home or over the radio.

Wire recording was invented prior to World War II. Sounds were translated into magnetic impulses and stored on a thin strand of metal. Following the war, tape recording became a reality. Plastic tape, coated with magnetically sensitive iron oxide, was used in place of the wire. Modern recording was born.

Direct recording was used for several years. Recordings of music and voice were transferred to discs to be distributed to the public and radio stations. Although the sequence of sounds on the tape could be changed by editing, the sounds remained exactly as they had been recorded.

Multitrack recording changed the way sound was designed and recorded. The concept of adding sounds to

previously recorded sounds had existed prior to the late 1950's. When the first multitrack recorders became available in the early 1960's, however, the face of tape recording changed forever.

The person who operated equipment for multitrack recordings became known as a recording engineer. At the present time, recording engineers work with music producers to design most of the music we hear today.

The recording engineer has been called the hands, the controller, the wizard, and the good fairy of the recording studio. The recording studio is the working environment of the engineer. It is here that, at the bidding of the producers of a song, jingle, or album, the engineer constructs the sounds that produce the final tape that the producer has in mind. The producer has a "certain sound" which, for a particular song or musical number, he or she considers the "right" sound. The engineer's job is to put together, technically, the elements of music and electronics to give the producer just that certain sound. Even though the processes, technically, may be the same from recording session to recording session, the way the engineer records, mixes, and adds special effects will differ from producer to producer. Each has his own right sound, and the engineer must give him that.

The engineer is responsible for the mixing of the instruments. Most instruments are picked up by one or more microphones, and the signal is sent to the console. A few electronic instruments may not require a microphone and the output signal is patched directly into the console. The engineer decides which instruments will be recorded in which order, and with which microphones, to get the best sound on the tape.

The engineer operates the console, recorders, out-board electronics (special effects equipment), and sets the microphones to obtain the sound the producer wants. He or she works with the instrumentalists and vocalists to make them feel calm, at ease, and able to perform at their best. The engineer interprets the wishes of the producer, which are usually communicated in a language unique to that producer. He then creates the sound, or changes the sound, the way the producer wants it.

The engineer's work goes beyond simply recording sounds in the studio as they happen. He controls that recording process. His work is not finished when the performers leave; it has only begun. The engineer now takes the sounds he has recorded on the tape and overdubs other sounds: more instruments, perhaps, or some other vocals or special effects. The engineer continues this process until he has recorded on the tape all the basic

materials he needs. Here the final step in his work begins.

After all the original recording and overdub sessions are complete, the engineer now begins the mixdown. If the engineer and the producer have worked well together before, the producer will often trust the engineer to produce what he wants. With or without the producer, the engineer begins the building of the final tape from the multitrack work tape he has produced during the recording and overdub sessions. The engineer controls the level, equalization, and special effects for each instrument and vocal recorded on the work tape in order to produce a final master tape, usually in stereo. This master tape is the final product and is used in the process of pressing a record.

The description above briefly highlights the work of the engineer from start to finish, the result of which produces the final sound you hear from original sounds in the studio. There are variations from this process when, for various reasons, the producer and the engineer are one and the same person, or when the engineer is called upon also to take part as a vocal or instrumental performer during a recording session. It is easy to see why a good engineer is more than a cog in the works, and why a good engineer is well sought after by producers. He is

the creative sound designer who can make the difference between a mediocre melody and a musical masterpiece.

### Statement of the Problem

Even though the expanding field of recording engineering touches many lives daily (through recorded music), there is no structured reference or text material available. To the professional in the field, or to the student who wants instruction in recording engineering, the only recourse is trial and error investigation.

Recording engineering is both an art form and a science. It combines technical skill and esthetic judgment in the design of the sounds known as music. Without a structured approach to the study of this field, education and advancement of recording engineering is hampered.

Increasing numbers of young people now express a desire to enter the field of recording engineering. Others, already pursuing a recording career, can profit from a concise reference source which provides an understanding of additional concepts and techniques.

Toward those ends the purpose of this work is to design a preliminary textbook on recording engineering. The text, with illustrations, is a compilation of technical and esthetic information from engineers on the field. The result, it is hoped, will serve as a classroom resource and as a reference guide for engineers. This

work brings together information about recording engineering for the student and the professional.

### Methodology

The information in these pages examines the world of the recording engineer. The electronic technology and esthetic considerations used by the engineer to design musical sound are discussed. Even though some technical discussion is necessary, this is not intended to be an electronics or physics textbook. Rather, it presents the basic technical and esthetic information recording engineers need to know to do a good job.

The material herein was gathered through interviews with top recording engineers. These engineers were selected because of their accomplishments and their reputations. Interviews were conducted in the studios where they work. Effort was made to cover all technical and esthetic facets of their work. Major interviews were conducted with:

1. Danny Dunkleberger, assistant engineer of LSI Sound Studios
2. Lynn Fuston, designer and recording engineer of Eagle Sound Studio
3. Lee Hazen, owner, producer, and engineer of Studio by the Pond

4. John Loudermilk, owner, producer, and engineer of The Nashville Sound Connection.
5. Warren Peterson, award winning recording engineer and artist of Sound Stage Studios.

Additional information was supplied by: Gary Musick, free-lance recording engineer; Johnathan Yeaworth, free-lance recording engineer and a performer at Opryland; USA; and Jon Bowsaw, sales representative for McLavier Electronics.

This study represents the cumulative knowledge of more than 100 years of experience of these top recording engineers. This work draws together information and experience to describe how the engineer operates in the studio to produce music recordings.

### Literature Search

The research used in the production of this study is primary in nature. The reason primary sources (engineer interviews) were used, is that little has been written on this subject. Most of the material available is written with the electronic designer or the physical scientist in mind.

This study pools information and experience from recording engineers on the field. The following books and articles touch on the subject areas in this study and may be helpful additional reading.

- Alfred, John. Manual of Sound Recording. New York: International Publications Service, 1978.
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- Riggs, M. "How Tape Recording Works." High Fidelity. February 1983. 33. 16.
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- Riggs, M. "The Whys and Hows of Digital Recording." High Fidelity. August 1982. 32. 40.

### Delimitations

This manuscript provides an overview of recording engineering as a professional practice. Every effort has been made to cover the major facets of in-studio music

production. Obviously, not every topic involved in this field can be covered. For example, musical artistry, performance, out-of-studio productions, record pressing and distribution, the responsibilities and duties of various technicians, the various and complicated business aspects--all these are beyond the scope of this work, except as they directly relate to the work of the recording engineer.

Some disagreement in esthetic considerations exists between engineers. This study offers general guidelines in which it notes some differences of opinion. Because of these differences in judgment, a student or professional may find he or she prefers a technique other than those suggested here. This difference does not minimize the value of the study. Rather, it emphasizes the field's general nature and the opportunities for self-expression in it. The following cannot be the last word in the music industry; it is simply a good starting point for the aspiring recording engineer, and a refresher for the practicing engineer.

### Stylistic/Syntactic Devices

The recording engineer is variously referred to as a recording engineer, an engineer, an audio man, or an operator. For clarity, a recording engineer will be referred to as a recording engineer, or an engineer, throughout this work.

AS stated, this work is designed to introduce a student who is only vaguely familiar with audio recording to the world of a recording engineer. It also can serve as a reference manual for engineers already in the field. Because of this dual audience, technical jargon associated with this field is used sparingly throughout. When such a word is used, if the meaning is clear, it will be set apart, the first time it is used, with quotation marks. If a technical term is used in the text, but its meaning not clear from the context, a short definition enclosed in parentheses will follow. All figures referred to in the text are located in the Appendix.

## CHAPTER I

### BASIC SOUND PRINCIPLES

The recording engineer uses a vast array of highly complex electronic equipment. He integrates technical skill and the musician's ear for one purpose: to produce just the "right sound."

Sound is at the heart of, and is the substance from which the final product of the engineer is built. This is the reason we refer to the engineer as both a technician and an artist. This rare combination is best achieved when a person has a clear and solid understanding of that basic building block, sound.

Sound is something we take for granted. Daily we encounter thousands of different sensory impulses we refer to as sound. In order to understand, and thus come closer to being able to use sound, we must first dissect and classify all that we know as sound.

#### Wave Theory

Sound is the result of a physical disturbance, or vibration, in the material substance which surrounds our audio perceptor, the ear. The vibration must travel to the ear, where it is converted into nerve impulses and sent to the brain for analysis. By our definition, then,

sound exists as a vibration, usually in the air, independent of hearing. Hearing is the effect of sound on the audio perceptors of the subject, the ears.

This is an important concept to the engineer inasmuch as the recording and processing of sound is designed to be heard by the listener. Having separated the concepts of sound and hearing, we will first examine sound in vibrational form and then look at the hearing process with reference to acoustics.

Sound is a vibration. Sound is transmitted generally through the air, thus making the air the medium for sound vibrations. The reason air is a good medium for sounds is that it is elastic, it can move and vibrate with the original source of the sound.

The sound source, a voice, musical instrument, or some other source, sets the air in motion. The pressure of the vibrating source causes the air to vibrate in waves going out from the sound source. These pressure-produced ripples, called "sound waves," radiate from the source, are transmitted through the air and are heard by the listener. Therefore the sound the listener hears is actually the result of a sound wave.

Sound waves, depending upon the source, vary in ways which allow the listener to determine what the sound actually is. The characteristics of the wave differ,

therefore the sound is different. The three characteristics of a sound wave which affect the listener's perception of it most are: amplitude, frequency, and composition.

### Amplitude

Figure 1\* graphically depicts a sound wave with reference to time. The height of the wave is referred to as amplitude. If we think in terms of force, the greater the pressure of the motivating force (sound source), the greater the amplitude. If a small stone is thrown into a pool of water, small ripples occur. When a large stone is thrown into the water, much larger ripples go out over the surface. This is precisely what happens with sound wave amplitude. The greater pressure the motivating force exerts on the air, the greater amplitude the resulting sound wave will have.

The importance of amplitude to the recording engineer is that the greater the amplitude of a sound wave, the louder the sound. Because we are constantly surrounded by sound, when a specific sound, such as that of an instrument, is to be "picked-up" for recording, the louder that sound is with respect to the surrounding stray sounds, the better the recording of that sound will be. Technically,

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\*All illustrations are located in Appendix.

this is referred to as the signal to noise ratio. The signal is the original, intended sound. Noise is the surrounding extraneous sound, which could include room noises, wind, other movements, and even background hiss and hum found in the pick-up and recording equipment itself.

The best recording attempts to reduce the signal to noise ratio to a minimum and thus get the cleanest or most accurate reproduction of the original sound.

We should recognize here that, often times during a record session or a mixdown session, tapes are played back at a very high volume. These high amplitude sound waves, although useful in capturing and creating just the right sound the producer has in mind, can also be damaging to the ear mechanisms over a period of time. Caution should be used in order to preserve the sensitivity and quality of hearing for the recording engineer.

### Frequency

Frequency is the measure of the sound wave of the number of cycles per second. Notice in Figure 1 that the sound wave rises above the center line, crosses the line, goes below the line, and then begins the process over again. A full cycle occurs when the wave goes from the zero line through the positive and negative sweeps and returns again to zero. When we plot this against time, we

can see how many times this occurs during one second and thus tell the frequency of the wave in cycles per second.

In honor of Heinrich Hertz, a physicist who did a great deal of pioneering work in wave physics, the frequency measurement unit is now called the hertz. One cycle per second equals one hertz, 1,000 cycles per second equal one kilohertz, and so on

Frequency is important to the recording engineer because the frequency of the sound wave determines the pitch of the sound we hear. The higher the frequency, the higher pitched the sound; the lower the frequency, the lower the sound.

We noted earlier that sound and hearing are independent and yet related. The human ear is capable of picking up and analyzing a great number of complex combinations of sounds at different amplitudes and frequencies; however, there are limits. Typically the range of human hearing goes from twenty to 20,000 hertz. Usually the very lowest and the very highest of these limits cannot be heard by many people. Notice on Figure 2 the range of sound frequencies and the types of sounds which occur in each range.

When referring to frequency, we often speak of treble, midrange, and bass. Many instruments the engineer uses will refer to the specific frequencies of a sound or band of sounds, or will simply use treble, midrange, and

bass. Refer to Figure 2; notice that treble frequencies are usually those above 5,000 Hz and are characterized as "high" or brilliant sounds. Midrange, which covers many instruments as well as the human voice, goes from 200 to 5,000 Hz. Base frequencies go from 200 Hz to the lowest audible sounds of the bass fiddle and certain powerful electronic instruments.

The bass frequencies provide power and fullness to the total sound of a recording. Midrange frequencies tend to dominate the sound spectrum. They are the most energy-related sounds and will often times cause fatigue and aggravation after long recording sessions which deal with these sounds. The treble frequencies give clarity and crispness to the total sound. Particularly the upper treble range is emphasized in today's recording industry. Care must be taken to avoid too much emphasis or the recording could sound too sibilant or acquire too much hiss.

It is important for the recording engineer to know the limitations of his or her own hearing and be aware that not everyone hears the same sounds in the same way. Frequent hearing checks are a wise part of the engineer's regimen to do the best possible job.

### Composition

The third factor to influence a listener's perception of sound is the composition of the sound waves themselves. A sound wave which is a pure, single tone, sine wave, as in Figure 1, sounds very sterile and, particularly at higher frequencies, very piercing. These sound waves are usually electronically generated and are often used as test signals for audio equipment. The reason that you can tell the difference between a French horn and a bassoon is not that they play at different frequencies (they are in the same frequency range) but rather that the composition of the sound waves they produce is different.

Several things affect the composition of a sound. The amount and type of harmonics present in a sound cause the sound to vary greatly in the listener's ear. An harmonic is a sound wave produced simultaneously with the original frequency sound wave. Harmonics are multiples of the original and may occur singly or several at a time and at varying amplitudes.

For example, a center frequency of 160 Hz played on a bassoon will also produce a sound wave of 320 Hz, the first harmonic, and a sound wave of 480 Hz, the second harmonic, and so forth. The number and the relative amplitude of the harmonics an instrument produces give it the sound we recognize as that instrument.

Other sources, such as multiple center frequencies and interference, can contribute to the distinct character of a given sound. The way a sound is generated--plucked, drawn with a bow, or by forcing a column of air--gives it a distinctive character called "timbre." Noise is the ultimate interference generated by clashing sound waves and unwanted sound sources.

We have now discussed the differences in sound sources, sound waves, and sound propagation. Now to complete our understanding of the basic principles of sound and hearing we will discuss the hearing world of acoustics.

### Acoustics

Sound, direct from its source, must travel in waves to the listener's ear. This process, which we have discussed under ideal conditions, in reality is not perfect. There are interferences along the way. These interferences and the way the room "sounds" affect how we perceive the sounds. Detailing these effects on sound and hearing is the science of acoustics.

Sound waves, as they propagate (spread out) from a vibrating source, go in many directions. Because of the elasticity of the air medium, these sound waves bounce off or are absorbed by objects and walls in the room or area. Since sound wave propagation is a physical wave effect, the reflected sound will reach the listener at different

times and from different directions. The effect of reflecting and bouncing sound waves on the actual sound is dramatic. A sound wave may bounce around the room for a long time, causing a spacial distortion--the room sounds hollow. Or it may be absorbed almost completely, making it sound "dead" or flat. There is no ideal structure for all recording situations for all instruments. All good sound studios contain a variety of room environments to obtain the best sound from certain instruments and voices. Specific instances of surface and miking principles are discussed in Chapter II.

Figure 3 gives you a rough idea of how sound waves are reflected to the listener in an enclosed area. The actual reflections of a sound wave decay (fade out) after a period of time. The effect these reflected waves exert on the human perception of sound is called psychoacoustics.

### Psychoacoustics

Whereas acoustics is the objective analysis of sound wave propagation and reflection, psychoacoustics is the subjective effect of differing acoustic environments on people.

There are three basic types of sound related to psychoacoustics: the direct or pure sound, the reflected sound known as echo, and reflected sound known as

reverberation. Sound enhancement during the recording and mixdown process will always have available some form of device to add echo and/or reverb to the recorded sound. In the creation of "good" or "real" sound, the recording engineer uses these effects to recreate the psychoacoustic perception of reality. The specific way in which this is accomplished will be discussed later.

Direct sound is the pure sound as heard directly from the source. This is the sound which is the representation of the original vibration which gave birth to the sound wave. If this were isolated, the effect would be a very sterile, lifeless feeling to the listener. This does, however, form the basis for the listener's perception of direction and character of the sound.

Echo is the first and most complete reflection of the original sound. Echo waves have hit only one surface before being picked up by the listener; however, there may be many echo waves from different surfaces causing multiple repeats. An echo wave tends to mimic the original sound. In an enclosed space, the echos occur within milliseconds of the original sound. When the size of the room is increased, the time for the echo return is also increased. In the mountains, it is possible for an echo to take several seconds to bounce back to the source. The size of the room, the amount and time of echo, help the listener to "place" the sound. It may feel very intimate in a

small room with short echo return and fast echo decay, while long echo decays feel like a large room or a concert stage.

Echo adds life and realism and fullness to the sound and is often used in recording by being added to the final product.

Reverb, a short term for reverberation, is the later sound reflection which has usually been reflected more than once before it reaches the listener's ear. Reverb is a sympathetic vibration as sound waves rebound from surface to surface. Reverb is also used to create realism, depth, and force in a recorded sound.

The well designed sound studio uses a variety of surfaces for different instruments so that the original recorded, direct sound is clear and realistic. Sometimes during the actual recording, and often during the mixing sessions, the recording engineer also uses echo chamber, echo production units, and reverb synthesis units to add these vital characteristics back to the original sound to produce a quality, realistic, and full sound in the final product.

## CHAPTER II

### MICROPHONES AND MIKING

Virtually all sounds, with the exception of those produced by some electronic instruments, must be processed through microphones before anything can be done to them. A microphone is merely an energy translation device. A microphone translates the sound pressure impulses it encounters into corresponding electrical impulses which can be handled by other studio processing and recording equipment. Quite simply, microphones are the ears of the recording studio.

While all microphones do essentially the same thing--change sound energy into electrical impulses--the method and character differ from microphone to microphone. Microphones are classified according to their internal structure, their pick-up pattern, and their response. A solid understanding of these methods of classification is essential to the engineer in the selection and placement of microphones to achieve the best results.

#### Internal Structure

A microphone is an electromechanical device. Sound pressure causes something to move in sympathy with the original waves, which in turn causes electrical currents

to be generated. The three internal structures used in professional (high-quality) microphones are: dynamic, ribbon, and condenser. There are other structures (crystal, carbon) that exist, but these are of inferior quality.

### Dynamic

Otherwise known as the moving-coil microphone, the dynamic microphone uses a thin diaphragm of a flexible material attached to a coil of wire suspended in a magnetic field to produce an electrical output. Sound pressure waves entering the microphone encounter the diaphragm and cause it to vibrate. This vibration causes the coil to move in the magnetic field and thus create an electrical current whose signal represents the original sound wave.

The dynamic element is one of the most rugged and dependable of the microphone internal structures, and is capable of producing a good quality response even under high stress conditions. Sound pressure levels (the intensity of the sound waves) can typically go higher before distortion occurs in dynamic elements than in other types of internal structures.

### Ribbon

The moving element in a microphone with a ribbon internal structure is a very thin strip of metal suspended in a magnetic field. The sound pressure is allowed

to contact the ribbon itself and cause it to vibrate. This causes an electrical signal to be produced which represents the original sound wave. Ribbon microphones are very sensitive and offer a very good frequency response. These microphones are also very fragile. A physical bump or a sudden high-pressure sound wave can easily damage or destroy the element.

### Condenser

Also called a capacitor microphone, condenser elements are the only configuration which require an external power source. The name and power source are there because the moving element in this type of microphone is the plate of an electrical capacitor. A small electronic circuit is connected to the capacitor in the microphone which detects any change in its value. When a sound pressure wave moves the pick-up plate of the capacitor, its value changes. These changes are detected by the electronic circuit and the resultant electronic signal represents the original sound wave. The power source is needed to energize the electronic sensing circuit.

