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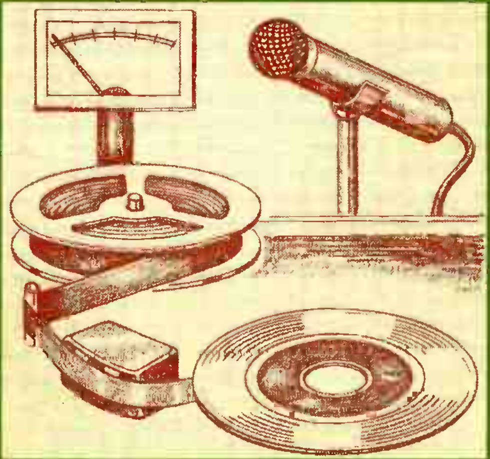
modern RECORDING TECHNIQUES



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Modern Recording Techniques

By

Robert E. Runstein

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Preface

The multitrack recording studio has become the source of almost all pop music recordings released in the United States, thus stirring the interest of many people. While I was Chief Engineer and Technical Director of Intermedia Sound Studios in Boston, the studio was approached by a virtually continuous stream of people who wanted to be trained as recording engineers. Most of them had, as their only qualifications, a love for music listening and perhaps some musical training or knowledge of hi-fi equipment. All of them wanted to learn how the records they enjoyed were made and to participate in making records as engineers, producers, or recording artists.

Although several books have been written about sound and sound studios, they all seem to emphasize either sound reinforcement, film sound, or radio/tv broadcast sound. Pop music recording has been only mentioned in passing, if at all. The purpose of this book is to fill the information gap for novice and experienced engineers, record producers, and recording artists.

The recording engineer's function is to act as an interface between the producer of a session and the studio equipment. He must translate the producer's ideas into microphone placement and electronic adjustments, and create a high-quality recording of the artist's performance. To do this effectively, the engineer must be well versed in the function, operation, and limitation of each control and piece of equipment in the studio, and he must know how to use them creatively to produce the desired results.

This book introduces the reader to the equipment and controls he will encounter in the modern multitrack recording studio in terms of both the operating techniques currently in use and the roles they play in creating the finished product. An understanding of the information contained herein, combined with sufficient session observation time for familiarization with the location of the different pieces of equipment in a particular studio, will enable a potential engineer to progress rapidly from observer to assistant engineer. At this point, he will begin operating the equipment and perhaps handling simple sessions on his own. Further experience at the controls during sessions will lead him to full engineer status.

Producers and recording artists will benefit from familiarity with the techniques and equipment they will be using in the studio. Understanding the concepts of sound and studio capabilities and limitations will enable them to better communicate their ideas to the engineer and create better records. More experienced engineers will find several topics and techniques with which they may not be acquainted. Interlocked tape machines have only recently seen much use in pop music recording, and automated mixdown and quad discs are entirely new fields.

In Chapter 1, the acoustical, mechanical, electrical, and magnetic transformations of a signal from live performance to reproduction from disc are outlined. Chapter 2 further develops the physical concepts and terminology used in recording as they pertain to human perception of sound waves. The conversion of sound energy to electrical energy by microphones is the topic of Chapter 3.

Chapter 4 details magnetic tape recording and the function and operation of the components and controls of magnetic tape recorders. Chapter 5, on signal processing, will enable the reader to understand and duplicate many of the special effects heard on current records. The studio equipment, from mike input to tape and monitor speaker outputs, and the operation of each console control is detailed in Chapter 6. The noise-reduction systems described in Chapter 7 provide a means of overcoming some of the limitations of present magnetic tapes. Monitor speakers, their components, and their interaction with the listening room are the topics of Chapter 8.

Chaper 9 presents the setup, operation techniques, and procedures used in recording, overdubbing, mixing, and sequencing sessions, as a guide to the novice engineer. Chapter 10 covers techniques of interlocking the speeds of several tape machines. Two different automated mixdown systems currently on the market are introducd, and the significance of automation in this area is discussed in Chapter 11.

The theory of disc recording is introduced in Chapter 12, and the cutting lathe and mastering console used to transfer signals from tape to disc are detailed. Quadraphonic sound and its storage on a two-channel disc are the subject of Chapter 13. Three of the encoding systems currently vying for dominance in this field are presented and compared.

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to my love and wife, JoAnne

Contents

CHAPTER 1

The Recording Chain

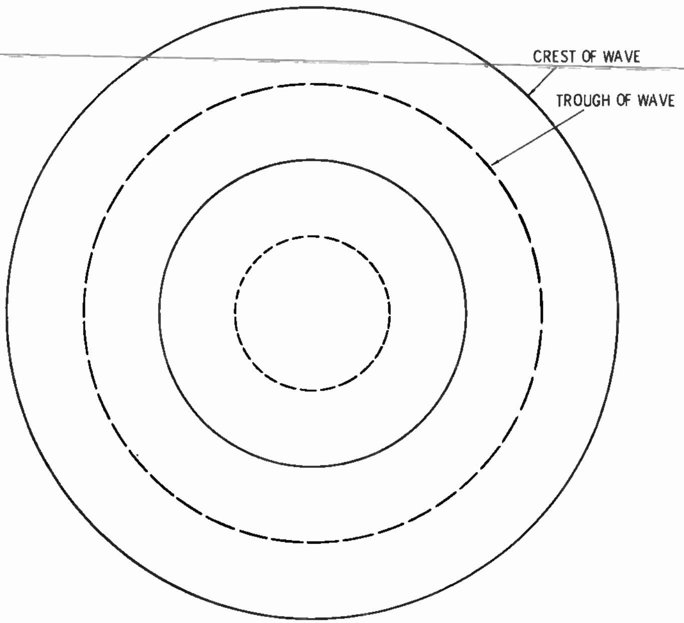
INTRODUCTION

The recording chain begins with the generation of a sound. Sound consists of *waves*, similar to those seen in a pool of water into which a stone is thrown (Fig. 1-1A). In a recording studio, the *medium* through which the waves travel is air, rather than water. The hills and valleys in the water of Fig. 1-1B correspond to levels of air pressure greater and lower than normal atmospheric pressure. These waves are generated by a moving body in contact with the air. This body could be a speaker, someone's vocal cords, a string from a guitar which vibrates the body of the guitar, which in turn vibrates the air next to it, etc. The motion of the generating body produces changes in the air pressure around it in proportion to both the frequency and the amplitude of its vibrations.

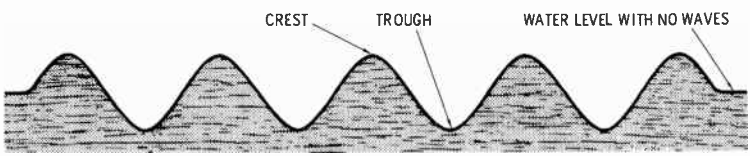
The *frequency* of a wave (Fig. 1-2) is the number of times per second that it passes through all of its values between its maximum positive and negative excursions and returns to its starting value. Each complete fluctuation is called a cycle, and the accepted unit of cycles per second is called hertz (Hz). The *amplitude* of a wave is its height above or below the zero line, which in this case corresponds to the level of normal atmospheric pressure.

The ear detects both the frequency and the amplitude of these variations in pressure and translates them into the sensations of pitch and volume, respectively. What is of interest in the recording studio, is storing these pressure variations in a manner which will enable them to be heard again at a later date.

In order to store the waveforms, the waves are converted into electrical impulses by a *microphone* (mike). The electrical impulses from the mike flow through wires to a series of amplifiers and level controls and then to the magnetic tape recorder where they are con-



(A) Top view.



(B) Side view.

Fig. 1-1. Waves created by a stone falling into a pool of water.

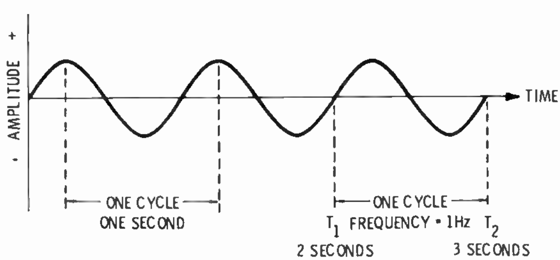


Fig. 1-2. A cycle of a wave can be considered to begin at any point on a waveform.

verted into magnetic impulses, called *flux*, by the *record head* and applied to the magnetic tape. The tape stores the magnetic impulses in the order they are presented to it, so that the electrical signals can be re-created by the tape recorder *playback head* at a later date. These re-created electrical impulses can then be amplified and fed to a *speaker*, which converts the impulses to mechanical motion which in turn re-creates the original variations in air pressure sensed by the microphone. Alternatively, the electrical impulses from the tape

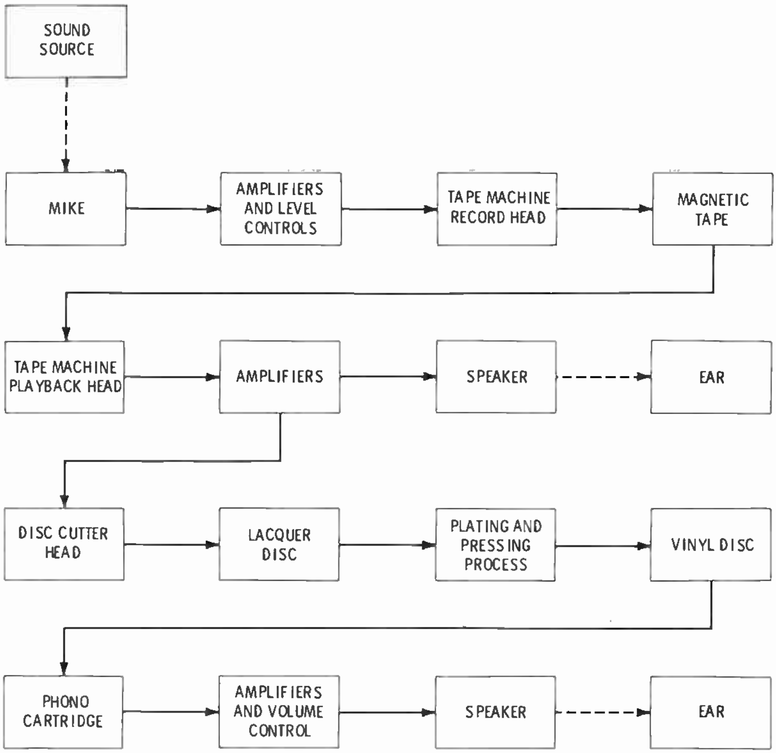


Fig. 1-3. The recording chain.

could be fed to the *cutting head* of a *disc mastering lathe*, where the impulses would be stored in the grooves of a lacquer-covered disc. The disc would be used as a mold for the metal plates which press the vinyl discs available at record stores. The signal is recovered from the vinyl by playing the disc with a *phono cartridge* which converts the impressions in the grooves into electrical impulses which can be amplified and fed to a speaker as before (Fig. 1-3).

Devices used to transfer the signal from one *medium* to another are called *transducers*. The most common transducers used in audio are the microphone, the record head, the playback head, the disc-cutting head, the phono cartridge, and the speaker. Table 1-1 shows the mediums between which each of the above transducers transfers the signal waveform.

The path just described is the most basic one used in recording. Every sound released on current discs has passed through all of the above-mentioned components, unless the sources of the signal were all electronic, in which case the microphone might be bypassed. In the past, recordings were made directly onto discs rather than magnetic tape, but the advantages of magnetic tape recording have made this process obsolete. The recording chain merely reproduces the signal picked up by the mike. By adding additional pieces of equipment at places inside the chain, the signal can be modified creatively. The proper use of these signal-processing devices is a large part of both the engineer's and the producer's roles.

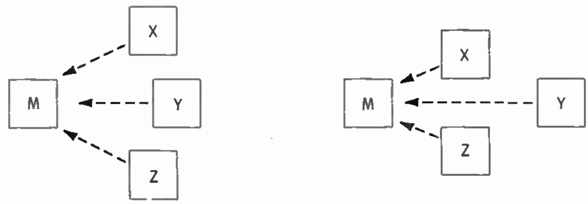
Table 1-1. The Mediums Which Transducers Use in the Studio to Transfer Energy

Transducer	From	To
Microphone	Air molecules	Electrons
Record head	Electrons	Magnetic flux
Playback head	Magnetic flux	Electrons
Disc cutter head	Electrons	Groove modulation on disc
Phono cartridge	Groove modulation on disc	Electrons
Speaker	Electrons	Air Molecules

In an actual recording session, there may be more than one sound source, and the producer may want to hear several different balances between them. One way of doing this would be to move the instruments to different positions with respect to the microphone to obtain the desired balance (Fig. 1-4). Fig. 1-4A depicts equally loud instruments being picked up at the same levels because they are the same distance from the mike, while in Fig. 1-4B, X and Z are picked up louder than Y because Y is farther from the mike.