Condenser microphones can be powered by an internal battery or an external power supply, usually called a phantom power supply. Any problem in the power source, a weak or dead battery, or a malfunctioning phantom supply,

will cause distortion or complete loss of the signal from the microphone.

This type of element can also attain very good frequency response and is very rugged. Because of the type of pick-up used, this element can achieve the greatest frequency response. It is also the costliest microphone type.

### Impedance

The last internal consideration of a microphone is its impedance. Impedance is an electrical property of the microphone which resists current flow. Typically, microphones are classified as being either high or low impedance. High impedance microphones present a great resistance to current flow, whereas low impedance microphones present little resistance. In the recording studio, low impedance microphones are used almost exclusively. A low impedance allows long microphone cords to be used and eliminates much low frequency interference from lights and other electrical devices.

High impedance microphones are used for small, hand-held tape recorders and similar devices. Because of their susceptibility to interference and the difficulty of using long cords, high impedance microphones are not generally used in the recording studio.

### Pick-Up Pattern

A microphone, much like an ear, does not pick up sound waves equally well in all directions and at all distances. This "pick-up pattern," is unique to each microphone design. Even though there are individual variations, microphone pick-up patterns fall into several general categories. Pick-up patterns are shown on a polar graph.

Figure 4 shows a polar graph with the center representing the microphone itself. There is no pattern on this graph; however, you can see how the microphone is positioned. When a pick-up pattern is imposed on the graph, as in Figure 5, it is possible to tell the area of maximum microphone pick-up and frequency response. Sounds occurring within the envelope are said to be "on-mike" and give the best, fullest response the microphone is capable of delivering.

The three major microphone polar patterns used most often in studio recording are omnidirectional, unidirectional, and cardioid. It is important to know a microphone's pick-up characteristics in order to select and place a microphone properly for recording. Manufacturer's literature is available which contains information on a microphone's internal structure, frequency response characteristics, and its polar pick-up patterns.

### Omnidirectional

The omnidirectional microphone, shown in Figure 5, is one of the oldest pick-up patterns. As its name implies, an omnidirectional microphone responds well to sounds in the  $360^{\circ}$  sphere around the microphone. Omnidirectional microphones tend to pick up sounds from all directions, and therefore give a feeling, to the final tape, of the room in which the recording took place. Because of this quality, these are popular mikes for live recordings and group recordings. Generally, when a specific sound, instrument, or voice is to be recorded an omnidirectional microphone is not selective enough. These are also generally rather cheap to produce.

### Unidirectional

Figure 6 shows the dramatic difference in pick-up from the omnidirectional. These microphones best pick up what is in front of them. This does not completely exclude other sounds, but emphasizes in force and quality the sounds located in the envelope depicted on the graph.

The unidirectional microphone is good for spotting a specific sound and is used widely during multitrack recording sessions where several instruments or vocals are performing at one time, but each is desired to be heard and recorded separately. The disadvantage is that there is little feel of special acoustics, thus the recordings

must be remixed to create the acoustic and psychoacoustic perceptions needed in the final product. This process is, nonetheless, used very extensively and is discussed in greater detail later in this book.

### Cardioid

The cardioid pattern, a more recent modification of the unidirectional, is a very popular multipurpose pattern. Figure 7 displays the typical cardioid pattern. The cardioid retains most of the unwanted sound rejection qualities of the unidirectional microphone, but allows some of the warmth and a small sense of ambient sound to come through as well. This type of microphone is often preferred to pick up vocals and several instrument types.

### Response

Figure 8 gives a good idea of the frequency range of many instruments as compared to the voice. The way a microphone responds to, or picks up, each frequency is called its frequency response. Frequency response is very important in selecting a microphone.

A sample frequency response graph and specification chart is included in Figure 9. Notice the frequency is listed along the bottom axis while the response, the given output at each frequency, is listed along the vertical

axis. The resultant graph allows the engineer to see how a microphone responds to the whole range of frequencies.

Generally, the flatter the response curve over the range of frequencies the microphone picks up, the purer the sound. Other effects are possible and are often used when the frequency response curve is not flat. A rising response curve will cause the sound to be brighter, sometimes to the point of being "tinny." A bump (a slight rise at a specific frequency) in the upper vocal range (1,500-4,000 Hz) will add greater presence to a voice, and sometimes gives better clarity to a selection.

Many unidirectional and cardioid microphones contain a circuit or structure that rolls off (reduces the frequency response of) bass frequencies. This feature is helpful in reducing room noise, rumble, and the proximity effect (the phenomenon that occurs when these microphones are used in a close miking situation). When a unidirectional or cardioid microphone is used to pick up a sound close to the source, the reinforcement of bass sound waves causes an unnaturally strong bass response. This enhances that "close" or personal feeling in a vocal, but causes an unbalanced response. A microphone with a bass roll-off allows close-up vocal work without frequency response distortions.

Selection of a microphone according to its frequency response should be tailored to the results the

engineer is seeking. The response curve should cover the range of the instrument or vocal to be miked. If a pure sound is desired, a flat response over that range is the best. Bumps, roll-offs, rises, or falls, in the response curves affect the sound of the microphone and can be used to the advantage of the engineer to obtain the desired sound.

The Shure Sm-58 microphone, for example, is often used as a single-performer vocal microphone. The Sm-58 has a bass roll-off as well as a bump in the midrange vocal frequencies. These features give a personal feel to the singer's voice. At the same time, it produces a crisp, clear sound which increases intelligibility and listening pleasure.

#### Environment and Response

The ultimate goal of an intimate knowledge of microphone characteristics, as well as the way they respond, is to achieve the sound pick-up that is most accurate and at the same time most pleasing and realistic to the ear. This is achieved in the studio by:

1. Selecting the correct microphones for the job according to their internal structure, pick-up patterns, and response--the way they sound.
2. Positioning the microphones properly to pick up realistic sounds which also provide as much of

the acoustic feel for the instrument or voice as possible, but to reduce unwanted noise to a minimum.

3. Providing an environment that is acoustically conductive to the normal perception of that instrument or vocal.

Most studios will contain a variety of environments. Some sections will be very live (hard, angular surfaces designed to reflect sound waves back and fill in some of the psychoacoustic need of fullness). These areas are frequently used for piano, organ, and a few other such instruments.

Much of the studio's area is usually dead (sound absorbing). Carpeting and padded walls and ceilings provide an environment which virtually eliminates reflected echo or reverberated sounds. Strings, woodwind, and percussion instruments will usually be recorded in this environment.

Isolation booths are often used for drums and electronically amplified instruments to prevent interference with other instruments or vocals which are being recorded at the same time.

The well-trained engineer will use a variety of environments, microphones, and placement techniques to achieve a realistic and pleasing sound from the various instruments and vocals that are being recorded.

### Microphone Technicalities

Before discussing actual microphone positioning techniques, we must understand several technical considerations. Most studio microphones use XLR connectors. These three-pronged connectors are typical of low impedance installations, and should be used exclusively. An electronic instrument may be patched into a system, with appropriated adapters, through the same wall patch panels as the microphones. Internal patches will differ in the control room, a topic to be discussed in Chapters III and IV.

Connectors, cords, and microphones should be handled with greatest care, and condenser microphone switches should be in the "off" position when not in use. Physical abuse leads to deteriorated frequency response and eventual destruction.

Many hand-held type microphones have a built-in windscreen, a foam piece which helps protect the element from wind blasts or "popping." Additional windscreens are often used over the outer wire mesh to further reduce breath noises associated with close miking techniques.

Microphones may be mounted on stands, on booms, or suspended from the ceiling. The mounts should be sturdy and resilient. At the same time, the mount should be isolated from shock or sound waves which could be picked up

during the recording. Foam or rubber suspension systems are often used to isolate microphones from environments, especially when it is necessary to mount a microphone on a hard surface or to a part of the instrument.

### Miking for Recording

This section is designed to give a general idea of some techniques for miking specific instruments to obtain a good sound. Not each illustration will be appropriate for every situation, but will vary from recording session to recording session depending upon the desired sound. These illustrations simply provide a starting plan. Also remember that, with the basic principles and microphone knowledge in hand, an engineer is free to experiment to achieve an environment and microphone set-up that best suit a particular situation to get just that "right" sound.

The microphone placements discussed here are assembled with the assumption that the recording is done in a studio with the appropriate environment and setting, and that these are being recorded for multitrack work to be mixed and overdubbed and equalized during and after the recording session. Numbers refer to the Figure in the Appendix.

Piano-10

Flute-11

Reeds-12

Brass-13

Acoustic bass-14

Acoustic guitar-15

Amplified instruments-16

Vocals-17

Drums-18

## CHAPTER III

### SIGNAL PATH FLOW PRINCIPLES

Multitrack recording is a complex, three-stage process which involves signal processing from the pick-up, overdub processing, and the final stereo mixdown onto the master tape. The most comprehensive way to become knowledgeable in the use of these processes is first to understand the "signal path flow chart." A signal path flow chart allows the engineer to understand how the signal is:

1. generated at the source;
2. inputted (patched) to the control device  
(usually the console);
3. outputted to outboard effects devices, or;
4. outputted to tape recorder;
5. re-routed back through the console for overdub;  
and
6. outputted to the final tape for mixdown.

And so, a signal path chart shows the engineer how to get the signal from place to place easily. Accordingly, he must build the signal path chart clearly in his mind until it becomes second nature to him. This helps the engineer to master the workings of the studio to make the facilities work for him and to produce the desired output.

Each studio is different. An engineer must learn the signal and control path of each studio before he can work effectively in that studio. The general suggestions which follow parallel the six objectives listed above to aid in the understanding of how most studios are put together and how they operate.

### Signal Generation

Signals are generated in two ways. In the studio, an instrument or a vocal is picked up by a microphone. The microphone generates an electronic signal which represents the original sound. This low impedance signal is fed into the three-wire microphone cable that is plugged into the wall outlets in the studio itself.

The second way a signal may be generated is by using the output of an electronics instrument. This output may be either a microphone-level output (meaning that its signal is the same in amplitude and impedance as that coming from a microphone) or it may be a high-level output. If it is a microphone-level output the instrument may be plugged directly into the outlets in the studio. If it is a high-level output, an equalizing transformer must be inserted in the line to match it to a microphone unit.

In the case of a high-level output from an electronic instrument, it is also possible, if the studio is

so equipped, to plug it directly into a studio outlet designed for a high-level input and eliminate the need for a transformer; but other patches may need to be changed, in this case, as noted in the following section.

Once a signal is generated by either of the above methods, the output is patched into the studio outlets and is transmitted through the wiring to a patch panel in the control room. The first step, then, in understanding signal flow is to determine where and how a signal is generated, then to determine where the signal ends up in the control room.

Typically, the wiring in the studio and control room are parallel numbered or lettered. In other words, a microphone plugged into wall input three in the studio will be available for use in the control room at number three on the patch panel. Accordingly, the engineer knows which microphones or electronic instruments in the studio correspond to which numbers or letters on the patch panel in the control room. This set-up is at the discretion of the engineer and, therefore, must be noted at the beginning of each session, since it will change from one recording situation to another.

### Input Patching

Once a signal is into the control room, the engineer has the option of what to do with it. A patch panel

consists of rows of 1/4-inch jacks arranged and labeled so that an engineer, using connecting 1/4-inch cords called patch cords, can plug the signals coming from the studio into something else. These signals may be low-level from microphones or high-level from other sources. For convenience, we will combine them simply into "signals." There are two general choices of pieces of equipment that can be selected as recipients of the input signals from the studio.

The control console is the most often used recipient of the studio signals. The microphone or high-level line signals are patched, one signal per one input, into the control console. Each input controls one microphone or high-level input from the studio. Using this arrangement, the signal travels, unchanged, from its source into the control console. The advantage of this direct patch is that signal purity is maintained. The disadvantage is that any effects which the engineer may desire to add to that input must come from effects built into the console or from outboard gear patched between the console output and the recorder.

The other option is for an outboard effect piece of equipment to receive the signal from the studio. Chapter VII deals with outboard devices. If some processing of the signal is desired before the signal is routed to the control console, a studio signal can be patched from the

panel directly into the outboard instrument and then patched into the control console. This allows the original signal to be changed (equalization, reverb, etc.) and then sent to the console. In either case, the original signal generated in the studio is routed, either directly or through some outboard equipment, through the patch panel into the control console for processing.

At this point in the signal path flow chart, the engineer enjoys control of all signal inputs from the studio at the control console. The third step is the routing of the signals to their next destination.

#### Output to Outboard Devices

With each of the input signals controlled through the console, the engineer must next decide which, if any, of the signals from the console should be routed through modifying equipment before going to the tape recorder. The outputs of the console--there may be from 2 to 30 or more--are either connected to a patch panel like the inputs or connected permanently to other equipment.

In the console, the original input signals are processed. Their amplitude, equalization (if desired), and special effects (if desired) included in the console circuitry, are controlled and placed on the desired output channels. The number of output channels the console contains determines the number of output signals which can

be routed. Therefore, although a console may have 28 input channels, meaning that 28 possible input patch signals can be used simultaneously, if there are only 16 outputs, there are only 16 output signals which can be routed to other equipment. The original signals can be combined within the console in ways described in Chapter IV.

Once the signals have been assigned to the console outputs, the engineer decides if any of the outputs should be run through outboard gear. If so, the output channel to be used is patched from the output patch panel to the gear to be used. The console output signals are all high-level line signals; therefore, care must be taken to assure that any signal input to outboard devices used will accept a high-level line signal. As discussed, input patches, line and microphone signals are not interchangeable, and must either be matched microphone to microphone and line to line or to a transformer which matches the signals.

The signal, after having been altered in the outboard equipment, must be patched back either into the console (through a special jack provided for that purpose) or be patched directly into the tape recorder.

#### Output to Tape Recorder

Once the signal has been processed by the console and the outboard equipment processing, if any, the

resulting line level signals next must be routed to the tape recorder. In a full-service studio, this is normally a 24-track recorder (it can accept and record up to 24 different input signals simultaneously).

Not all studios offer this number of input possibilities; some studios have fewer tracks on which to record, some have more. A console should have at least as many outputs as the number of tracks available on the primary tape recorder. A more detailed discussion of tape recorders appears in Chapter VI.

The output patch panel, like the input patch panel, is a series of 1/4-inch jacks which are labeled to indicate which console output or tape recorder input they represent. The engineer patches the desired console output to the desired tape recorder input. Although there is no real electrical difference in selecting the order in which the inputs are selected, it is usually best to organize the sounds logically on the tape for future use. For instance, many engineers will put all the percussive instruments on adjoining tracks, the vocals on adjoining tracks, etc. This process helps simplify the signal path flow chart which the engineer is developing in his mind.

Figure 19 gives a simplified pictorial representation of the signal path flow chart which has been built during the first four steps. These four steps are involved in the tracking sessions. These sessions involve

setting the studio to record sounds onto the tape to be used at a later date. Several tracking sessions may be necessary to get the primary sounds onto the tape which are to be used later.

The following steps five and six deal with studio set-up for the signal path flow which occurs during overdub and mixdown sessions.

### Rerouting for Overdub

The overdub session(s) occurs after the initial tracking is finished. Overdubbing is the process of adding sounds from free tracks onto the original material from the tracking session. The sources of the material which may be added are:

1. other instruments or vocals in the studio,
2. already recorded sounds which are now run through an altering outboard device and recorded on another track,
3. a mixture of several sounds already on the tape to a different free track on the tape.

The overdub sessions are used to flesh out the sound, to add fullness by adding special effects, adding other instruments, adding other vocals, or combining instruments to achieve the right sound. The overdub session signal flow is a little more complex than tracking

because both the input and output of the tape recorder must be used.

Notice the changes between Figures 19 and 20. The output of the tracks of the tape recorder which were recorded during the tracking session are plugged into the console to be used as a cue line, sometimes referred to as fold-back. This feature allows the overdub instrumentalists and vocalists to hear the previously recorded material and play or sing in time and pitch with it.

This also means that the tape recorder is required to be set so that the tracks being fed to the console cue are being played, while the tracks which are being fed with the new sound from the studio through the console are being recorded.

Patch panel set-up is similar to the set-up described earlier for tracking. These are the parameters:

1. All inputs from the studio are routed as before.
2. Tape recorder outputs are plugged into console inputs as desired.
3. Console outputs containing new studio material are routed to unused tape recorder inputs.
4. Console outputs containing previously recorded material are not plugged into the tape recorder. The cue function on the console

sends these signals to the studio headphone system.

A special case exists when overdubbing occurs without new studio sounds. In this case, some of the tracking outputs from the recorder are routed from the input patch panel through some outboard device and then into the console. This set-up allows the engineer to add to the recording a modified version of something already on the tape. In this case, the outputs of the console which carry the modified signal do not go to the cue system, but are patched (output patch panel) into the tape recorder on an unused channel. This signal path set-up will allow the engineer either to combine modified sounds with the original tape or to mix several sounds on the tape to a free track on the tape.

One other variation should be noted here. Some studios have a completely separate cue (fold-back) system, separate from the console. In such a situation, the tape recorder outputs which would have gone back to the console to be used as the cue signals are routed to the cue system instead. This frees the console from the task of serving as a mixing control as well as a cue control.

The engineer must repeatedly go over the connection set-up to clearly see the signal path flow in a studio before tracking and overdub sessions. Failure to do this will cause the engineer to be inefficient, time-consuming,

and eventually to destroy a recording by getting inputs confused with outputs.

### Outputting for Mixdown

After all the sounds have been recorded, the work tape (the 24-track in this case) is ready to be mixed down onto a final master tape. This process requires two tape recorders, one to play back the original tape and the other to record the stereo product.

Figure 21 shows the configurations used to accomplish proper signal flow during mixdown. The outputs of the tape recorder are patched into the inputs of the console. The console outputs are patched to the inputs of a two-track master tape recorder on which the final tape is recorded.

The signals, typically up to 24 in a full-service studio, are fed into the console and are combined at the engineer's discretion into two outputs which are recorded on the final tape. Additional outboard effects can be added at any point along this process by patching between the console and tape recorder at the input or output patch panels as described above. Effects may be added to one or several tracks if patched in at the input patch panel; however, when patched in at the output (a little used method) the effect occurs on all sounds on one or both of the output channels.

The studio monitors, not mentioned in this section, are controlled and fed from the console controls. Since most studios have this feature hard-wired in, the control of the monitor mix is at the console and not through a patch panel. This means that anything which is patched into the console through the input patch panel can be monitored in the control room at the command of the engineer.

It is critical the engineer get to know the signal path flow of the studio he is working in order to operate efficiently and to achieve good results. The suggestions in this chapter are a general starting point and should be used to begin the formation of a mental road map for quality use for the recording studio.

## CHAPTER IV

### THE CONTROL CONSOLE

The control console is the heart of the recording studio. Through this device pass virtually all of the signals which are processed, changed, or recorded. The console, or control board (board, for short), is a "melting pot" of sound. It is through this piece of equipment that the engineer designs the sounds for recording, overdubbing, and mixing that become music.

Chapter III, dealing with signal path flow, shows how the board is in the center of all studio operations. Proper understanding and operation of the console is essential if the engineer is to produce quality material efficiently.

#### Inputs and Outputs

Beginning at the heart of the console, the mixing bus, the function of the board becomes clear. The engineer uses the board to control and combine input signals into the number and types of output signals desired. This function is accomplished by mixing buses.

Each console has its own set of specifications, number of possible inputs and outputs, special effects abilities, and optional features. The number of inputs

and outputs is part of the design of the console. There are as many buses as there are outputs. This is important because a bus is an electrical pathway for signals, and mixing is the process of putting the desired inputs onto the desired output buses.