A less time-consuming and more flexible way is to use several microphones and a *mixing console* to combine the outputs of the different sources in the desired proportion, forming a composite signal to be recorded on tape (Fig. 1-5). The setting of the console level controls determines the loudness of each instrument in the composite signal created by the console.

Another method, used extensively in multitrack recording, is to store the output of each microphone separately on an individual channel of multitrack tape. These separate signals can later be combined in any desired proportion (Fig. 1-6).



(A) Instruments same distance. (B) Instrument Y farther from mike.

Fig. 1-4. Obtaining different instrumental balances by rearranging the relative positions of the microphone (M) and the instruments (X,Y,Z).

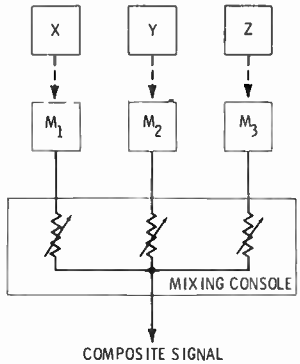


Fig. 1-5. Controlling instrumental balance by using separate mikes and a mixing console.

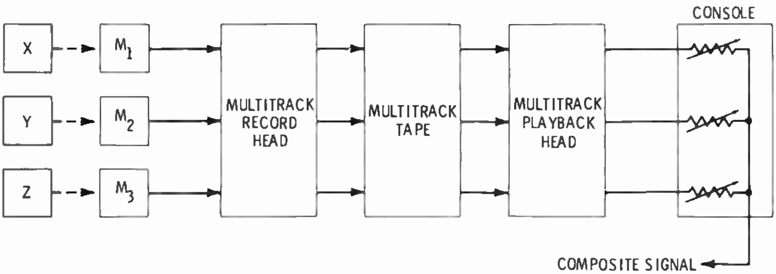


Fig. 1-6. Instrumental balance by recording instruments on separate tracks of multitrack tape.

ACOUSTICS AND RECORDING

The environment in which a sound is generated has a large effect on the way the ear interprets sound. As a result, there are two major methods of recording. The first considers the environment to be a part of the sound of the instrument, while the second tries to eliminate the effects of the environment as much as possible.

When sound is generated, it leaves the source in an arc (Fig. 1-7), the angle of which is determined by the nature of the source. Some of the sound reaches the listener directly without encountering any

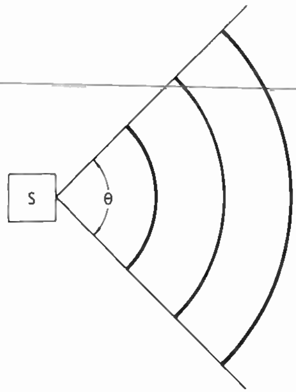


Fig. 1-7. Waves leave the sound source (S) in an arc with angle θ.

obstacles (Fig. 1-8), and some of it reaches the surrounding surfaces first. If these surfaces are hard, they will reflect the sound waves and some of these reflections will also reach the listener. If these surfaces are soft and absorb the waves or allow the waves to pass through them, little energy will be reflected back to the listener.

Sound travels at the constant speed of 1130 feet per second in air at 70 °F. The wave which travels a straight line follows the shortest path and arrives at the listener's ear first. This is called the *direct sound.* Those waves which bounce off the surrounding surfaces must travel farther to reach the listener, and therefore they arrive after the direct sound. These waves form what is called the *reflected sound* which, in addition to being delayed, can also arrive from different directions than the direct sound. As a result of these longer path lengths, the ear hears the sound even after the generator stops emitting it.

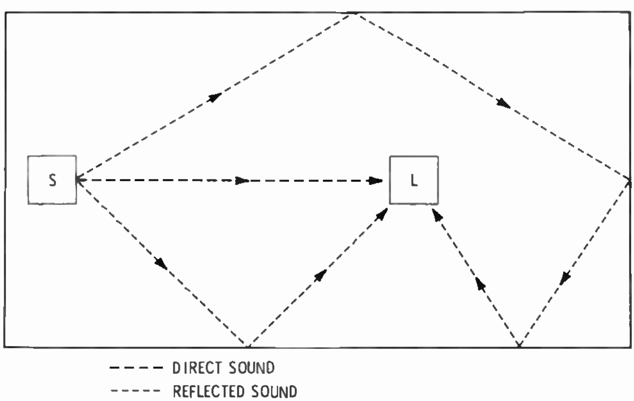


Fig. 1-8. In an enclosed space, some sound reflects off the surrounding surfaces before reaching the listener (L).

Highly reflective surfaces absorb less wave energy at each reflection and enable the sound to persist longer after the generator stops than highly absorptive surfaces which would dissipate the wave. The persistence of the sound after the generator has stopped emitting it is called *reverberation*. Reverberation is actually composed of a series of many echoes of the original sound which arrive so closely together in time that the ear cannot separate them.

The time it takes for the sound to decrease to one-millionth of its original *sound-pressure level* is called its *decay* or *reverberation time*. The absorption characteristics of a room determine its reverberation time. The ear recognizes the reverb time and uses this information to judge the hardness of the surrounding surfaces, while the time elapsed between hearing the original sound and the onset of the first reflection provides the ear with information as to the size of the room. The closer the listener moves to the source, the louder the direct signal becomes because its path becomes shorter and the softer the reflected sound becomes because its path becomes longer

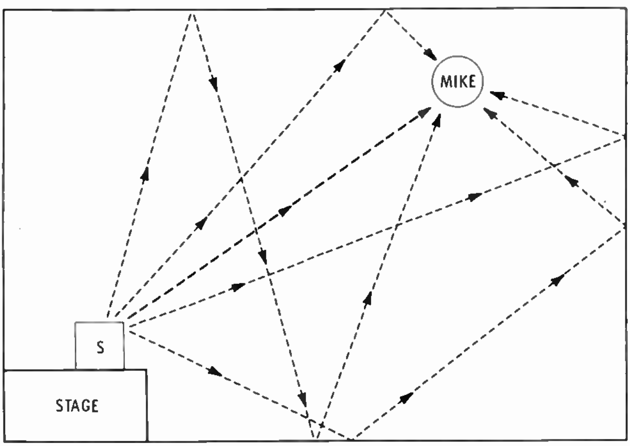


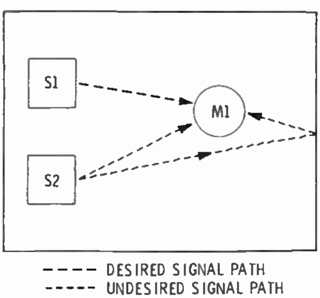
Fig. 1-9. Distant microphones pick up both direct and reflected sound.

(loudness varies in inverse proportion to the square of the distance from the sound source). The ratio of direct to reflected sound enables the listener to determine his distance from the sound source. Thus, the longer the reverb time, the harder the room surfaces; the longer the time before the first reflection, the larger the room; and the greater the amount of reverberated sound in proportion to the direct sound, the farther the listener is from the source.

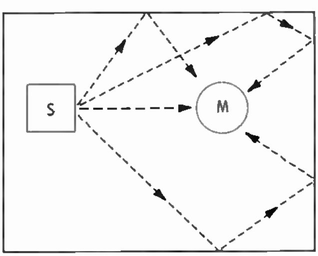
Classical recordings rely heavily on the reverberation or *ambience* of the halls they are recorded in. This is because the instrumental scoring of the music is determined by the way the instruments sound to an audience in a performance hall. The hall is usually large with hard surfaces, and the audience sits at a distance from the performers, so the time before the first reflection, the reverb time, and the ratio of reverberated to direct sound are large. To capture these characteristics, classical recordings usually use only two microphones suspended in the air some distance in front of the orchestra, with close-up mikes used for soloists or special instrumental groups only when necessary for proper balance. Because they are at a distance from the sound source, the main mikes pick up both the direct sound and some of the reflected sound (Fig. 1-9). Quadraphonic classical recordings usually use two more microphones positioned near the back of the hall to pick up mainly reverberant sound to be used in re-creating the concert hall's spatial characteristics in the quadraphonic listener's home.

In popular music, on the other hand, it is often desirable to control the position and volume of each instrument independently of the others in the stereo or quadraphonic end product. In addition, it is often desirable to postpone this positioning and balancing until sometime after the recording session to eliminate paying studio musicians waiting for these decisions to be made. To control the balance of the instruments independently, a multitrack tape recorder is used, and each instrument or group of instruments for which separate control is desired is assigned to a separate channel of the tape.

Because most of the instruments are usually played at the same time and in the same room to permit musical interchange between the performers, two problems arise. First, the direct or reverberated sound of one instrument may be picked up by the microphone intended for another instrument (Fig. 1-10A); this can cause both interaction of volumes and problems of frequency cancellation. Sec-



(A) The direct and reflected sound of S_2 is picked up by the mike intended for S_1 only.



(B) Microphone M picks up the reflected sound as well as the direct sound from S.

Fig. 1-10. Ways reflected sound can be picked up.

ond, the reflected sound from an instrument may be picked up by its own mike, thereby limiting the spatial placement of the instrument in the final mix (Fig. 1-10B).

To overcome these possible limitations, many studios reduce the reverberation time of their recording rooms through the use of highly absorptive materials in their construction. This assures the presence of almost entirely direct sound for the mikes to pick up. Spatial localization, with respect to listener-source distance and size of the performance hall, is provided at a later time through the use of artificial reverberation devices.

To prevent the *leakage* of the direct sound of one instrument into the mike assigned to another, *uni-* or *bidirectional* mikes which discriminate against sound coming from certain directions are frequently used and placed as close as possible to the sound sources. This technique is called *close miking*. Sound-absorbing *baffles* are also used between the instruments to further increase the acoustic *separation* as shown in (Fig. 1-11). The baffles absorb the sound

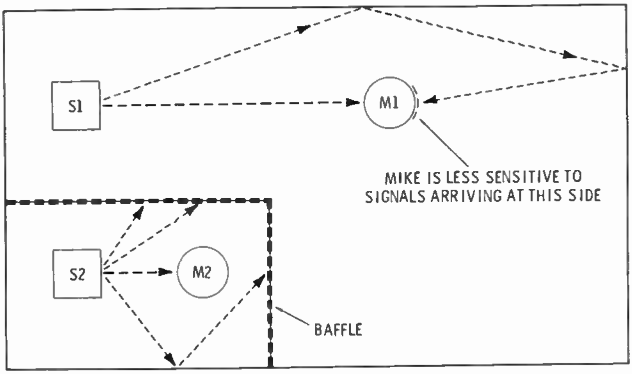


Fig. 1-11. Close miking using sound-absorbing baffles and unidirectional mikes.

waves and prevent them from passing through or being reflected. Unidirectional mike M1 has its "live" side pointing at S1 so it is less sensitive to the reflective sound arriving at its rear. However, not all pop music studios are completely "dead." Many are designed with longer reverberation times on one side of the room so that certain instruments, such as strings and horns, can be recorded with more of a concert hall sound than would be obtainable through the use of artificial reverb alone.

The multitrack method of recording is often referred to as *multiple-mono* recording because each of the channels may be acoustically separate from the others or even recorded on a different day. The performance resulting from the multitrack tape is created by assigning each track to a location on an imaginary stage between the speakers through the use of electrical front-back-left-right positioning devices (Fig. 1-12) called *pan pots* and creating an imaginary room around the stage by adding artificial reverb to simulate reflections from the surrounding surfaces. For those who think that a recording should only preserve a live performance, this is annoying.