What all of this means to the engineer is that if, for instance, there are 24 output buses and 30 inputs in a particular board, he may select any combination or all inputs to go to any or all 24 outputs. Any of the output buses may have connected to as few as zero or as many as thirty input source signals. What's more, a single input source can be put on as many buses as desired by the engineer. Therefore, one input signal could be assigned from one to twenty-four different output buses.

As stressed earlier, there can be only one input signal per console input. If the console mentioned above were connected in a studio, the input patch panel would allow only one input signal to be plugged into an input channel of the board at any given time. A total of 30 inputs could be controlled at one time because there are 30 input channels.

Within the control board any number of signals can be connected to an output bus. Even though only one input signal can be patched into an input channel, any number of signals in the console can be switched onto an output bus.

### Line/Microphone

A given input channel on a console is usually able to accept either a line (high-level) signal or a microphone (low-level) signal. Many studios will have separate input patch panels which allow the operator to feed in either a line or microphone signal. A channel can never be used to control both signals at the same time. There is usually a switch located at each input channel which must be set to allow the console to accept either the line or the microphone level input which is being fed from the input patch panel.

It should be noted that if two or more input signals are to be combined into one signal, this mixing must occur within the console. It is not wise to attempt to combine signals by using adapters outside of the console. The sound quality will deteriorate and no individual control of each input signal is possible if the signals are combined before the console. The board's main function is to allow the operator to combine, at will, input and output signals with the best quality results possible.

All consoles serve the mixing function. Some make it easier to comprehend by their physical design, but some also have more capabilities than others. Each engineer has a specific console he or she may like better than others, sometimes because of the layout and ease of control, many times because of the sound quality which the

unit delivers. All consoles mix, but not all sound the same. Each board imparts its own quality and texture to the sound. One engineer may not like that sound, whereas another will. This is an esthetic judgment, obviously, that cannot be reflected in the frequency response charts for the console.

### Cue and Monitor

Most consoles have additional buses which are not connected to the regular outputs. The cue (fold-back) and monitor buses allow the engineer and the performers to hear what is happening.

The cue bus is connected to the headphone fold-back system in the studio. This bus allows the engineer to feed signals from the console to the instrumentalist and vocalists. During a tracking session, the engineer may feed some of the other sounds being picked up in the studio which they cannot hear well (drums, for instance, which may be isolated in a booth) to the performers on the cue system.

The fold-back system is very important during overdub sessions. The performers need to hear a mix of the material which has been recorded up to that point, as well as to hear themselves. The engineer must feed the console inputs coming from the tape recorder to the cue bus. He must also feed the inputs from the studio to both the

appropriate output bus to be recorded and to the cue bus to be heard (within the pre-recorded material) by the performers.

The amount of each input which is fed to the cue bus to the performers is a matter of personal preference. The engineer must be sympathetic and yet firm in working with the performers. Too much fold-back will sometimes cause a performer, especially a vocalist, to hold back so that the sound becomes hollow or weak. Not enough of the rhythm instruments can cause the performer to lose tempo and fail to stay within the beat. Not enough of the basic tone-setting instruments can cause pitch problems. There is no set way to mix the cue system. The engineer must try to give the performers what they think they need while also adjusting the mix to get the best performance from the artists.

The monitor bus is connected to amplifiers and speakers in the control room. Any signal connected to this bus will be played through the monitor speaker system in the control room to be heard by the engineer. This is used not only during tracking, overdub, and mixing by the engineers, but is also the way the engineer can demonstrate to the producer what is happening, what it sounds like, and what it will sound like when the process is finished.

The monitor system usually has its own volume control, and also offers the capability of having headphones plugged in for listening. The relative levels on the monitor system are controlled through the system input pots described below.

### Control

The control device is a device used to modify and route audio signals according to the wishes of the engineer. Because each board differs in placement and identification of control devices, it is impossible to detail here which knob or switch to flip to accomplish a specific desired result. Thus, this section deals generally with the operation of a recording console. Specifics of operation for a particular board and studio set-up can be obtained from manuals and information provided by the manufacturer and equipment distributors. Please refer to Figure 22 as a general guide throughout this section.

### Input Controls

All input signals come into the board and are coupled to input controls. These controls, usually of the sliding variety, control the amplitude of the input signal that is delivered to the rest of the circuitry. The signal is processed through a small amplifier which changes the signal, line or microphone, into one signal which can

be worked with in the board. At this point, after the signal has been inputted into the console through the input patch panel, a specific control, a slide pot, will now control the amount of that signal which the console can use. If the pot is off or "dead," no signal goes through the board on that channel. The engineer has control of the amount of each of the signals through each of the channels which the board is processing and can therefore change the strength of each signal individually.

#### Assignment Controls

Once the signal has passed the input pot control, the engineer must decide where to route the signal. Each channel will have a series of switches which allow the engineer to select which of the output buses will receive the signal from that channel. It is possible to assign that channel to go to one or more outputs at a time. It is also a wise practice to "clear the board," that is, turn off all unused inputs and outputs, before starting. This not only eliminates some confusion, but also keeps unwanted stray noise from being placed on a bus that is being used.

During a tracking session, the engineer will usually select two or three outputs for each input. One will go to a track of the tape recorder on which the engineer has decided to put that particular sound. Another will go to

the control room monitor so that the sounds can be heard as they are being recorded. A third output may be to the cue system to fold-back to the performers.

### Pan Controls

One of the most valuable controls used extensively during a mixdown session is the pan pot. This control allows the engineer to fade between two output buses. Stereo recording rarely has exactly the same sound coming from both the left and right channels. The pan pot, somewhat similar to the balance control on a home stereo unit, allows the engineer to select how much of the signal being processed through that particular input channel will go to one output bus and how much will go to the other. This control allows any degree of balance shift--from all left, through the center with equal strength to both, to all right channel output.

There is at least one of these controls on each channel of the console, and the engineer can, therefore, "position" each sound that is being mixed onto the final tape. This is one of the methods the engineer uses to compensate for the lack of acoustic or psychoacoustic reality in a multitrack recording. Since few of the instruments and vocals which appear on the final recording were ever in the same room at the same time, and the recording situation is not a concert stage, the engineer

must design, re-create, an acoustic sound in the final tape that makes the listener feel a homogenous musical performance. The pan pot helps in this effort by "placing" instruments in stereo space so that the listener perceives the instrument or vocal as coming from a specific place.

### Onboard Equalization

Most boards are equipped with some equalization built into each channel. This means that the frequency response of each input signal can be altered at the board itself. An engineer may choose, depending on the quality of the equalization circuits built into the board, to equalize an input signal with an outboard instrument or with the onboard equalization. Or he may choose to allow the signal to remain as is. Equalization is another of the sound tailoring tools the engineer has at his disposal. Equalization, an expanded term for a tone control, is the process of selectively altering frequency response. The engineer can increase the treble response of a singer that sounds a bit "muddy" or muffled. A more complete discussion of equalization is found in Chapter VII, which deals with outboard equalizers.

Console equalization, "EQ" for short, is most commonly found in two forms: equalization pots and filter switches. EQ pots allow the engineer to control a band of frequencies and either increase or decrease the frequency

response of the input signal in that band. The greater the number of bands, the greater flexibility the engineer has in tailoring the sound in the precise way desired.

The other EQ control typically found on console inputs is the filter switch. A filter switch, when activated, will cut or boost a specific band of frequencies on that input. Like the EQ pots, the greater the number of switches, the greater the flexibility, and hence the better control of the sound to obtain the desired output.

Filters are most often used to eliminate stray noises, such as unwanted hiss on the high and/or hum and rumble from interference on the low end of the frequency spectrum.

Engineers using consoles without a good variety of equalization built in will frequently have the signal equalization done by outboard EQ devices. In signal processing, it is very important to get the right sound, and equalization is a major way engineers achieve that goal. If a console is used in conjunction with an outboard equalizer, the engineer will either use controls on both or will set the console for flat response. Flat response means that, to the extent that the electronics of the specific console will allow, the frequency response of the original input signal remains unchanged as it passes through the console.

### Monitor and Cue Controls

Many times a console will have a set of switches and pots on the monitor and cue output buses which is separate from, and in addition to, the regular output bus assignment switches. These allow the engineer to assign an input to the monitor and/or cue buses as well as regular output buses for recording. The pots on the monitor and cue controls allow the engineer to feed a separate mix of inputs to the studio and to the monitor system in the control room.

### Master Controls

The number and function of master controls will vary from console to console. All boards have at least one master fader which simultaneously increases or decreases the levels of all of the inputs. Usually, the engineer sets the master pot to achieve the best signal-to-noise ratio, then rarely changes it. The master pot is used, especially during overdub and mixdown sessions, to set the proper output level for recording, without changing all of the input pots up or down. Thus a proper mix of all inputs is maintained.

Other master controls may vary a particular set of output buses, may vary monitor and cue buses, or may vary the fold-back and monitor system volumes. The specific

function of a console and its controls should be mastered before attempting any studio operations.

### Metering

The final element in the console function is the meter. VU and/or peak meters on the console give a visual indication of the signal strength of each output channel and the master or mixed channels. An engineer must trust his ears to determine the appropriate mix of several instruments and vocals, but it is impossible for an engineer to tell by listening whether the right level of signal intensity is going to the recorders.

The input pots control the strength of the signals through the amplifiers in the console. The meters on the console allow the engineer to see if the output levels for a particular output bus are too high or too low. Levels that are too high, too intense, cause the electronics in the console and in the tape recorder to distort the signal. In other words, when the signal is too high the recorded sound is not a true representation of the original signal. When the level is too low, the recorded sound will pick up a lot of extraneous noise, hiss, hum, etc. This causes the signal-to-noise ratio to go down and the result is a poor quality recording.

The VU meter (see Figure 23) has been the most commonly used meter over the years. It gives a visual

indication of signal intensity in volume units (arbitrary signal strength divisions). With advances in semiconductor circuitry, the VU meter may be mechanical or electronic. Light emitting diodes, LEDs, are used in electronic meters to give a visual indication. Any signal which sends the indicator into the red, above 100, is subject to high-level distortion. A good rule of thumb is that for a specific signal, the level should be set at the input fader so that the greatest signal intensity (loudest sound) will cause the meter to go to 100, but not over. This method helps prevent low levels as well. Generally, the registration should be in the 60 area over half of the time.

The peak program meter (see Figure 23) is a variation of the VU meter. Labeled in decibels (a signal strength measurement), this meter shows the loudness of the peaks (the highest strength signals). The peak meter is useful especially when recording percussion instruments which have a rapid signal attack and a swift signal decay; in other words, a quick, loud signal. In this situation, the VU meter would not respond to the change in level fast enough for the engineer to see a signal which may create too high a level. The peak meter would register the intensity of the peaks in such a signal and allow the engineer to adjust the input faders to give a signal with the appropriate levels.

All the information in this chapter is intended to provide a starting point and foundation for the understanding and operation of multichannel recording studio consoles. There is no substitute for practice and in-depth study of the particular console and signal path flow of the studio in which an engineer is working. The specification manuals for the studio and console should be required reading for the competent engineer.

## CHAPTER V

### CUE SYSTEMS

Studio cue systems are very important to the engineer and to the performers. The two main cue systems in a studio are the monitoring system in the control room and the fold-back system in the studio for the artists. Both of these systems are fed and controlled from the console itself. A closer examination of each will aid in understanding and use of these systems.

#### Monitor System

The monitor system is an amplifier (two-channel) and speaker set permanently mounted in the control room. The quality of this system must be the best since it is used to hear what is being recorded or mixed. The engineer and the producer, and occasionally the performers, evaluate miking, performance, signal processing such as equalization and echo effects, and final mix through this system. Ideally the system should be able to reproduce the full sound frequency spectrum faithfully and be able to generate high sound levels without distortion.

The speakers are at the heart of this system. A speaker is, like a microphone, an electromechanical device which takes the signals from the console and amplifier and

produces sound from those signals. The accuracy with which this process occurs is the quality of the system. Just like microphones, no two speakers are the same.

Most speakers are dynamic, moving coil, speakers. There are other types (ribbon and condenser), but the dynamic are the most versatile and cost effective for high quality sound reproduction.

In a dynamic speaker, the signal causes a coil to move in a magnetic field. This coil is attached to a diaphragm which, when vibrated by this coil, causes air movement and produces sound. Because of the complexity of sound waves (see Chapter I), it is impossible for one speaker to reproduce accurately all sounds through all sound frequencies. Therefore, in order to reproduce sound most accurately, several speakers must be used together.

Small, tightly coned speakers are called tweeters. Those are capable of most accurately reproducing high treble sounds. Midrange speakers, usually from six to eight inches in diameter, reproduce midrange sounds with the greatest clarity. Woofers, the largest of the speakers, are used to produce bass sounds with power and clarity.

A crossover network is an electronic frequency dividing device which is used in speaker systems to send a speaker the sounds it can handle best. High frequencies are sent to the tweeter, low frequencies are sent to the

woofer, and midrange frequencies are sent to the midrange speakers. Figure 24 shows a block diagram of a four-way speaker system and the bands of frequencies which are sent to each speaker. This is the most accurate way of reproducing the sound signals from the amplifier.

A speaker system has its own characteristic sound, some of which can be interpreted by its frequency response graph, and most of which must be heard to be evaluated. Many engineers build their own speaker systems to obtain just the right sound. The most important characteristic of the speaker system is that it reproduce the sound as exactly as possible.

The control room is designed to accommodate both the equipment (console, recorder, monitors, and tapes) and the audience (engineer, producer, etc.). The monitor system is placed to distribute stereo sound to the listeners at fairly high volume levels. Speakers are positioned toward room corners and three to twelve feet in front of the audience, usually angled to face the audience. The speakers are set slightly above the head level of the audience. This positioning helps the audience take full advantage of stereo positioning and provides the widest frequency response and truest sound reproduction.

The amplifier for a typical monitor system is a high quality, high power electronic device which is capable of reproducing high signal levels with little

distortion. The frequency response should be as flat as possible, and each channel (there are typically two for stereo) usually is capable of delivering 50 to 100 watts of signal power to the monitor speakers. This high amount of power allows for high volume levels, but also gives good, clean response at lower levels as well. The output level of the amplifier is controlled by the monitor bus output of the console, and can be fed two separate signals (left and right) from the board.

The monitor system sometimes will also have a headphone outlet available. Usually, the engineer can choose to plug headphones into the console outputs, fold-back outputs, or the monitor outputs.

A final addition to the monitor system is a small, low quality speaker. This speaker, typically an Auratone or similar brand, is used during a mixdown by the engineer to see how the product will sound under less than ideal conditions. The studio control room and monitor system are designed to provide near-perfect sound reproduction. Much of the material produced there, however, will be listened to by the public on home equipment and in cars. Therefore, the engineer uses these monitor speakers to hear what the average person will hear through the radio or television speaker. If it still sounds good under those conditions, a quality product has been produced.

### Fold-Back System

The studio fold-back system consists of a small amplifier, connected to the cue output of the control console, and headphones in the studio. Usually, the fold-back system is hard-wired into the studio outlet panels so that headphones can be plugged into the system at various locations throughout the studio.

The fold-back system is simple, its operation is not. The system amplifier is a small, medium quality amplifier and the headphones vary in quality. It is usually best to use as good a set of lightweight, airy headphones as possible. The better the quality of the headphone system, usually the better the artists can perform. Some performers, especially vocalists, will perform with only one side of the headphone on, with the other ear exposed to the room sound. Whichever method delivers the best performance is the best method.

The two variables to be controlled in a fold-back system are the mix (relative amount of each sound) and the headphones. As discussed earlier, the mix will sometimes vary with the sessions and will vary from performer to performer. Most performers will complain they do not hear enough of themselves. It is best to try to achieve a happy medium. The engineer is in control, but he is also trying to get the best out of the performers. To help eliminate timing problems, the engineer may increase the

volume of the rhythm instruments or the tick track (discussed in Chapter VI). If a pitch problem is occurring, it may be wise to increase the volume of the melody instruments or feed more of the performer's own vocal or instrument back to him or her. Ultimately, the mix that elicits the best performance and attitude is the best mix for that session.

During the overdub sessions, fold-back mix is critical. The performers must hear a pleasant mix of the tracks which have already been recorded. They must also hear a good amount of the tick track, if used, and their own signals.

The other variable in fold-back is the volume. As mentioned earlier, performers (especially with punctuated, high energy music) typically claim they want it louder. The dilemma is that if the fold-back is too soft, pitch and tone problems can occur and the recording becomes inferior. On the other hand, if the fold-back volume is too loud, the performers tend to exhibit less energy and the recording lacks vitality.

A good solution is to give the performers what they feel they want, but then gradually increase or decrease the volume to give the final recording what it needs. For example, if a group of high energy musicians insists on having the fold-back from a pre-recorded segment too loud in an overdub session, begin the session as they wish,

then slowly decrease the fold-back volume until their instrumental or vocal performance meets the maximum level of quality which they can achieve.

All cue systems are designed to aid the performers, engineer, and producer to interpret what is being processed and recorded. The engineer then must adjust the miking, equalization, echo effects, and even instrument tuning to achieve the right sound for that selection.

## CHAPTER VI

### TAPE RECORDING

Tape recording is the process of storing representations of sound impulse signals on magnetic tape to be played back on demand. To this point, discussion has centered around the production and processing of electronic signals which represent the original sounds. This chapter deals with the storage and recall of those signals in the process called tape recording.

The basic principle which allows tape recording to occur is magnetism. Recording tape is a strip of plastic which has been coated on one side with a compound of iron oxide. This combination allows the tape to pick up and remember (become magnetized) any magnetic fields which contact or come close to its surface. Further, the iron oxide compound allows small sections, called domains, of the tape to be magnetized without seriously affecting the magnetic status of the surrounding tape.

The recording process begins when the audio signal entering the recorder is processed through its preamplifier. The signal then goes to the record head of the tape recorder. Here, a change occurs as the signal is translated from electrical impulses which represent the original sounds to magnetic impulses which also represent the

original sounds. Therefore, sound impulses have been transformed into corresponding magnetic impulses. With sound signals in this form, the storing can take place.

The transport mechanism causes the recording tape to move across the record head at a constant rate of speed. As magnetic impulses appear at the record head, the tape becomes magnetized at different points and in different amounts representing the magnetic signal. As the tape moves, a magnetic record is formed on the tape of the impulses generated at the head. Therefore, a semi-permanent magnetic impression of the original sound impulses has been stored on the tape.

#### Magnetic Tape

The quality and form of the recording tape varies. The quality is determined by the type of compounds used and the way they are fixed to the plastic strip. Recording tape comes in a variety of qualities which are controlled by its manufacture. Professional recording tape is designed to permit the greatest signal intensity to be recorded accurately and to maintain its quality over a period of time and use. Its quality is also determined by how well the oxide remains glued to the plastic backing, as well as how uniformly the film is distributed on that backing.

The form of recording tape also varies. Even though tape is essentially a coated plastic strip, its width and packaging vary with the intended use. Recording tape varies from 1/8 inches wide for cassette use to two inches wide for 24- or 32-track recording. The width is an important factor in determining recording quality, as explained later in this chapter.

Recording tape is generally either "open reel," "cassette," or "cartridge." Open reel tape, most commonly used in studio applications, is designed to fit professional tape machines, and is the type referred to in the following discussion of tape recording. Cassette and cartridge tape formats require a plastic or metal case in which the tape is permanently mounted on internal reels for ease of operation. Because of size and speed factors, these methods cannot produce the quality and versatility needed in professional studio recording.

The engineer must select the appropriate width and quality of reel tape for the job. Typically the 24- and 32-track recorders use 2-inch tape while 16-track uses 1/2-inch or 1-inch tape (sometimes 2-inch) and the 8-track machines use 1/2-inch tape. Usually the 2-track mastering decks, on which the final mixdown is recorded, use 1/4-inch tape.