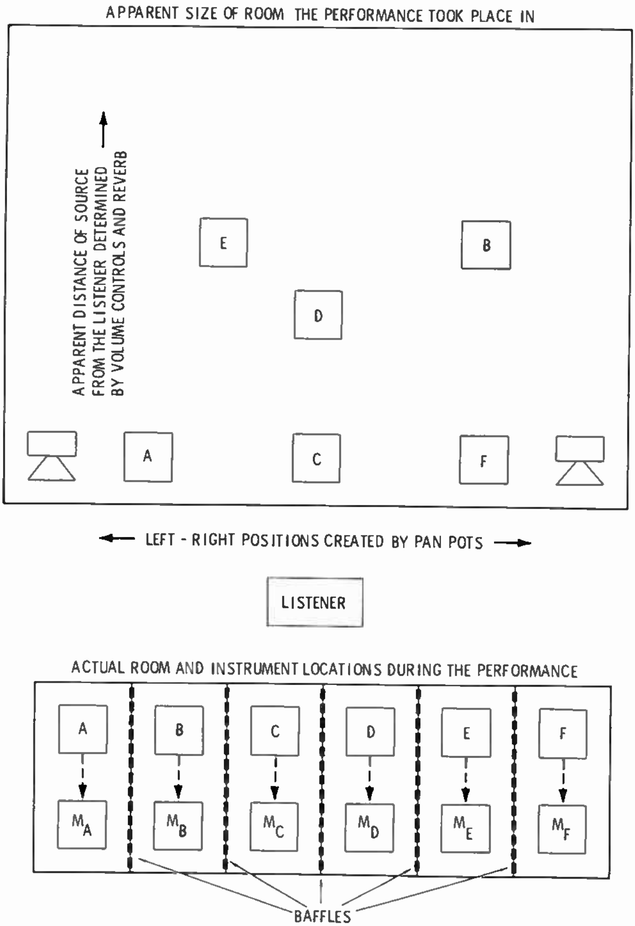


Fig. 1-12. Close-miked instruments A through F can be made to appear as if they originate from various locations in an imaginary stage behind and between the stereo speakers.

Others, however, think of this type of recording as an extension of the performance, or even as a new art form, for it enables musicians to express their music beyond what they are technically or financially able to do live. Each method has its applications. A violin does not make the same smooth sound when close miked as it does when it is picked up as part of an orchestra, while a bass guitar loses its punch in a reverberant hall. Whether the recording is to expand the artist's capabilities or whether it is to document them is a decision which is made by the producer of the particular session.

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CHAPTER 2

Sound and Hearing

WAVEFORM CHARACTERISTICS

Every wave has characteristics which distinguish it from other waves, these are: (1) frequency, (2) amplitude, (3) velocity, (4) wavelength, (5) shape, (6) phase, (7) harmonic content, and (8) envelope.

Frequency

In the diagram (Fig. 2-1), the value of the waveform starts at zero at time t = 0, increases to a maximum value in the positive direction, decreases through zero to a maximum in the negative direction, then returns to zero and begins the process again. One completion of this path is called one cycle of the wave. A cycle can begin at any point on the waveform, but to be complete, it must pass through the zero line and end at a point moving in the same direction (positive or negative) having the same value as the starting point. Thus, the wave from t = 0 to t = 2 constitutes a cycle, and the wave from t = 1 to t = 3 is also a cycle. The number of cycles which occur end to end (that is cycle two begins at the cycle one end point) in one

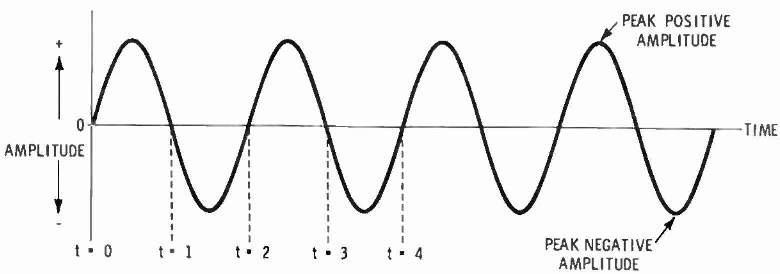


Fig. 2-1. Graph of waveform amplitude vs time.

second is called the *frequency* of the waveform and is measured in the unit *hertz* (Hz). The term hertz is equivalent to cycles per second.

Amplitude

In the case of sound waves, the positive and negative excursions of the diagram represent increases and decreases in the atmospheric pressure of the air caused by the sound source and perceived by the listener's ear, with the zero line representing normal atmospheric pressure. The distance above or below the zero line is called the *amplitude* of the waveform at that particular instant of time, and the maximum positive and negative excursions are called the positive and negative peak amplitudes, respectively.

Velocity

The *velocity* of a wave is the speed with which it travels through a *medium* (Fig. 2-2) and is given by the equation:

$$V = \frac{d}{t_2 - t_1} \qquad \text{(Eq. 2-1)}$$

where,

V is the wave velocity of propagation in the medium,
d is the distance from the source,
t is the time in seconds.

In the case of sound waves, the medium is air molecules; for electricity, the medium is electrons. The wave velocity determines how

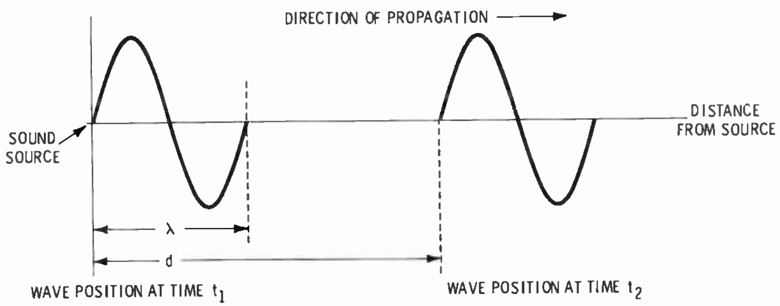


Fig. 2-2. Demonstrating velocity of a wave.

fast a particular cycle of a waveform will travel a certain distance. At 70 °F the speed of sound waves in air is approximately 1130 feet per second. This speed is temperature dependent and increases at a rate of about 1.1 feet per second for each degree Fahrenheit increase of temperature.

Wavelength

The *wavelength* (λ) of a wave is the actual distance in the medium between the beginning and end of a cycle and is equal to:

$$\lambda = \frac{V}{f} \qquad \text{(Eq. 2-2)}$$

where,

λ is the wavelength in the medium,
V is the velocity in the medium,
f is the frequency in Hz.

Thus, the wavelength of one cycle of a 30-Hz sound wave is 1130 ft/sec divided by 30 Hz, or 37.66 feet. Although we speak of wave velocity, the particles of the wave medium do not move far. Sound travels as a compressional wave. The air in one spot is compressed by the sound source and it compresses the air next to it as it expands back to normal. The location of the compression moves at the velocity of sound, but the air molecules only move to the extent that they are pushed together and are then returned to their normal spacing. The molecules do not travel with the wave. The action of electrical waves in a wire is the same, except that electrons rather than air molecules are compressed.