Individual preference will often influence tape selection as well as technical specification. The engineer

should try a variety of brands of tape under many conditions in order to discover the brand and quality which does the best job of accurately reproducing the signals which are to be recorded on it.

### Tape Transport Mechanisms

There are two basic types of mechanical apparatus which are used to move the tape across the heads. The mechanical control apparatus uses the force applied to the control knobs to engage the pulleys, belts, and springs to move the tape at a constant speed and tension. This method is generally outdated and is almost never used any more in most recording studios.

The second method, servo-logic control, is used extensively to control tape movement. Servo-logic control couples solenoids and semiconductor circuitry so that the operator need only push a button and the machine automatically engages the appropriate mechanisms to move the tape in the desired manner. Servo-logic control systems are easier, more reliable, and less likely to harm the tape than mechanical control. It is also possible to operate a servo-logic machine from a remote control location. The controls are often located at the console in studio recording.

Figure 25 shows the transport diagram for a typical reel-to-reel tape recorder. The supply reel, which

contains the tape to be used, is placed on the left side of the recorder. The tape is threaded according to the instructions for the individual machine. The tape passes over the heads with the oxide coated side facing the head. A tape is never turned over. Even with home open reel or cassette recording the plastic side of the tape never faces the heads. If this occurs, a muffled, inferior quality recording will result.

After the tape passes the heads, where the recording or playback occurs, it goes between the capstan and pinch roller. These devices control the speed of the tape. The supply and take-up mechanisms just keep an appropriate amount of tension on the tape; the speed is controlled by the capstan and pinch roller. These must be kept clean at all times, along with the heads, to insure accurate sound quality and constant speed. Wow or flutter (variations in the speed which can ruin a recording) are often caused by dirt and oxide build-up on the heads, guides, capstan, and pinch roller.

The take-up reel collects the tape which has passed over the heads and through the capstan/pinch roller assemblies. During the recording process, new information is recorded on the tape. During playback, the information is decoded so that the listener can hear/enjoy it.

In addition to recording and playback, the transport mechanism serves two more functions: fast forward

and rewind. These functions allow the tape to pass freely over the heads while the capstan and the pinch roller are disengaged. Thus, either the supply reel or the take-up reel revolves rapidly, allowing the engineer to go forward or backward quickly to a desired location on the tape.

Normally, professional machines allow the engineer to listen through the play head during rewind or fast forward. This capability permits him to hear what is on the tape and then locate the beginning or end of a selection. A tape counter is usually included, but it indicates only the relative number of revolutions a reel has made. This can sometimes be useful in finding a general area on the tape.

### Electronic Functions

The basic electronic functions of a tape recorder are divided into two major areas. The preamplifier section of the recorder and the tape head section of the recorder.

#### Preamplifier

An incoming signal from the console is processed through the input preamplifier and sent to the record head. There is a separate preamplifier and accompanying VU meter for each input channel. A tape recorder has as many channels (electronic signal pathways) as it does tracks (recorded lines of information) on the tape.

The job of the preamplifier is to convert the signal into a form which can be used to generate a magnetic field by the record head. This signal, depending on the quality of the electronics, is an exact replica of the original audio signal entering the tape recorder on that channel.

Another job of the preamplifiers is to supply the appropriate output signals back to the console from the play head during playback. When a signal is reproduced from the information on the tape by the play head, the preamplifier duplicates the signal, putting it in appropriate line level form, and sends it to the console. A single preamplifier on a channel can perform either of these functions but not at the same time. In professional recording, when additional information is added to a tape, it is added to a separate track, not on top of information already on the tape.

### Tape Heads

The tape head, as discussed earlier, is the device which changes electronic signal energy into magnetic impulses and back again. The record head is designed to take signals from the preamplifier and record them magnetically on the tape. The play head reads the magnetic information on the tape and translates it back into electronic signals to go to the output preamplifier. Each

channel has its own input and output preamplifier connected to the record and play heads, respectively.

The erase head is used, when activated, to scramble the magnetic signals on a specific track of the tape and thus rub out any information on that track of the tape. The erase head is only used when recording.

Refer to Figure 25 once more. The order of the tape heads is shown from left to right. As the tape passes the heads, it is erased if the machine is set to record, but is not erased if the machine is set to play a specific track. The tape then passes the record head, where information is encoded if recording is taking place. The play head, which decodes information if playback is occurring, is last. It is possible to record information through the input preamplifier and record head, then listen to the information almost instantaneously, through the play head and output amplifier during the same pass of the tape. This process is important and will be referred to later.

In multitrack recording, there are several signals which can be processed through different sections of the same record or play heads at the same time. Each head has as many sections as the machine has tracks. Further, each section is separated by a small space from the adjoining section so that different material can be recorded side by side on the same tape with little interference.

Figure 26 illustrates a piece of 2-inch tape recorded on a 16-track tape recorder. Each track has a separate preamplifier and has a separate signal recorded on it. Notice that the space between the sections of the head creates guard bands between the tracks of recorded material. These guard bands help prevent "cross-talk" (leakage of the signal from one track onto another). This concept is the foundation of modern multitrack recording. The engineer can select at will which tracks will be recorded, which will be played, and which will remain unused.

### Quality

Two major factors influence the quality of a recording. Assuming the electronics and tape are of high quality, the speed and amount of tape surface used in recording affect the final quality. Because it is necessary to encode a lot of information to get full frequency response from a tape, the higher the tape speed (the greater amount of tape going by the head in a given amount of time) the better the quality of reproduction. Tape speed is measured in inches per second (ips). Standard tapes distributed for radio are recorded at 7-1/2 ips. For recording purposes, master tapes are recorded at either 15 ips or 30 ips. Usually, a multitrack work tape is recorded at 30 ips to obtain the highest quality and

to make editing during overdub sessions easier. The two track masters from the final mix are recorded at either 15 or 30 ips.

The amount of tape surface used is the other major factor influencing quality recording. The more tracks a tape surface is divided into, the smaller each track is. The smaller the tape track, the more inferior the quality. A 16-track recorder using 2-inch tape, other factors being equal, will do a higher quality job than a 24-track recorder using 2-inch tape. To compensate for this fact, an engineer sometimes will use otherwise unused tracks of a tape to record a signal on two or three tracks at once. This increases the total surface area and quality, which is very important with rhythm and bass instruments. Of course, it is not always possible if all the tracks are needed for different signals, but is a good practice when possible.

A final consideration in recording quality is signal level. Each preamplifier has its own VU meter on the tape recorder. Often the engineer will adjust the input pots on the tape recorder preamplifiers so that the VU meters on the console match the VU meters on the tape recorder. The engineer then must only be concerned with maintaining a proper level at the console (see Chapter IV) to ensure that the proper signal levels are being delivered to the tape recorder.

A weak input signal, low VU reading, will cause a noisy, lifeless recording. A high input signal, over 100 on the VU meter, can cause distortion on the recording. Some recording tape, referred to as "hot tape," has a greater headroom. This means that very high level signals can be recorded without distortion. It is generally not wise to try to overload a tape during recording, partly because of the possibility of distortion, and partly because of increase in cross-talk. The engineer should know in advance how much a particular tape and tape recorder can be loaded with signal before distortion occurs, then avoid exceeding that level.

### Operation

Operation of a professional multitrack tape recorder is not difficult. To master and use all of its capabilities completely takes work and practice. Following are some control functions of multitrack recorders which enable the engineer to build a recording.

### Record/Play Head Functions

The separation of record and play heads and use of separate preamplifiers mentioned earlier give the engineer several capabilities. A-B comparisons are possible through the console by listening to the signals on the monitor system which are being outputted to the console,

and then quickly switching to the play head outputs of the recorder to hear the sounds which have been recorded. In a high-quality situation where professional, well-maintained equipment and practices are used, there should be very little if any perceptible difference between the two. This allows the engineer to spot check the equipment, tape and methods to assure quality in the final product.

Another function of record and play head separation is called tape delay. This is a special effect, somewhat similar to reverb or echo, which can be created within the tape recorder. Refer to Figure 25. A signal which is recorded will pass the play head a fraction of a second later. If the engineer takes the playback signal from the play head and patches it through the console and recombines it with the signals that are being recorded, the effect is that of a sound repeated a fraction of a second after it is first recorded. The amount of the playback signal mixed in with the original signal will determine the amount and depth of the tape delay effect. In this mode, the tape recorder is recording and playing back the same track at essentially the same time. A variation of this method which permits the tape delay to be added during mixdown is to route the playback signal from one track so that it is recorded on a separate track. During mixdown, the engineer adds to the original track as much of the tape delay track as desired to achieve the appropriate effect.

## Synchronization

Simultaneous synchronization (simul-sync) or selective synchronization (sel-sync) are terms applied to the process which allows overdub recording to take place. Because of the separation between the record and play heads, a singer or instrumentalist in an overdub session would not be in time with the prerecorded material. The material recorded during tracking would be played through the playback head while the new material during overdub would be recorded at the record head a fraction of a second after the prerecorded material. This effect is fine for tape delay mentioned above, but is unacceptable for in-sync recording. Sel-sync recording, or simul-sync, was developed to eliminate this problem and make multi-track recording possible. The engineer can select which channels are to be processed through sel-sync. A track on sel-sync uses the record head as playback, not the play head. A multitrack record head can be used either to record or play back material. If a channel is in sel-sync the record head is used to play back that channel instead of recording information onto it. This allows the engineer to record overdub material in exact synchronization with prerecorded material on other tracks.

An example will help clarify the value and importance of synchronization. A tracking session involving a piano, bass guitar, string synthesizer, and trumpet has

been completed. The piano is on channels 2 and 3, the bass on channel 4, the synthesizer bass and treble on 5 and 6, and the trumpet on 7. A vocal is to be overdubbed on channels 8 and 9 in time with the music. The engineer will thread the work tape onto the machine and set the levels. Then channels 2 through 7 are set to sel-sync. The outputs (from the record head) of these channels are patched into the console and fed to the monitor systems. The microphone inputs from the vocals are fed through the console to the recorder's channels 8 and 9, which are not on sel-sync. The engineer monitors the vocal levels as the recording takes place. When the session is finished, or the tape is to be heard, all channels on the recorder are returned to normal operation and tracks 2 through 9 have been recorded in sync. Multitrack overdub is impossible, and should not be attempted, without the use of a multitrack recorder with synchronizing capability.

Another function of synchronization is single track correction, known as "punching in." Physical editing of multitrack tapes is not done. If a mistake is made in one section of a track, the engineer simply allows the performer to do that section over again. The recorded tracks are placed in sel-sync, and the performer plays along with it. When the section to be replaced comes up, the engineer punches in the new material by placing that track in record. At the end of the section which was to be

replaced, the engineer "punches out" either by stopping the machine or by taking that track out of record. This method requires a lot of skill and practice and, usually, a little luck.

One other function of synchronization is the ability to bounce tracks. In smaller studios, 8- and 16-track, and sometimes in full-service studios, there is a need for more tracks than there are available. One way to solve this problem is to combine several pre-recorded tracks onto one open track, thus freeing more tracks for use. This process is called "bouncing" or "pinging." On new recorders, the quality of reproduction from the record head is as good as that from the play head. Therefore, to ping tracks 7, 8, 9, and 11 onto track 17, the engineer puts 7, 8, 9, and 11 into sel-sync (so that they do not get out of sync with the other tracks) and routes them through the console to be recorded on 17. The engineer must mix the signals in the proportions desired during this process because the relative levels of the four tracks which were combined cannot be changed once they are bounced to track 17. This is a good method to gain more tracks, but cuts down on the flexibility in the mixdown, because the levels of the tracks which were pinged together are set and cannot be changed relative to one another.

## Digital Recording

Most recording now is analog, which means that the signal in its original form (see Chapter I) is recorded on the tape. But digital signal processing has arrived. Digital processing is the encoding of the original signal into a binary code which represents the signal. This binary digital code, not the original signal, is recorded on the tape. The signal is reproduced by reading the digital code and reconstructing the analog sound.

The digital code, a binary system developed by the computer industry, is simple and capable of rejecting virtually all extraneous signals or noise.

Digital recording eliminates one of the major drawbacks of analog recording. When a sound is recorded on tape, the recording is called first generation. When a recorded sound is recorded (dubbed) onto another tape, it is called second generation. Each time a recording goes down another generation there is a loss of quality. In digital recording, because a code is being passed along instead of a complex signal, each generation is as good as the original. This allows the final product passed along to the consumer to retain studio quality sound.

The advantage of digital recording is that it eliminates noise. The disadvantage of digital recording, at present, is the cost. A new 32-channel digital recorder

costs over \$150,000. Good analog studio recorders and tape can come close to digital quality, but digital is still better. In time, most, if not all, signal processing and recording will be digital.

Multitrack recording forms the soul of the engineer's work. The engineer must be sure to clean and maintain the recorders well to produce a quality product. Many studios carry on a regular program of cleaning and aligning (adjusting) the tape heads and tape path. A wise engineer will use equipment manuals to assure the recorders are in top shape before using them in a recording session.

## CHAPTER VII

### OUTBOARD INSTRUMENTATION

The final element in the technical arsenal that the recording engineer has at his disposal is sound altering devices. They are called outboard devices because they are patched between the control board and inputs or outputs. These devices aid the engineer in designing just the right sound for the right occasion.

These devices allow the engineer to reproduce for the listener the acoustic and psychoacoustic effects which add realism and feeling to the final recording. Instrument positioning through psychoacoustic ambient additions gives a feeling of spaciousness to the recording. Vocal reverberation, echo, and chorusing add depth and fullness to the final product.

Outboard devices also are used to change the real sounds of the recording. "Tinny," "boomy," "punchy," and "woofy," represent only a part of the recording industry jargon which you are not likely to find in a conventional dictionary (if they are indeed there at all), but they play an important role in the dialogue between producer and engineer. The engineer uses the signal altering outboard devices to produce a sound for a single channel (instrument or vocal), or a whole section of sounds

(rhythm, vocal, or entire output in mixdown), which matches the producer's desires.

The use of signal altering outboard effects is an esthetic decision on the part of the engineer. Some ideas concerning esthetic concepts in this area are discussed in the final chapters of this book. This chapter, however, discusses the various types of outboard instruments, how and why they operate, and basically what they do to the sound. Because it is impossible to completely describe a sound solely in words, this information is intended only to aid in the basic understanding and possibilities of signal altering devices. There is no substitute for the engineer's use of, and experimentation with, these devices before recording sessions. A good engineer knows the effect of using these different devices and can therefore call upon his knowledge and experience during a recording session to make the sound of an instrument more "mellow" or a vocal more "punchy."

Outboard signal processing usually takes three forms. Outboard devices change the frequency response curve of the signal, change the timing or pitch of the signal, or change the level of the signal or frequencies. Much of the signal processing is digital (see Chapter VI) since a greater amount of flexibility is gained using this type of processing. Signals that are unprocessed are

called "dry." A signal, after processing through one or more of the following devices, is called "wet."

### Devices That Affect Frequency Response

In the Chapter IV discussion of console controls, equalization within the console was mentioned. Virtually all devices which alter the frequency response characteristics of a signal are called equalizers. The equalizer, an enhanced tone control, is used to change the sound of a signal by placing greater or lesser emphasis on specific frequencies or bands of frequencies.

#### Shelving

Shelving equalization controls are the simplest and least flexible controls. A shelving control, typically a "treble/bass" control, increases the frequency response above or below a preset frequency. A treble control, if it is variable, with a set frequency of 10,000 Hz, will increase or decrease the level of all the frequencies above 10,000 Hz. A bass shelving control will do the same for all frequencies below a certain set frequency. Because this is a very general control, found on many consoles with low-grade equalization, it is not used much in recording.

### Peak/Dip

Peak/Dip equalization is more precise than shelving in that it affects a specified band of frequencies. Peak/dip filters can increase (peak) or decrease (dip) the level of a specific band of frequencies. These equalization controls can be either a filter switch that peaks/dips a band a fixed amount, or a variable control which peaks/dips a band whatever amount the engineer wants.

Peak/dip equalization allows the engineer more control than does shelving equalization, but each control is tied to a specific center frequency. Typically, this type of equalization device will offer three to six separate controls, each set at a different frequency. When a specific control is increased or decreased, the levels of the frequencies around that point are increased or decreased.

Peak/dip filters and shelving filters (high or low pass filters) use peak/dip and shelving equalization. These filters, discussed in Chapter IV, allow the engineer selectively to cut or boost the response of a band of frequencies. They also allow the engineer to reduce the level of very high or very low frequencies, thus limiting hiss and rumble.

### Graphic Equalization

A graphic equalizer, pictured in Figure 27, is a combination of the above types and is widely used in signal equalization. Graphic equalizers are divided into bands of frequencies with a control for each band. The engineer can select which frequency band is to be increased or decreased with respect to the others. The greater the number of divisions, the smaller the band of frequencies and the greater flexibility and control are afforded the engineer.

Figure 28 illustrates the normally flat frequency response graph when a graphic equalizer is used to boost the band of frequencies around 1,500 Hz. The band of frequencies includes all those which have been increased above the center line. The band is narrow if the equalizer has many controls, and is wide if it has few.

The name of the graphic equalizer comes from its design. Figure 27 shows that its controls, arranged in ascending order of frequency from left to right, easily show the engineer what is being done to each frequency band. Its control face resembles a frequency response graph.

Most graphic equalizers also have high pass and low pass filters which cut bass or treble frequencies, respectively. The main level control and bypass switch control

the amount of influence the equalizer exerts on the circuit, from major down to no effect at all.

Most studios will have several graphic equalizers for use in signal processing. Many engineers use a couple of graphic equalizers in the monitor system to tune the speakers to the control room. This allows the listeners to get a true impression of the sounds. It can also help to eliminate fatigue during long hours of listening at high volume levels.

#### Parametric Equalizer

A parametric equalizer is the most versatile equalizer available. This device allows the engineer to select the amount of peak or dip equalization a band of frequencies will receive. It also allows the engineer to determine the width of the band of frequencies to be effected. It is therefore possible, with a good parametric equalizer, to select a specific frequency and greatly increase or decrease the response of that frequency. For instance, a 60 Hz power hum can be cut out by tuning the band width small enough around 60 Hz to eliminate the hum without affecting the quality or sound of the music. Figure 29 shows a notch cut out of the frequency response graph at 300 Hz. This is done with a parametric equalizer.

The parametric equalizer is the most preferred out-board equalizer to correct problems in frequency response of individual signals, as well as to alter overall response during mixdown. The engineer can use the equalizer, patched into the board, to spot specific problems and correct them. If a drum is too "boomy," the engineer can adjust the band widths and frequencies which are causing the problem, and design a sound that is pleasing to the producer. If a vocal is muddy, the engineer can select the right frequency to pull certain sounds up higher and cause the vocal to be heard and understood clearly above the background. An equalizer is used to alter signal frequency response to add clarity, change the character, or eliminate unwanted sounds from an audio signal.

#### Devices That Change Pitch or Timing

The harmonizer, phaser, and flanger are devices which can be used to produce changes in signal pitch and timing. The harmonizer is a digital device which takes an input analog signal and changes it into a digital signal. The digital signal is then altered, according to the control of the engineer, so that the result is a signal which has been delayed and changed slightly in frequency. This effect sounds like a choral ensemble, or a singer overdubbing his or her voice with itself. This

means one vocal can sound like a chorus by using the harmonizer.

A phaser is a little less complex in that it separates the original signal into two parts and passes the original half on unaltered. The other half is changed so as to be out of phase with the original. When these signals are recombined at the output, the sound has a tremolo or wavering quality. This is used in many effects for modern rock sounds dealing with electronic instruments and guitars.

A flanger is an electronic device much like the phaser in that it splits the input signal. The flanger causes the half which is altered to be out of phase and delayed with respect to the original. The recombined sound is used for many non-realistic sounds and special effects.

Each of the above devices is patched through the console and their effects can be added to either the original work tape or used in the mixdown. Many studios do not incorporate all of the instruments mentioned in this chapter, but most have at least an equalization device and a harmonizer. The harmonizer is used to double sounds, but can also be used to produce synthetic reverb. The following devices, however, do a better job of reverb production.

Delay units, reverb units, and echo devices are used by the recording engineer to recreate acoustic and psychoacoustic effects in recording. The delay unit is a digital device which does the same thing as tape delay, discussed in Chapter VI. The advantage of a digital unit is that it is less complex to use and the signal is very good. When using physical tape delay, a second generation recording is made from the original. Each succeeding generation in analog recording means a loss of quality. The digital delay unit eliminates that problem.

Reverb units, one of the most common outboard accessories in recording studios, help restore the desired amount of multiple reflected sound to recorded sound. Reverb units come in several forms, the two most common mechanical forms being plate and spring.

The basic principle for both plate and spring reverbs is the same. The original sound, brought through the console from the tape recorder, is used to cause the plate or spring to vibrate. This vibration is then picked up by electro-mechanical devices and converted into audio signals that are multiple reflections of the original signal. When this delayed signal is combined with the original, in proportion decided by the engineer, a feeling of spaciousness is added to the recording.

Reverb units are capable of producing delays from a few milliseconds to three seconds. This expands the

listener's perception from the feeling of a small, rather dead room up through that of being in a large concert hall.

There are also digital delay, reverb units. These units, unlike the harmonizer, are designed digitally to alter the signal in order to imitate the physical reverb. The effects of the two types of reverb are similar. However, subtle differences in sound, response, and signal variation cause definite preferences among recording engineers. It should be noted that the digital unit is also more costly than the physical units.

Echo devices are also available in digital and physical configurations. Figure 30 shows a diagram of the inside of a physical echo chamber. The sealed chamber can be adjusted with baffles to control the sound wave reflection between the speaker and microphone. Most engineers still prefer the sound of a physical echo chamber.

Echo allows the psychoacoustic effect of placement to take place in a recording. When the engineer, by ear, adjusts the amount and type of echo added to the original sound, he creates a feeling of spacial placement in the mind of the listener. The instruments sound as though they are placed closer or further away from the listener. The engineer can use equalization and other timing and pitch control devices to create a musical recording that sounds to the listener as though it was recorded in

concert--perhaps better, even though the instruments and vocals were never in the same room at the same time. The engineer must use these signal processing devices to design musical sound which is both believable and what the producer ordered.

### Devices That Change Level or Frequency

Signal compressors and limiters are devices used to vary signal levels. Good compressors and limiters are something close to automatic volume controls. As the input to a compressor increases, its output decreases. This allows the engineer the freedom to get the most out of a signal without the fear of distortion or muddy soft sounds. The compressor is used in line with the console during recording.

A Dynamic Sibilance Controller (DSC), is often used in conjunction with a compressor for vocals. This piece of equipment mutes the effects of high sibilant sounds (such as "ch") from the vocal. A compressor and a DSC, when combined, help produce clean, crisp vocals without annoying sibilant sounds.

A limiter is more strict than a compressor and serves a slightly different purpose. The limiter has an adjustable threshold level above which it no longer increases in output, even though the input level continues to increase. To avoid pushing signals too high, an

engineer must watch the VU meters and adjust the console input faders accordingly. The limiter helps the engineer by not allowing the signal to go too high. A quick increase or decrease in level is to be avoided by the engineer. The limiter must be adjusted so it does not quickly increase or decrease the signal level being recorded. A good limiter that is well adjusted should be seen and not heard.

Dolby, or more specifically dbx for multitrack recording, is a frequency compression and expansion system. The concept of dolby noise reduction systems for tape recording was developed several years ago and has spawned several forms. The concepts behind the systems are similar.

The frequency spectrum being inputted to the tape recorder is compressed by the dbx system. That means all frequencies are reduced dynamically as they are taped. This compression includes low level tape noise as well. When the tape is played back, the dbx system sends it through an expander. Because the signal was compressed before recording and expanded later, the original signal is outputted, with one major exception. The original tape noise (hiss, hum, etc.) was at a low level. When the expansion takes place at the end of the process, this low level is many times lower than it was originally. Dbx

results in a much quieter tape because of the reduction of tape noise.

There is a great difference of opinion surrounding the use of dbx in high quality studio recording. Many engineers insist that the dbx causes the recorded sound to have a character different from that of the original, and that the dbx system can adversely affect the high frequency response of a recording. Although some swear by dbx for clean recordings, other argue that it is possible to get as clean a recording on a well-done tape mastered at 30 ips.

Room for personal preference is important in developing esthetic qualities of a good engineer. As a rule of thumb, however, it seems wise to use dbx when dealing with small track widths and slower speeds in recording. When it is possible to obtain excellent results without its aid, do not use it.

There do exist other sound-shaping devices, and more are being developed daily. The devices discussed here are the major ones in use now. The engineer should constantly search for new and better ways to improve signal processing to increase the capabilities of the studio and to deliver just that right sound.

## CHAPTER VIII

### THE TRACKING SESSION

The remaining chapters of this book deal with the esthetic considerations involved in each step of designing a multitrack recording. Esthetics have not been ignored in the technical discussions of Chapters I through VII; however, these chapters provide a closer look at how the engineer makes his or her artistic decisions.

It is impossible to explain thoroughly in writing the esthetic decisions which the engineer must make in the construction of a musical recording. It is the goal of these chapters to explain how those decisions come about, as well as the methodology used to implement them. The final product is a reflection of the character and the skill of the engineer and producer team. These chapters will aid in the understanding of the operation of that team.

Prior to any recording effort, the engineer must realize that he is in the driver's seat. The way a session progresses, for the most part, is due in large measure to his skill and attitude. The engineer's job is to get the sounds the producer wants. The performers will respond better, giving a better signal to work with, when the engineer is competent enough to make the session flow

smoothly. Intimate understanding of the equipment, its operation, and its limitations is essential for smooth session flow.

The engineer is often dealing with strong and sometimes unpredictable egos in the performers. The engineer must never let his own attitude get in the way of dealing with the rest of the people in the studio. A calm, assured (but not proud), and happy character enables the engineer to sooth ruffled egos and impatient producers. A well-mannered engineer is a blessing to both the producer and the performers. Each participant will be willing to give his or her best when the engineer is personable, warm, happy, and confident. A good attitude usually affects the final product with positive results.

### Preparation

The engineer must be well prepared before the session setup begins. He should know, in conference with the producer, what is being recorded when. He should know basically what the producer wants in the final product, and should help in making those decisions which result in the best possible sound. It is extremely important that everything has been prepared before the performers arrive. The tape recorders should be cleaned and aligned, the studio and control room cleaned and ready. Some general

miking of studio instruments may be done in advance to save time when the instrumentalists arrive.

A click track may be needed, depending on the song and instrumentation. This is a track recorded in advance which contains a metronome-like clicking sound to keep the instrumentalists and vocalists in sync. This track is played through the headphone cue system but is not included in the mixdown. It exists only for the benefit of the performers and is not heard on the final product. The engineer should confer with the producer and/or instrumentalists in advance as to the use and speed of a click track.

Most tracking sessions fall into one of two categories. One session may consist of the entire group--all of the instrumentalists and vocalists to be used. This method is seldom used. The more common way of conducting tracking sessions is to record the rhythm section first. This means bringing in the bass, piano, guitar, and drums (depending on which are to be used) prior to bringing in the vocalist(s) and other instrumentalists. The rhythm section is recorded first so that color instruments and vocals to be added during overdub sessions can follow the basic beat and simple melody tones generated by the rhythm instruments.

As the instrumentalists arrive, the engineer places them so that they and their instruments are placed in the

best and most comfortable locations. Placement of the individual instruments is important. Much of the final ambience, discussed in Chapter II, is determined by the instrument environment and microphone placement.

Placement and miking of instruments should parallel the mood and requirements of the song and producer. Starting with the drums, the engineer adjusts microphones and even drum head tuning to achieve the sound that best fits the musical number. The decision to mike a bass guitar or use a direct line input depends on the character of the sound desired. A direct line input eliminates some of the roundness of a bass sound.

The piano sound can be mellowed by using ribbon microphones, or peaked by using dynamic unidirectionals aimed toward the hammers. Guitar sounds can also be altered by aiming a unidirectional microphone toward the frets instead of the hole. These changes should be determined by the character of the song. Esthetically, the engineer adjusts the placement and miking of instruments to achieve just the sound the producer is looking for.

With the microphones and instruments in place, the engineer instructs the instrumentalists to begin playing so that he can set the levels. Final adjustments are made to the miking and placement, and the input fader levels marked and set. These levels will not remain unchanged

through the session, but those set at this time give the engineer a ball-park idea of the range and settings.

The last element of preparation concerns effects. Onboard/outboard equalization or digital signal processing may be used during tracking. Adding these effects will save time and free outboard gear for other uses later during overdub or mixdown. Miking, placement, and signal processing vary according to the song and style of music. The levels should be set and special effects added while the performers are playing the song that is to be recorded or the sound will not be correct.

### Recording

The fold-back system should be adjusted during a dry run to determine if the levels are correct. As discussed in Chapter V, the mix is determined by what makes the performers happy and what makes the performers give their best. Rhythm instruments always cry for more of themselves, while strings often only want to hear simple tones (like a piano) to pitch from. Each instrumentalist or vocalist should be fitted with comfortable headphones and fed a cue mix which helps them do their best.

The tape is then threaded onto the recorder and cued to the blank section to be used. Master console levels are set to zero before taping begins. The recorder should be started in record when all are ready and quiet.

After five to ten seconds, the engineer signals the instrumentalists to begin and simultaneously brings the master faders up to the predetermined levels.

During the recording the engineer is concerned that the performers feel good and that the levels are correct according to the VU meters. An accurate recording must have correct levels. If a live mix for some instruments is being used to conserve tracks, the engineer must balance the mix as well as the level. A live mix (several inputs combined onto one channel during the performance) makes it impossible to change the relative levels of the signals on that track later in the mixdown. A popular phrase, "fix-it-in-the-mix," should not be the rule when tracking. The best possible recording during tracking sessions will net the best possible overdub and mixdown later.

### Wrap-Up

At the end of a recorded segment, the engineer returns the levels to zero before turning the recorder off. This allows smooth beginnings and endings to the recording. When everyone is fairly well satisfied, the producer and performers will listen to the "rough mix" of what has just been recorded.

As the listening session takes place, the engineer combines what he feels are good relative levels of each

of the tracks recorded thus far. The producer may disagree with his mix and request changes in the level or signal processing used. The recording and listening process continues and is refined until the producer is happy with the sound.

Occasionally a cut during a tracking session is just the way the producer wants it, except for one section of one instrument. In consultation with the producer, performer, and the musical score, the engineer works out where the bad section can be altered by punching in a live segment. Punching in is both a technical and esthetic consideration, because the cut-in and out must not be noticeable to the listener. Chapter VI describes the procedure for punching in a recording.

When all punching in is finished, the tracking tape is played again. During the playback, some ideas are formulated about which instruments should be doubled, which instruments should be added in overdub, and what kind of signal processing should be done to the tracks before the final mix.

The tracking for a single song, not including the overdub and mixdown sessions, can take from one to five hours or more. Attitude and smoothness are critical ingredients supplied by the engineer as he performs the technical work to design a piece of music. The engineer

plays his own instrument, the console, in concert with the other musicians to produce the final musical product.

## CHAPTER IX

### THE OVERDUB SESSIONS

The overdub and mixdown sessions provide the engineer with the greatest opportunity for creativity. Although some engineers view overdub sessions as a routine operation to fill the remaining tracks with sounds, a creative engineer enjoys these sessions as a way to build the sounds that become the final product of the mixdown.

The engineer has a great deal of flexibility regarding the order and types of instruments and vocals to use during overdub. The first decision usually made, which, for convenience, occurs at the end of the tracking session, is whether or not to double any of the rhythm instruments or vocals which were a part of the original tracking. If a particular part sounds weak or needs to be filled in, the engineer will often choose to double.

#### Doubling

Doubling is the process of recording the same instrument or vocal on a separate track. When the two are played back together, the result is a fuller, richer sound: a more orchestral sound from instruments and a more choral sound from vocals. Doubling can also be achieved in overdub or mixdown by means of digital

equipment, such as the harmonizer discussed in Chapter VII. There is a different character to the sounds produced by physical doubling and digital processing. The engineer must decide which, if any, of these sounds is to be used in the overdub sessions.

Throughout all overdub sessions, the engineer must use the synchronization functions of the tape recorder to ensure that the tracks being recorded on are in sync with the tracks which have already been recorded. The engineer is also careful to fold-back the right mix (Chapter V) in each performance to get the best from the overdub instrumentalists and vocalists.

### Adding

Overdubbing is the process of adding similar signals to prerecorded signals in order to build a final work tape. Once all doubling signals from the tracking session are finished, the producer and engineer decide what other instruments and vocals should be added for use in the final product. An instrument or vocal may be added in overdub which has already been recorded. Sometimes, the purpose of this addition is not simply to double but to harmonize with the prerecorded material. For example, it is currently popular for a musician to play several instruments or to sing double, triple, or more voices in harmony with himself. This is accomplished by the

engineer adding harmony to the original recording in overdub. One voice can be made to sound like many, even a chorus, using this method.

The usual order for instrumental additions is:

1. Rhythm instruments recorded during tracking.
2. String instruments during overdub.
3. Horn (brass) instruments during overdub.
4. Wind (other than brass) instruments during overdub.
5. Vocals when all instrumentation is finished.

This order is not even a strict rule of thumb. Sometimes the vocalists record with the rhythm during tracking.

Instruments may be recorded in reverse order due to scheduling difficulties. Complete changes may be made in order to achieve the goals of the producer. The engineer constantly must be ready to change and adapt to the different situations. His creativity cannot be stifled by changes in order or setting.

### Strings

The strings will often prefer to work as a section by themselves, with little other instrumentation coming over the cue system. Usually the beat and a piano melody are sufficient for them to pitch from. A very nice effect is to set a string section in a live portion of the studio and mike at a distance. The nature of the reflection will

add a nice effect to the sound. Special signal processing can be added at any time during the tracking, overdub, or mixdown.

It is often desirable to double, or stack, the string sections. Stacking is repeated doubling to achieve a full, rich sound. Some engineers will vary the equalization or digital processing during successive doubles to increase the variation in pitch and time from one performance to the next. This effect, if used carefully, can enhance the feeling of largeness or fullness in the final mix.

### Horns

The horns to be used are usually recorded next. Brass has a tendency to overpower and needs to be strictly controlled. Each instrument is best miked separately, both for ease of correcting mistakes and for blending in the mixdown. Most horn parts are doubled or stacked. Often it is economically advantageous to have a limited number of horn players, if they are good enough, to add other harmonizing parts. This saves time and complexity.

### Wind

The wind instruments are overdubbed in the same way as the horns. Usually high wind (piccolo, flute, etc.) is accentuated with equalization but the top shrill

frequencies are rolled off to make the sound pleasant, not piercing.

Once all the instrumentation has been recorded that is to be used, the engineer will go over the track and listen to see what changes (doubling, equalization, reverb, echo, digital processing) can be made to get the sound close to what the producer is looking for. Most engineers and producers prefer the instrumental part of the selection be near perfect before adding the vocals.

### Vocal

The overdubs added to the work tape are vocals. The setup for the best results depends on the type and number of vocals used. Group vocals, especially those who have worked together before, are best blended themselves. This means that the miking and fold-back are not separated into separate channels. The relative intensity (balance) of each performer is controlled by that performer, not the engineer. This is best for a chorus (background) or an ensemble.

Usually vocal parts and performers are miked separately. Each performer has his own headphone set and microphone (see Chapter II). The engineer records each vocal in overdub on separate channels and blends them in the mixdown. If a sufficient number of tracks are not available to record all vocals and subsequent doubles, a

live mix of several vocals to a single track may be used. The engineer must record carefully, because the blend on a live mix track cannot be changed during mixdown.

As mentioned earlier, it is often desirable to add effects to signals as they are being recorded. The fold-back system should be fed with dry (unaltered) signals, but the recorder may be fed with wet (special effects added) signals to save time and to free tracks and out-board instrumentation for overdub and mixdown.

Vocals will almost always need to be processed with echo and/or reverb. Equalization and digital processing may be used as well. If these effects are used as the signal is first printed, the engineer is free during mixdown to concentrate on other more general effects on the whole tape. Many vocal signals are processed through a compressor to add clarity and bring them out over the background.

When the final doubling, stacking, and processing are finished with the vocals, the engineer and producer listen to the final work tape and the engineer mixes the signals as they are heard. If there is any other processing and/or overdubbing necessary, they are arranged. The work tape is now full and complete, and the engineer begins the final step in completing the sound creation.

## CHAPTER X

### THE MIXDOWN SESSION

The mixdown is the engineer's opportunity to create the sound the producer is after. Here the engineer deals with shades of sound. He begins blending all of the sounds and effects recorded on the work tape with additional processing to come up with the final mix. The engineer is an audio artist who is now painting a picture in sound. If he has done a good job in the tracking and overdub sessions he now has all the raw material required to produce the final product.

A wise engineer will listen over and over again to the work tape. He will adjust levels, add signal processing, and remix many times to determine how much of each signal should go where at what time. He will listen to the differences which occur in the sound by changing the equalization of one or many channels. He will add echo, reverb, digital effects, to achieve just the right sound. There is no substitute for repetition to achieve the goal.

#### Panning

The engineer must paint a sound picture in panoramic stereo. He decides which signals go to the right and which go to the left. Few signals will ever be only

on the right or the left channels. Putting a signal only on the right or left is reserved for special effects or one side of a duet. Most signals are panned (adjusted so a greater amount goes to one channel than the other) more to one side than the other. This positions the sound in the listener's mind. The location of sounds is important as the listener constructs a real image of the music. The engineer must look at the sounds he has to work with. The final product is to be balanced. Panning all instruments left and vocals right sounds unnatural, lopsided.

This process, because it can become complex with many signals, is usually broken down by instrumental sections. The engineer will usually begin panning and balancing the signals in the order suggested during the tracking and overdub sessions.

The rhythm section is the first laid down. Usually, the most percussive instruments, the drums and the bass, are balanced first. Rhythm tends to sound better closer to center, though this is by no means a hard and fast rule. The engineer adds the piano and guitar to round out the rhythm section.

Each section is balanced by itself and a suitable sound obtained before adding other sections. The engineer will usually balance the rhythm first, the strings second, the horns third, the winds fourth, and then add the vocals last.

During each step of the process, sound alterations with equalization or outboard effects may be added to any or many channels. When a digital or echo effect is added to one or more signals, and then panned left or right, the result may sound pleasing on the stereo monitor system. The audience, however, will not always enjoy the advantage of stereo reproduction (most AM radio stations, for instance). Therefore, it is wise to combine the left and right channels into a mono signal and listen to see if the high frequencies are altered. Echo may be cancelled, as well as frequency response altered, when a stereo mix is combined into mono. If this occurs, the engineer must change the way the signals have been altered or panned in order to eliminate the problem.

### Mixing

Whereas panning takes care of left/right placement, mixing creates intensity differences and front/rear placement. Any ambient psychoacoustic effect not printed with the original signals is now added to specific signals. These digital and echo/reverb effects allow the engineer to place the instrument in aural space. When the engineer adds digital phase changes, echo or reverb, an instrument or section is perceived to be in front of or behind other instruments by the listener. Vocals, too, can sound with the instruments or like solos in front of them. Solo

vocals can stand out from the background chorus by adding more reverb or echo to the background. Creating the proper ambient atmosphere in the sound gives 3-dimensional realism to the final product.

The engineer now fine tunes the relative level of each of the inputs. This level will not always remain the same throughout the recording. Different sections of the song will require certain instruments or vocals to be more prominent. The blend is an esthetic judgment which is controlled by the producer's desire. Generally, a good mix gives the feeling that all instruments and vocals are being heard, and none sticks out when it is not supposed to.