Shape

The diagrams up to this point have only shown one *shape* of waveform, the sine wave, but the eight characteristics apply to simple waves of all shapes. Other simple waveforms such as square waves, triangle waves, and sawtooth waves are shown in Fig. 2-3. These are called simple waveforms because they are continuous and repetitive. One cycle of a square wave looks exactly like the next, and they are all symmetrical about the zero line. Complex waves are waves which do not necessarily repeat and which are not necessarily symmetrical about the zero line. An example of a complex waveform would be that created by the speaking of a word (Fig. 2-4). Since complex waveforms often do not repeat, it is difficult to divide them into cycles or to categorize them as to frequency. Fortunately, all possible simple and complex waveforms can be constructed through the use of combinations of sine waves of different frequency, phase, and amplitude. The discussion focuses on sine waves.

Phase

Since a cycle can begin at any point on a waveform, it is possible to have two wave generators producing waves of the same shape, frequency, and peak amplitude, which will have different amplitudes at any one point in time. These waves are said to be out of *phase*

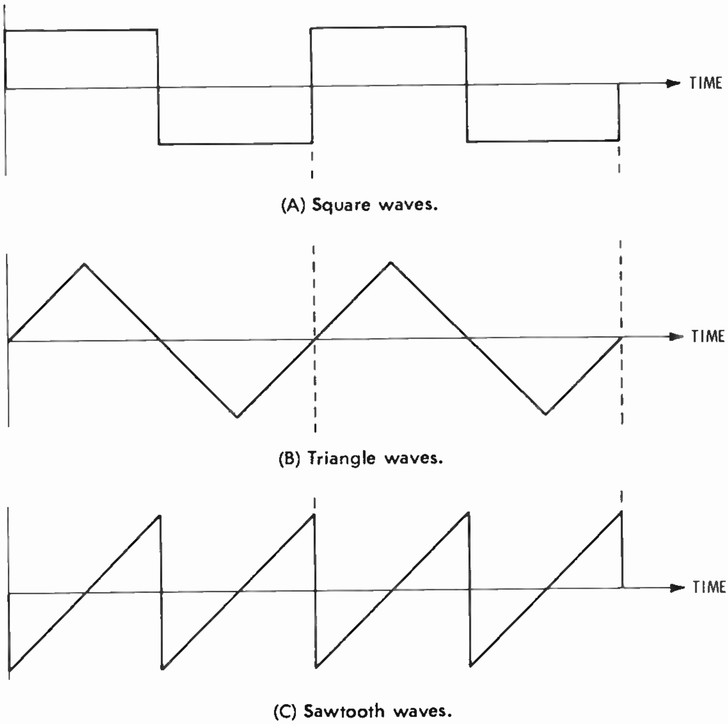


Fig. 2-3. Simple waveforms.

with respect to each other. A cycle can be divided into 360°, and the sine wave (so named because its amplitude follows the trigonometric sine function) is usually considered to begin at 0° with 0 amplitude, increase to a positive maximum at 90°, decrease to 0 at 180°, decrease to a negative maximum at 270°, and return to zero at 360°. The first wave (A) in Fig. 2-5 can be considered in phase with the "ideal" sine curve because their amplitudes match at each

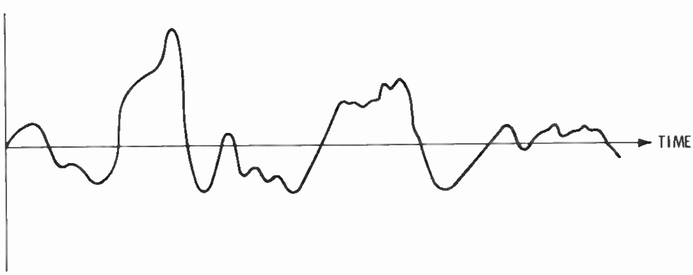


Fig. 2-4. A complex waveform.

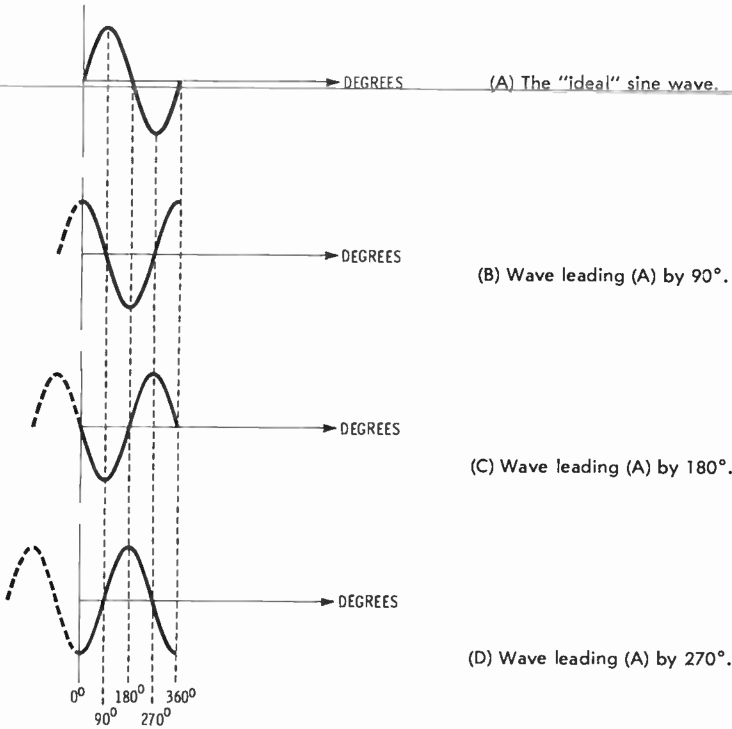
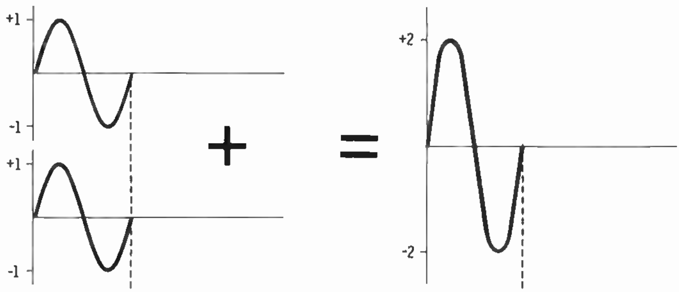


Fig. 2-5. Demonstrating phase relationships of sine waves.