The problem of controlling the mix over 24 channels can be very difficult. Some consoles have the ability to remember the mix levels during a mixdown, so that a computer will replay the mix the engineer earlier decided upon. These consoles allow the engineer to mix section by section, then add them together in the final run. Most studios do not enjoy that luxury.

Before the final mix is recorded, the engineer may run it through a low quality speaker system, such as the Auratones mentioned in Chapter V, to see how the panning, special effects, and mix sound through a system like a common car radio. Here alterations can be made to help

the sound reproduce well on good listening systems as well as medium and poor quality systems.

A mixed stereo master is now produced from the multitrack work tape. This two-track tape is the final product the engineer has been working toward throughout the entire recording process. A song may have taken one day or two weeks to get into this form. Many hours of recording and listening are essential before this tape can be produced.

One channel of the two-track master is the left, the other is the right. This master will be evaluated by the engineer and the producer, then re-recorded if necessary. The work tape is saved so that other mixes are possible and changes can be made.

The master tape is the source used in the record pressing process to produce the records and tapes which are distributed to outlets and radio stations to be heard by the public. So ends the process which is the work of the recording engineer.

## CHAPTER XI

### SUMMARY AND CONCLUSIONS

It would be folly to suggest that a novice, having read these chapters, will be ready to jump into the world of the recording engineer. Much time, practice, patience, perseverance, and on-the-job learning are necessary for a person to become an accomplished engineer. A person must love music, hard work, long hours, and be willing to accept disappointment to work as an engineer.

The engineer's life centers around his work. He must be caught up in listening, learning, and practicing the tools of his trade. The field of recording engineering is exciting, challenging, and rewarding to those whose skill and talent make them the audio artists of the twenty-first century.

What does it take to make a career in recording engineering? Here are a few suggestions taken from the lives of people who are engineers today.

#### Recording Engineering as a Career

There is no standard, all-inclusive, programmed formula for the background and experience that make a person a good candidate for a recording engineer. There are well-known and respected engineers in the field today

who have extensive electronic and mathematical backgrounds. At the same time, there are those who have little or no training in electronics. What common characteristics qualify a person to be a recording engineer?

### Musical Background

To be effective, a recording engineer must have a solid appreciation for and understanding of music: its creation, its performance, and its sound. Most engineers play different instruments and/or sing. Most can read music and understand score symbols so that they can communicate well with the producers and performers. All good engineers have an "ear" for music. A great deal of exposure to a variety of music, musical styles, and artists is necessary to be able to hear what makes music sound the way it does; and what changes in miking, instrumentation, mixing, and special effects do to the overall sound of the final products.

The engineer must be able to understand the needs and the difficulties of the vocal and instrumental performers who are working with him. Much of this empathy is generated by the engineer's experience in the field of the performer.

### Technical Background

Although some engineers are skilled enough technically to change the electronics of the equipment with

which they work, this level of the electronic expertise is not shared by all engineers. Contrary to the stereotypic inferences of the term engineer, the recording engineer is so named, not because he is necessarily able to fix or reconstruct the recording equipment, but because he uses the combination of his technical and esthetic skills to construct a final musical product.

The engineer needs to know how a piece of equipment works and how it does the job for which it is designed. He must know what can be plugged in where in order to achieve the best and the desired result. The engineer must be able to correlate the adjustment of equipment characteristics with the related changes in the sound processing which occur.

### Personal Background

Engineers come from a variety of backgrounds and job experiences. The most important qualification is love of music and machinery. You must like the control and the music. Most engineers today did not start out being engineers. Some were musicians, some were technicians, and many were in other, tangential fields. The love and fascination of sound design drew them to their profession. Experience in operation of equipment and in musical performance usually aids in the preparation of the person to be an engineer.

Like people in most experience-oriented professions, the recording engineer who is highly acclaimed in his field is the one who began by taking advantage of small recording opportunities and perfected his technical and musical skills to the professional point of expertise.

### The Future

As digital technology revolutionizes recording and replaces analog, we are reminded of the rapid changes which have taken place over the last thirty years. As new and improved sound systems become part of everyday life, we wonder what the future holds. What marvels wait to replace digital? Automated mixdown controls put much of the work of the engineer in the hands of a computer. Electronic marvels synthesize sounds never heard before. Sophistication replaces drudgery, and greater flexibility to design sounds is placed in the hands of the engineer.

Where is it all going? Will electronic complexity eventually put the engineer out of work? No! The future holds a greater promise for the recording engineer. He will always be an integral part of the music design process. No machine can make the decisions an engineer must make. No machine can exercise human perception or appreciate what other humans want to hear. The equipment will improve, the techniques will improve, the demands will

increase, but the engineer will remain the one who controls the process of musical sound design.

## APPENDIX

APPENDIX

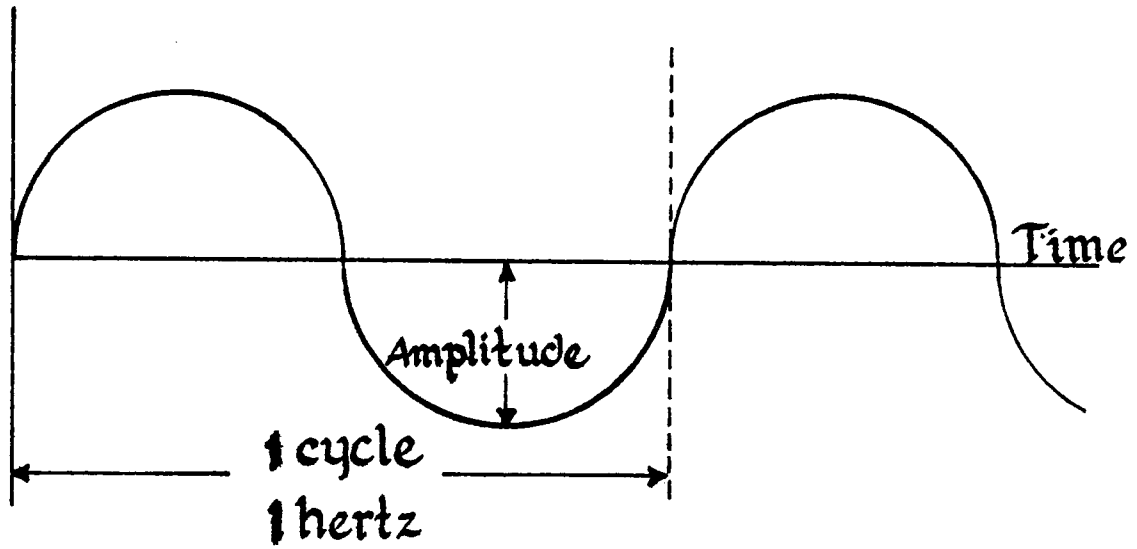


Figure 1. A Sinusoidal Sound Wave.

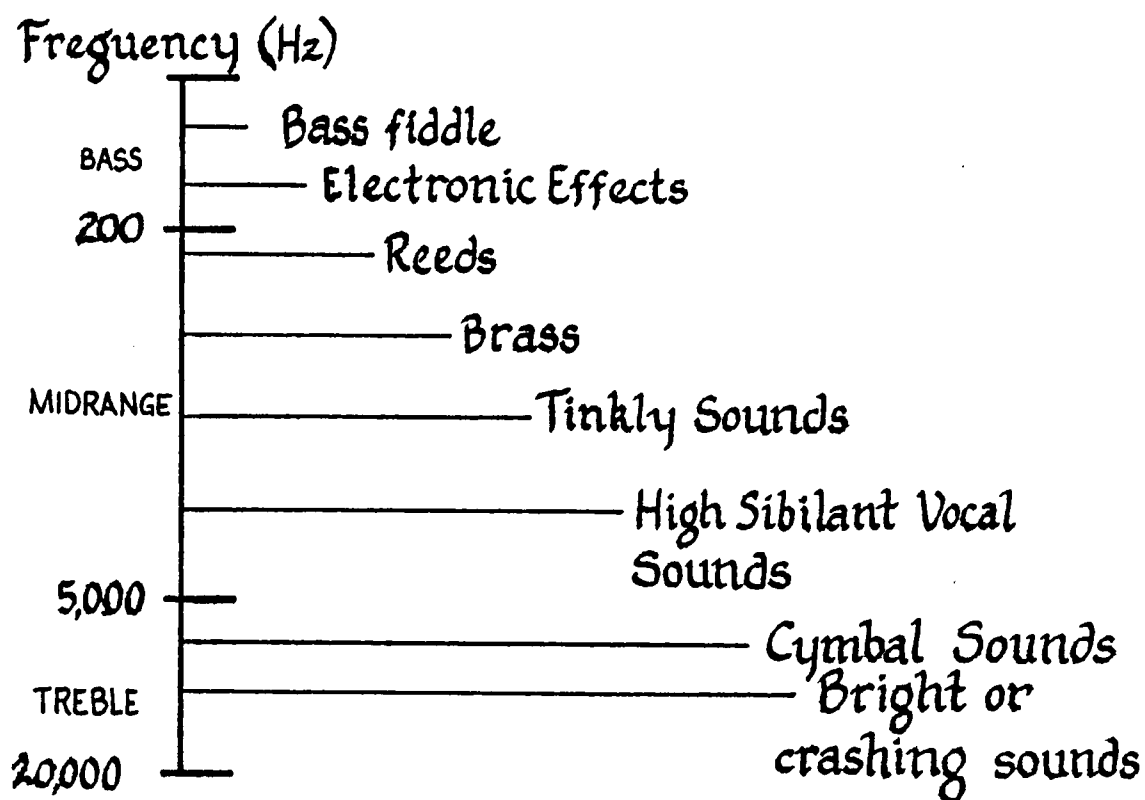


Figure 2. Sound Frequency Chart.

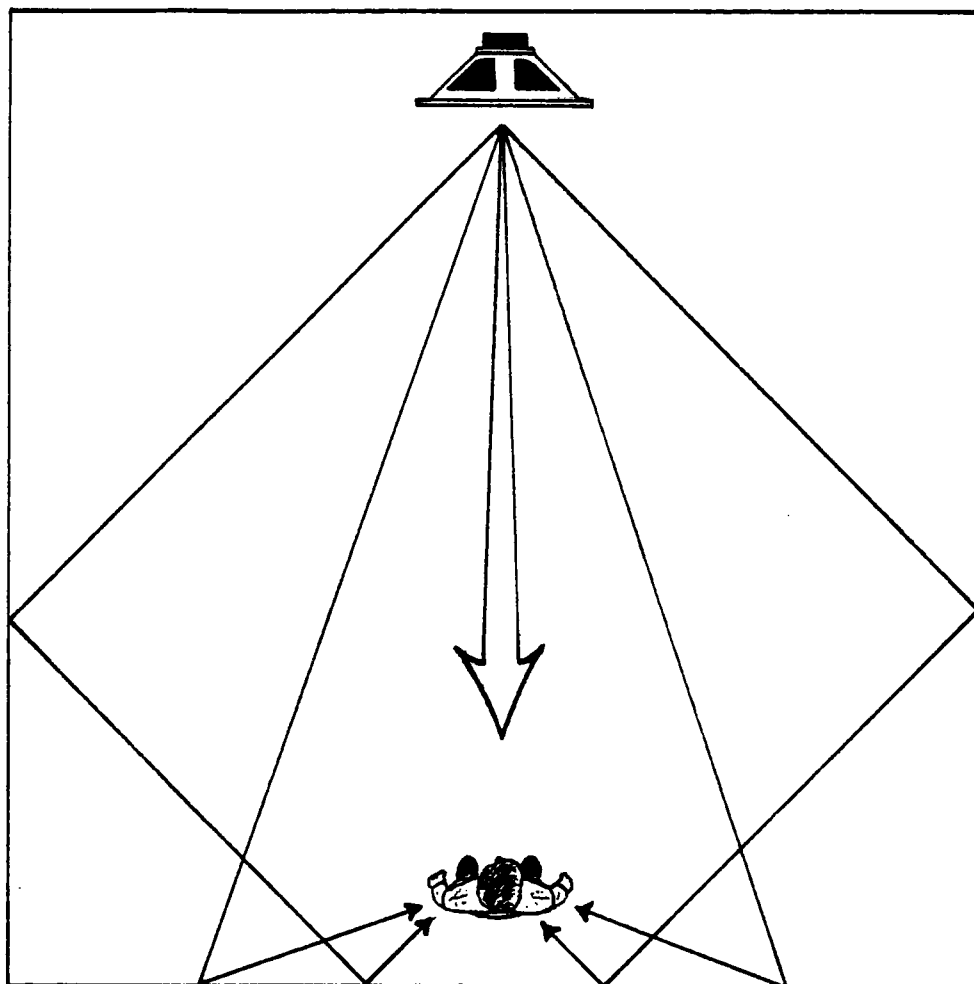


Figure 3. Sound Wave Reflection.

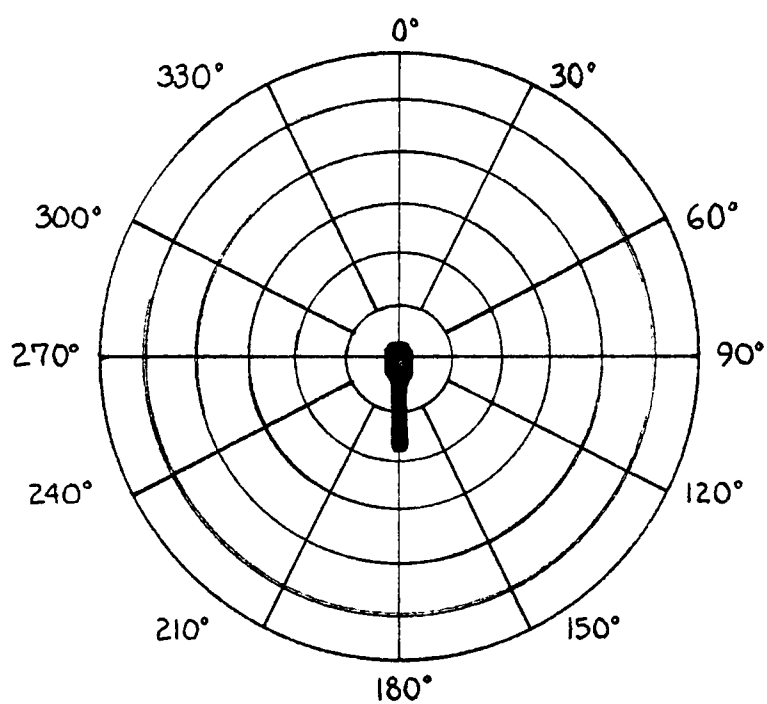


Figure 4. Blank Polar Graph.

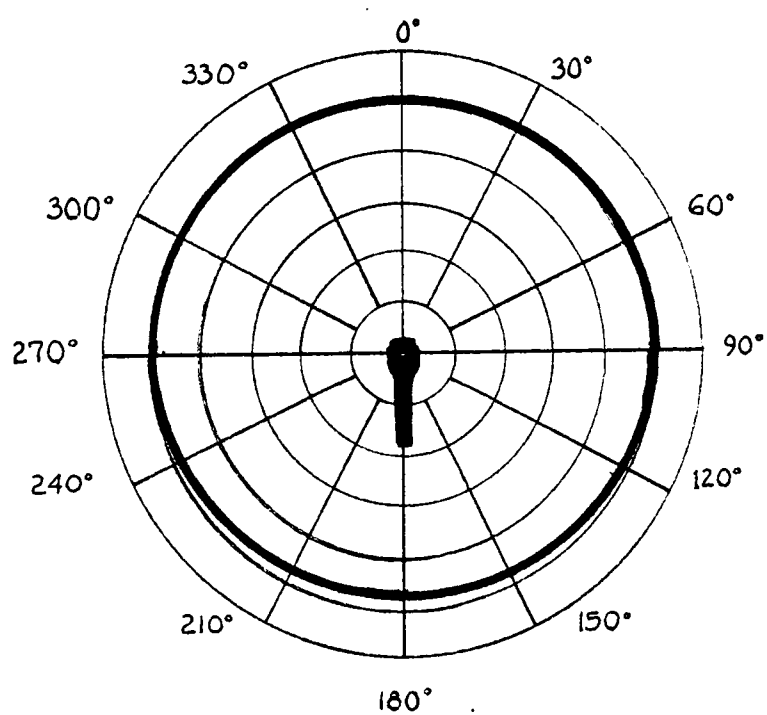


Figure 5. Omnidirectional Polar Pattern.

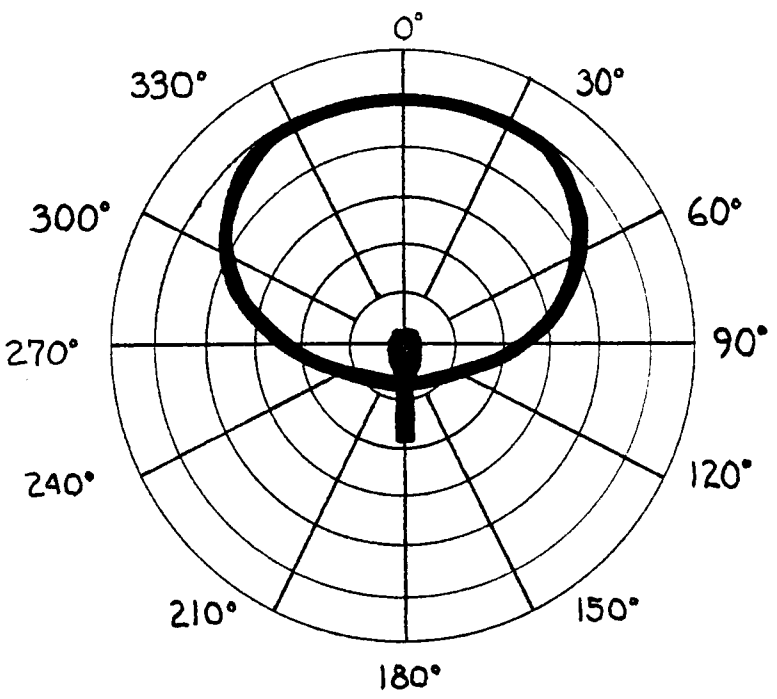


Figure 6. Unidirectional Polar Pattern.

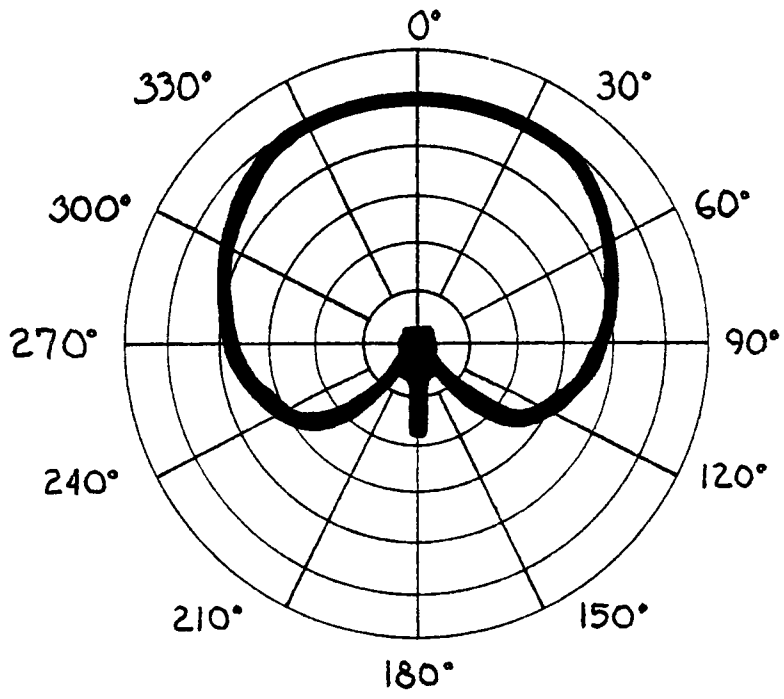


Figure 7. Cardioid Polar Pattern.

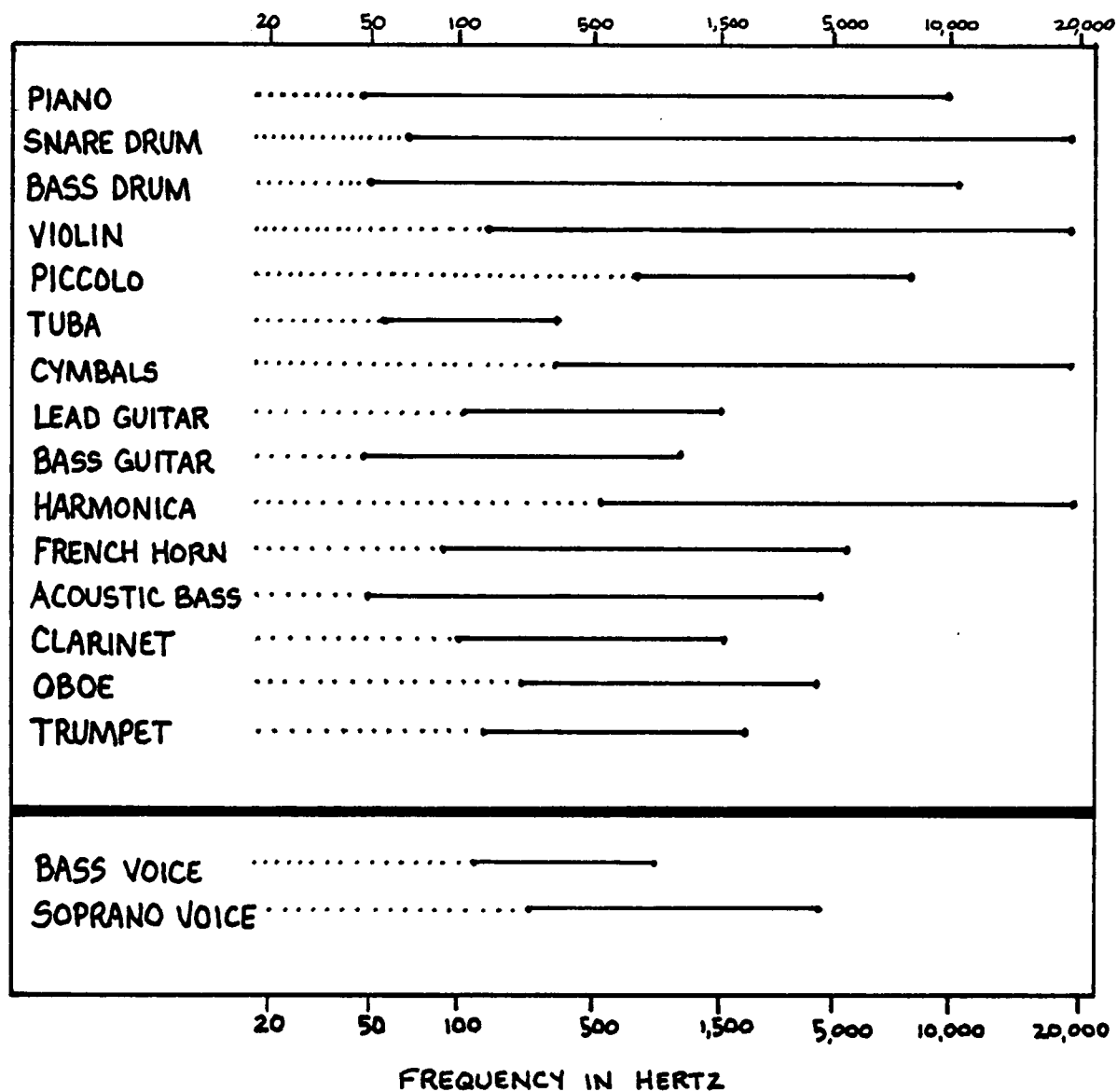
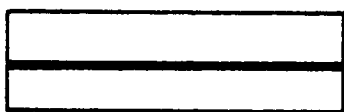


Figure 8. Instrument/Vocal Frequency Chart.

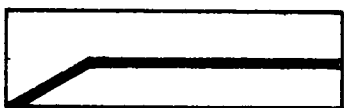
## Idealized Frequency Response Curves



Flat Response



Presence Bump



Bass Rolloff



Presence Bump &  
Bass Rolloff

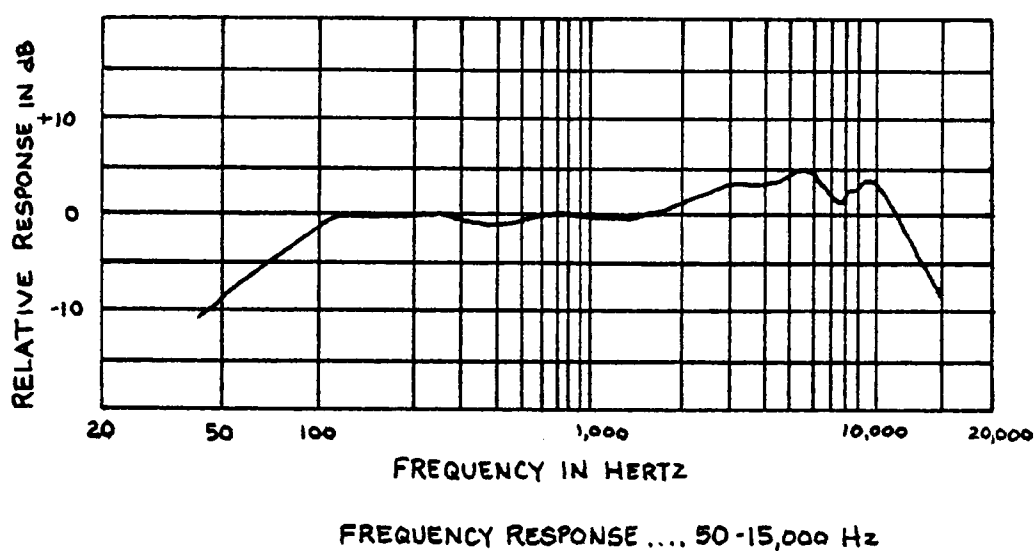


Figure 9. Microphone Frequency Response/Graph Chart.

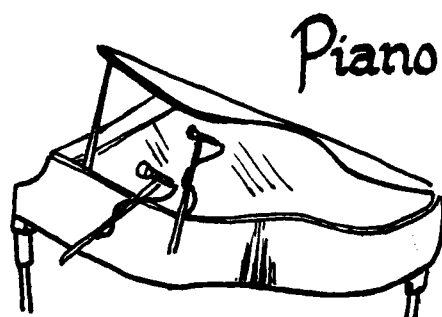


Figure 10. Miking for the Piano.

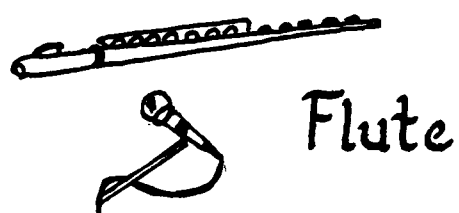


Figure 11. Miking for the Flute.

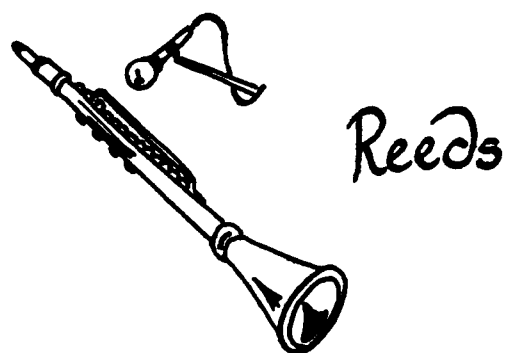


Figure 12. Miking for Reeds.

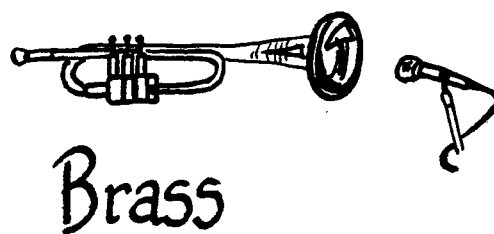


Figure 13. Miking for Brass.



Figure 14. Miking for Acoustic Bass.



Figure 15. Miking for Acoustic Guitar.

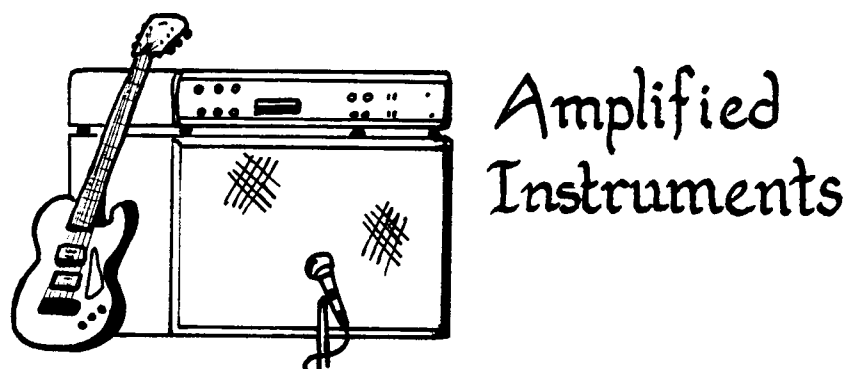


Figure 16. Miking for Amplified Instruments.



Figure 17. Miking Vocals.



Figure 18. Miking Drums for Full Control.

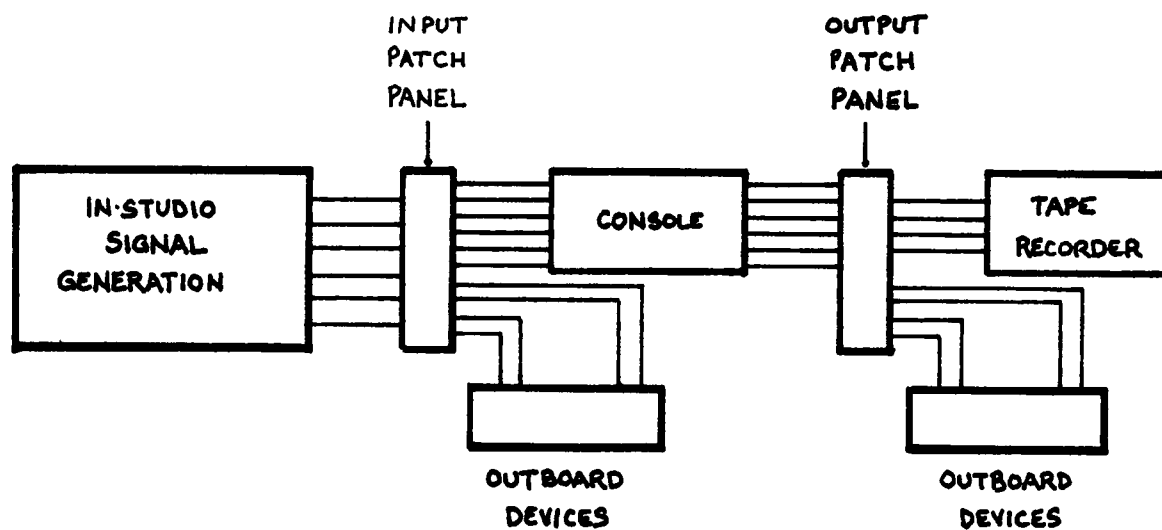


Figure 19. Signal Path Flow for Tracking.

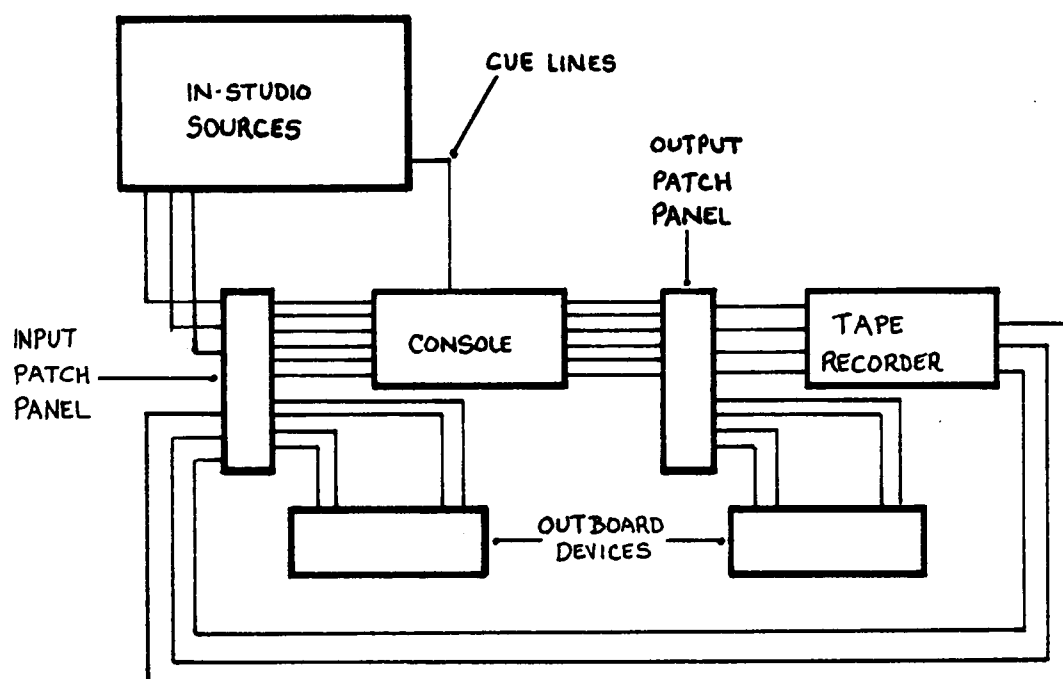


Figure 20. Signal Path Flow for Overdub.

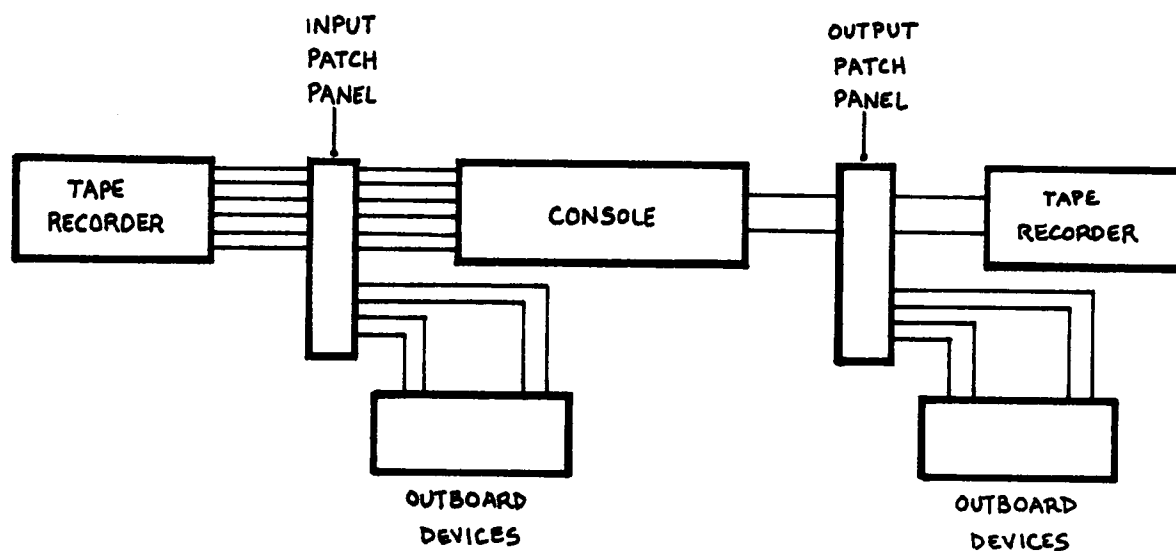


Figure 21. Signal Path Flow for Mixdown.

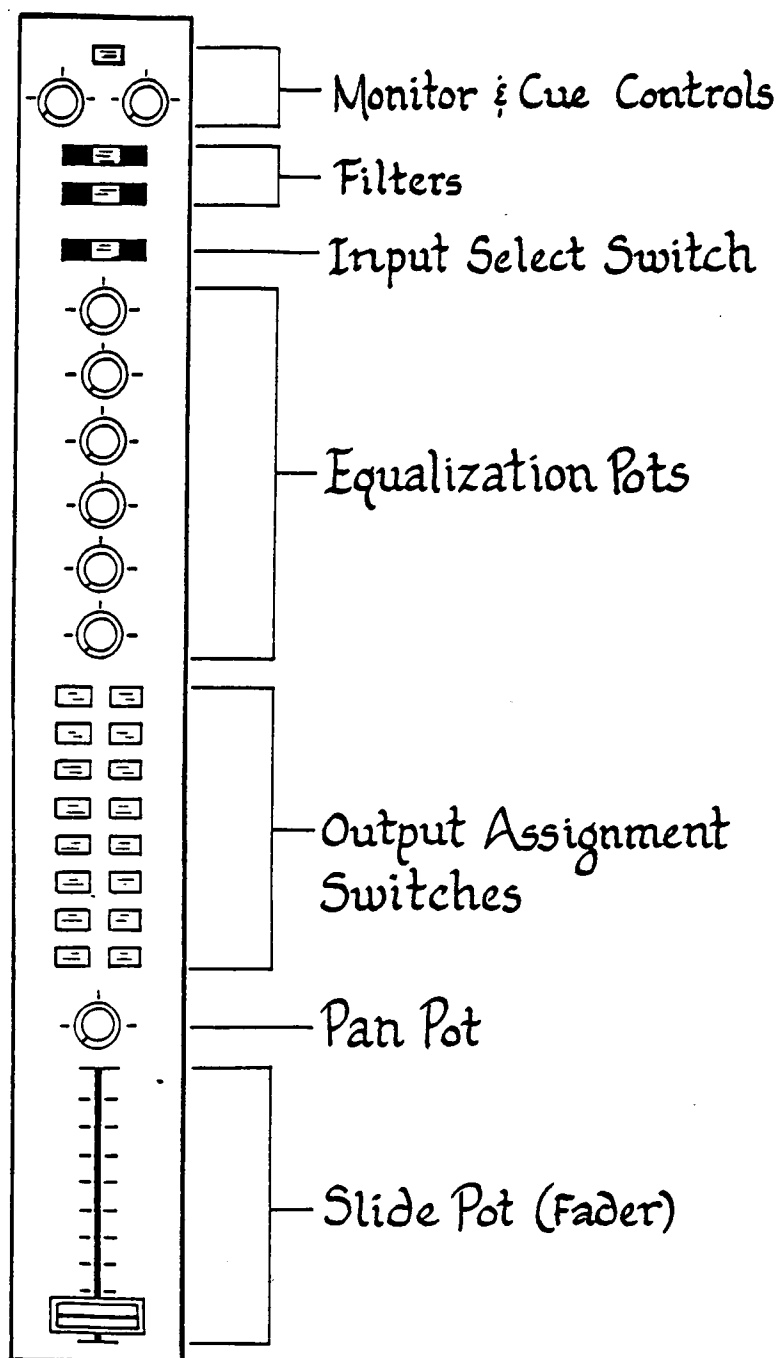
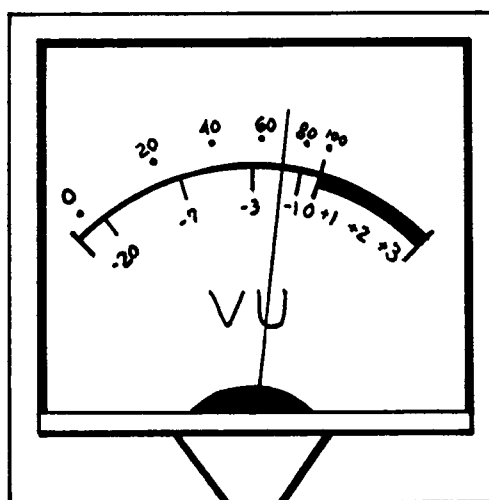
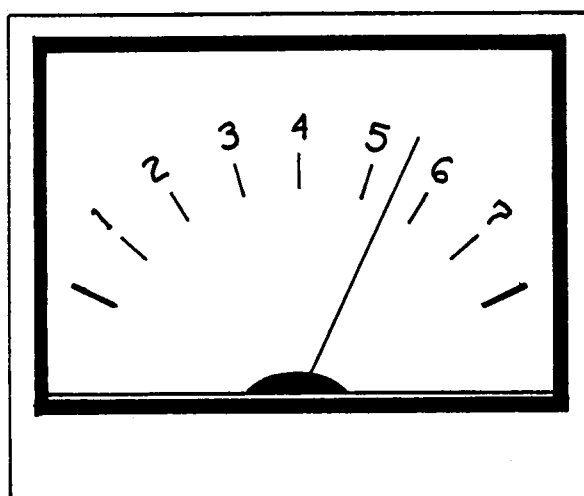
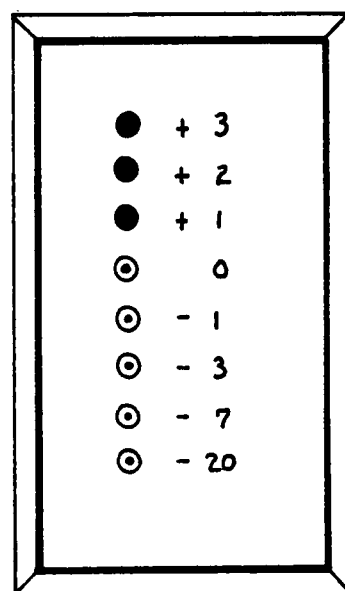


Figure 22. Typical Console Channel Controls.