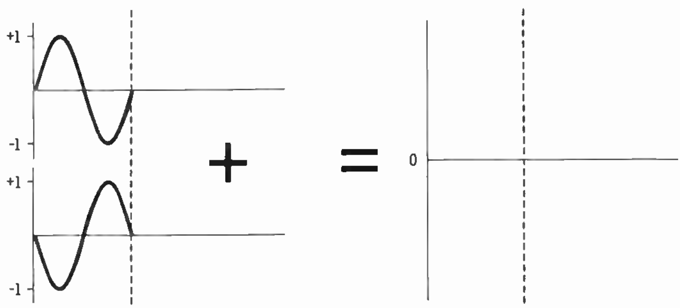
point on the X axis. The second waveform (B) reaches its maximum positive amplitude 90° before the first and is out of phase with the first because it leads it by 90°. The third wave (C) begins decreasing from zero 180° before the first, and is 180° out of phase. The fourth (D) leads the first by 270° and is also out of phase.

Note that it is assumed that the waves began at the zero degree time; if they had begun earlier we could have said that wave one lagged behind wave two by 90°. A wave which is a multiple of 360° out of phase with another can be considered to be in phase with it if the waves are continuous; there would be no way to distinguish that one of the waves actually lagged behind the other. However, if the waveforms were observed to begin, end, or change in amplitude or frequency, the lagging waveform would do so late and could thus be distinguished.

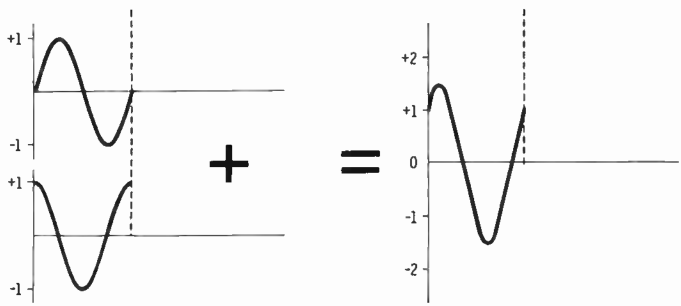
Waveforms can be added by adding their signed amplitudes at each instant of time. When two waveforms which are completely in phase (0° phase difference) and of the same frequency, shape, and peak amplitude are added, the resulting waveform is of the same frequency, phase, and shape but will have twice the original peak amplitude (Fig. 2-6A). If two waves are the same as the ones just described, except that they are completely out of phase (phase difference of 180°), they will completely cancel each other when added, resulting in a straight line of zero amplitude (Fig. 2-6B). If the second wave was only partially out of phase (not exactly 180° or $(2n - 1) \times 180°$ out of phase), it would *interfere constructively*



(A) The amplitudes of in-phase waves add when they are mixed.



(B) Waves of equal amplitude cancel completely when mixed 180° out of phase.



(C) The amplitudes of partially out-of-phase waves add in some places and subtract in others when mixed.

Fig. 2-6. Adding sine waves.

in some places, resulting in a more positive amplitude in the combined wave than in the first wave at that point in time and *interfere destructively* at other points, resulting in a more negative amplitude at those points in time than the first wave (Fig. 2-6C).

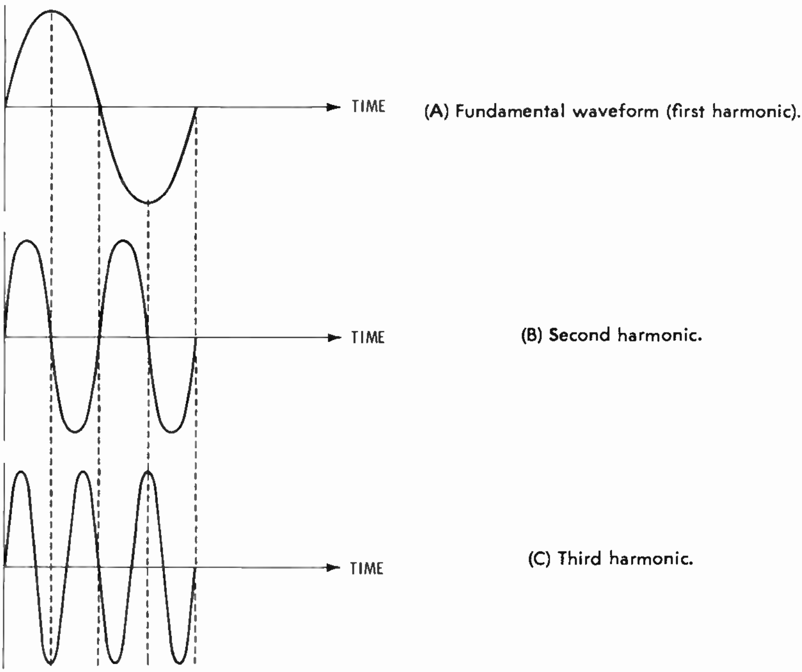


Fig. 2-7. Illustrating harmonics.

Harmonics

A waveform having a frequency that is an integral multiple of the frequency of a second waveform is called a *harmonic* of that second waveform. For example, a 1000-Hz wave is a harmonic of a 500-Hz wave because it is two times the 500-Hz frequency. In this case, the 500-Hz wave is called the *fundamental* or first harmonic (Fig. 2-7A), and the 1000-Hz wave is called the second harmonic (Fig. 2-7B) because it is the result of multiplying the fundamental frequency by two. The third harmonic would be 1500 Hz (Fig. 2-7C).

As mentioned earlier, sine waves can be used to create any other waveform. A square wave can be constructed by adding together a fundamental wave and all of its odd-numbered harmonics in the proper proportions (Fig. 2-8). The harmonics subtract from the fundamental where they are out of phase with it and add to the fundamental where they are in phase, eventually making the combination a square wave. Similarly, a sawtooth wave can be formed

(A) A square wave with frequency f.

(B) A sine wave with frequency f.

(C) Sum of a sine wave with frequency f and a lower amplitude sine wave of frequency 3f.

(D) Sum of sine wave of frequency f and lower amplitude sine waves of 3f and 5f begin to resemble a square wave.

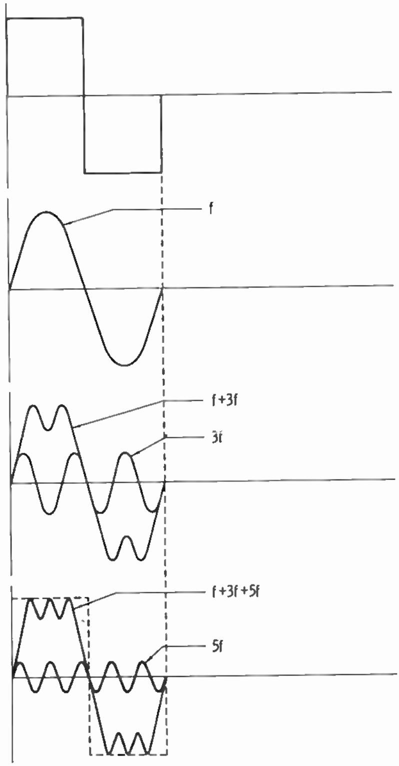


Fig. 2-8. Obtaining a square wave by adding odd harmonics.

by adding both the odd and even harmonics of the fundamental wave in the proper proportions.