VU Meter

Peak Program  
Meter

Led Meter

Figure 23. Console Metering.

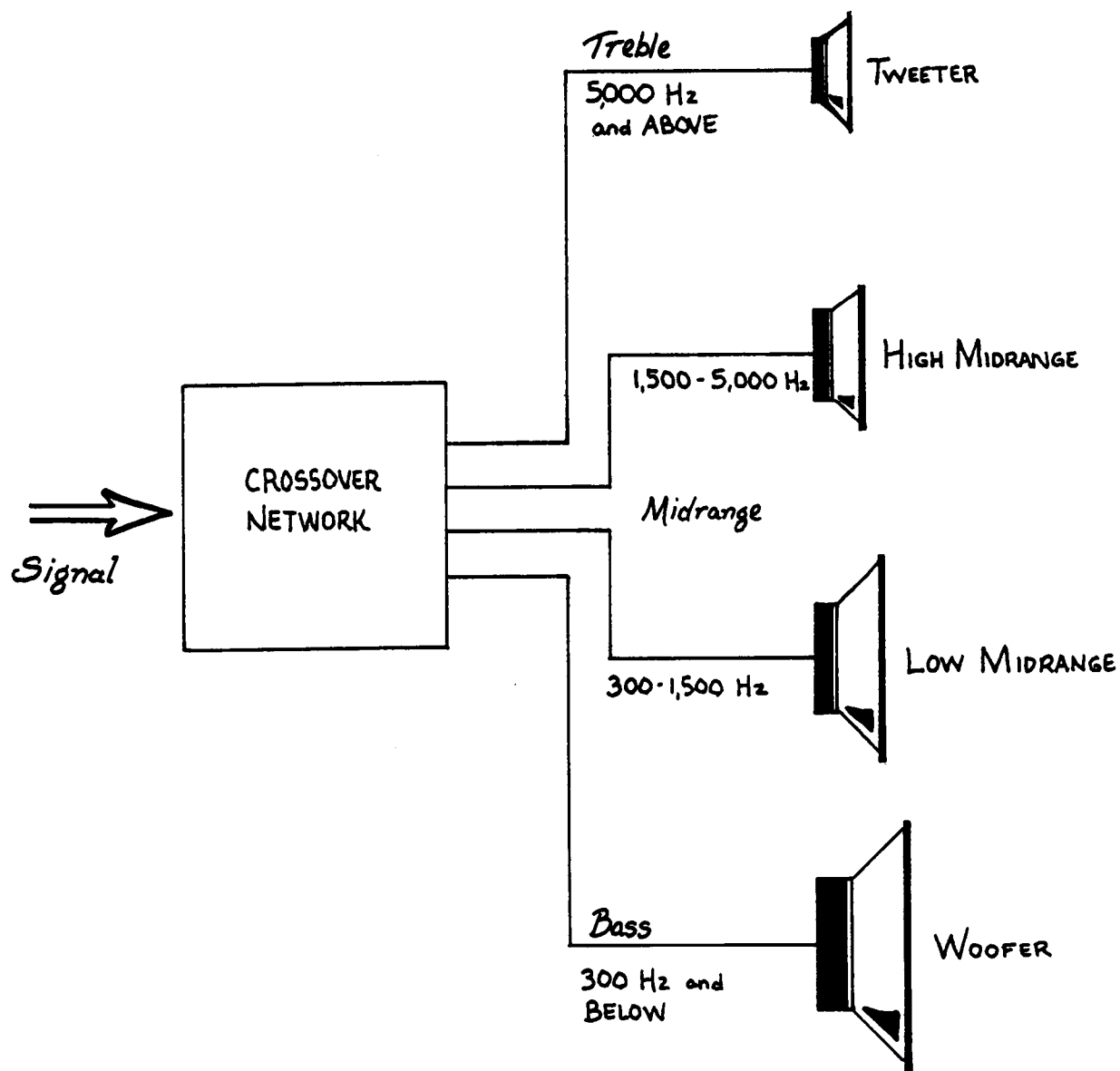


Figure 24. Four-Way Speaker System.

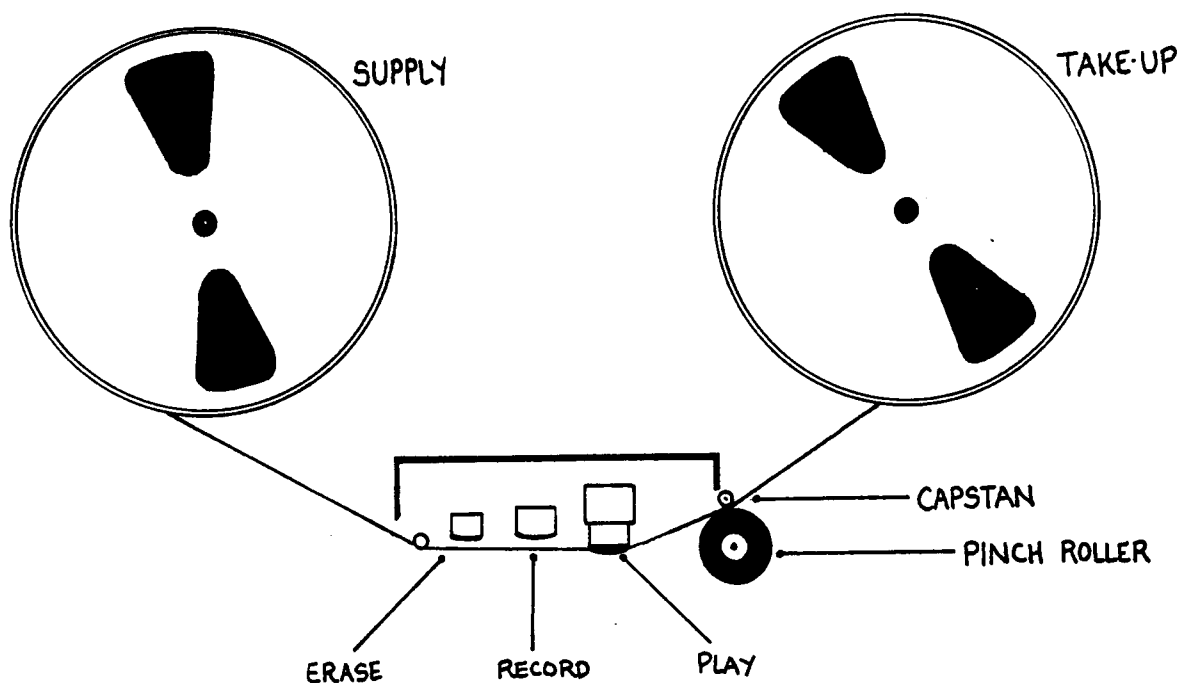


Figure 25. Tape Transport Diagram.

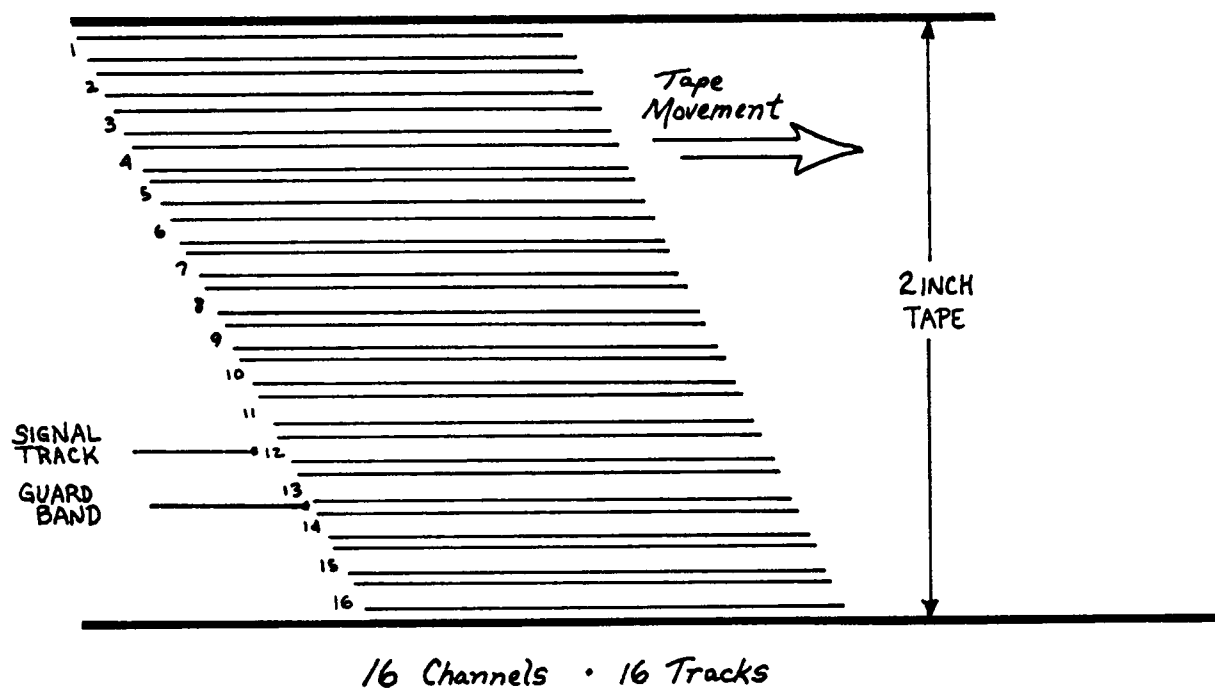


Figure 26. 16-Track Tape Configuration.

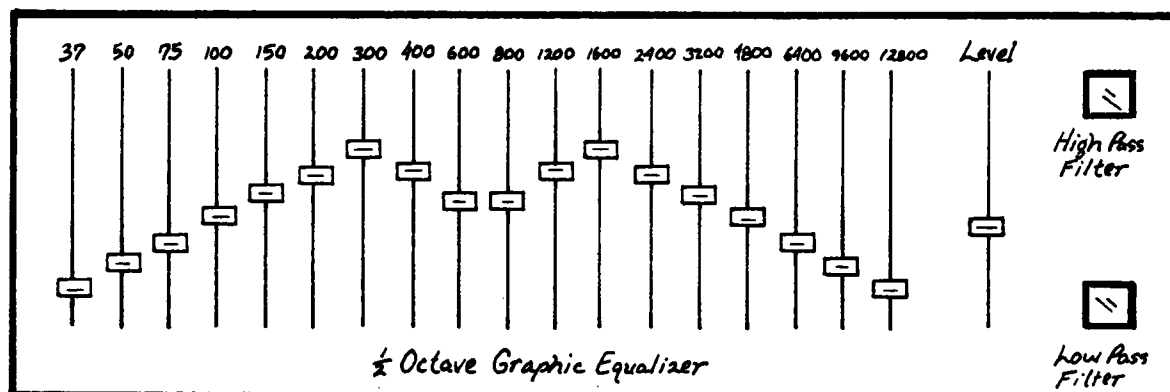


Figure 27. Graphic Equalizer with Low and High Pass Filters.

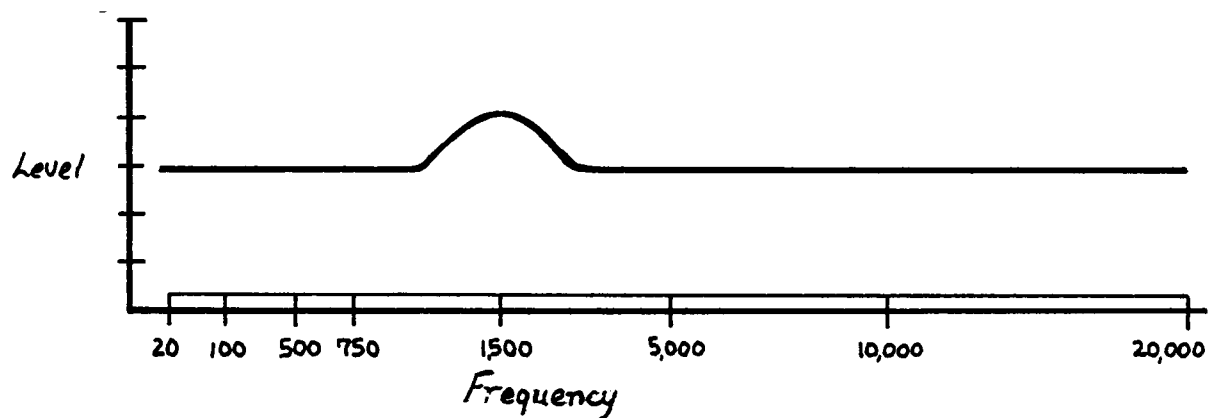


Figure 28. Frequency Response Graph with 1500 Hz Peak.

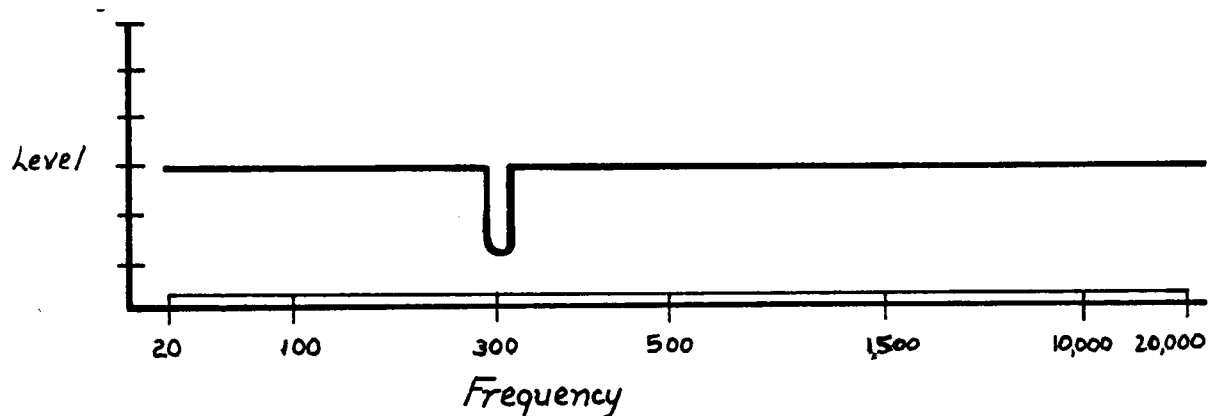


Figure 29. Frequency Response Graph with 300 Hz Notch.

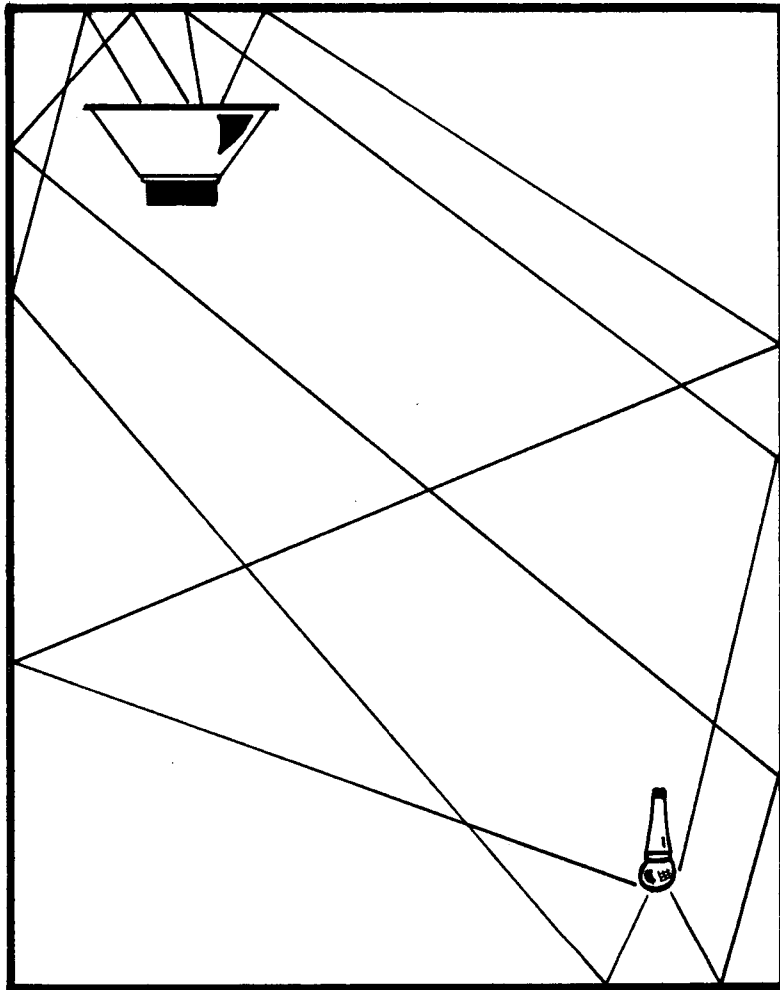


Figure 30. Reflections in an Acoustic Echo Chamber.

## VITA

Richard David Phillips was born near Pittsburgh, Pennsylvania, on January 24, 1956. He attended elementary and secondary schools in Irwin and was graduated from Norwin Senior High School in June 1973. The following September he entered Carnegie-Mellon University in Oakland, Pennsylvania. After studying electrical engineering for two years he transferred to Johnson Bible College in Knoxville, Tennessee. In May 1979 he received a Bachelor of Arts degree in Biblical Studies with specializations in Missions and Communications. In the fall of that year he entered The University of Tennessee, Knoxville, to begin study toward a Master of Science degree in Communications. This degree was awarded in August 1983.

The author has worked in radio and television design and production for over eight years. He is a member of the National Religious Broadcasters, Intercollegiate Religious Broadcasters, and Christian Communications Fellowship. He has served as producer, performer, and engineer for several audio and video productions with nationwide distribution. He is an ordained minister of the Christian Church. Mr. Phillips built the radio, recording, and television studios at Johnson Bible College where he has been employed since 1979 as professor of communications.