Envelopes

The *envelope* of a waveform describes the way its intensity varies. For a sound wave it describes changes in loudness. The envelope is composed of three sections: attack, internal dynamics, and decay. *Attack* is the manner in which the sound begins and increases in volume. *Internal dynamics* describes volume increases, decreases, and sustentions. *Decay* is the manner in which the sound stops. Each of these sections has three variables: time duration, amplitude, and amplitude variation with time. Fig. 2-9 illustrates the envelope of a clarinet note. The attack (A) and decay times (D) are long and the internal dynamics (C) consist of sustain. Fig. 2-10 illustrates the envelope of a snare-drum beat. Note that here the initial attack (A) has much greater amplitude than the internal dynamics (C). In addition, the attack, initial decay (B), and final decay (D) are fast. A cymbal crash (Fig. 2-11) has a similar high-amplitude, fast

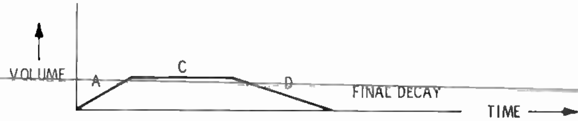


Fig. 2-9. Envelope of a clarinet note. A is the initial attack, C is the internal dynamics, and D is the final decay.

attack (A) with fast initial decay (B), but it sustains longer (C) and decays slower (D). An organ tone (Fig. 2-12) has very rapid attack (A) and decay times (D) and a constant internal amplitude (C) unless the volume pedal is used to vary the envelope (Fig. 2-13).

Fig. 2-10. Envelope of a snare-drum beat. B is the initial decay.

Envelopes with short attack times followed by fast initial decays are characterized as sounding percussive, while slow attacks and decays have gentler, smoother sounds.

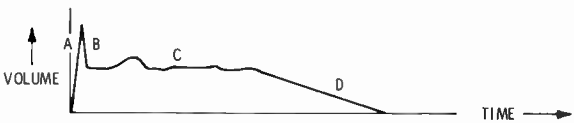


Fig. 2-11. Envelope of a cymbal crash.

THE EAR

A sound source produces sound waves by alternately compressing and rarefying the air between it and the listener. The compression causes an increase of pressure above the normal atmospheric pressure on the ear corresponding to the positive section of the waveform in the illustrations. The rarefaction causes a decrease of pressure below the normal atmospheric pressure on the ear correspond-

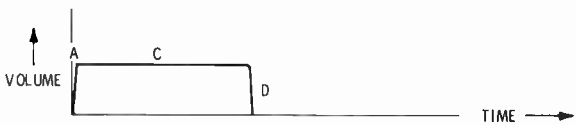


Fig. 2-12. Envelope of an organ tone with the volume pedal stationary.

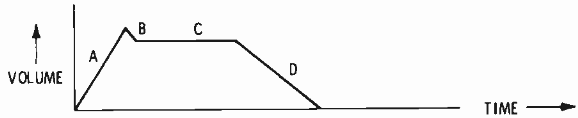


Fig. 2-13. Envelope of an organ tone with the volume pedal varied.

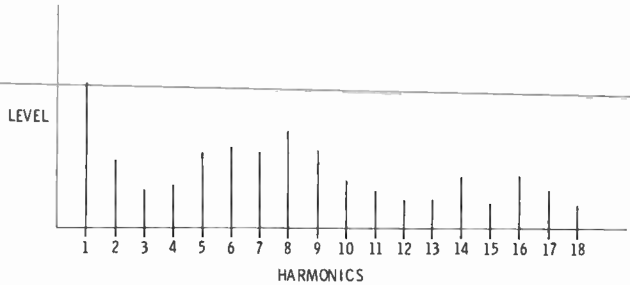
ing to the negative section of the waveforms. The ear responds to these variations in pressure by producing the sensation of hearing. The frequency of the wave reaching the ear determines the pitch the listener hears.

The ear perceives frequencies which are even multiples of each other to be specially related, and this relationship is the basis of the musical *octave*. For example, since concert A is 440 Hz, the ear hears 880 Hz as having a special relationship to concert A; namely, that it is the tone higher than concert A that sounds most like concert A. The next note above 880 Hz that sounds most like 440 Hz would be 1760 Hz. Therefore, 880 Hz is said to be one octave above 440 Hz and 1760 Hz is said to be two octaves above 440 Hz. The human ear does not respond to all frequencies of pressure waves. It responds to a frequency range of about 15 Hz to 20 kHz, a range of ten and one-half octaves. Some young people may hear as high as 23,000 Hz, but the high-frequency response of the ear drops off with age and many people over 60 years of age cannot hear above 8 kHz.

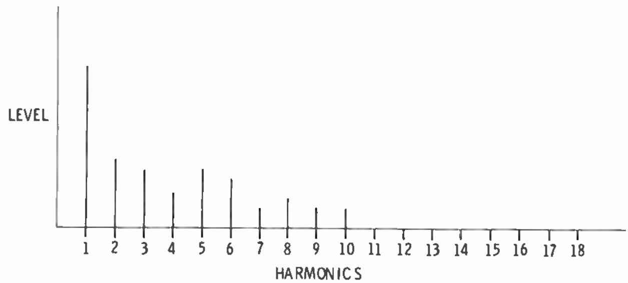
The ear operates over an energy range of more than 10^{12}:1 (1,000,000,000,000:1) and compresses the perceived intensity level in order to protect itself. The loudness of a sound is perceived by the ear as varying approximately in proportion to the logarithm of its energy. As a result, increasing the power output of an amplifier by 10 watts, from 10 to 20 watts, gives a significantly greater volume *increase* than increasing power output from 60 to 70 watts. The ear is sensitive to the ratio of the two power levels, and 20 watts is two times ten or a ratio of 2:1. To get the same increase in loudness, the 60-watt power output would also have to be doubled to 120 watts.

The ear has its greatest sensitivity in the range of 1 to 4 kHz. This means that a 1-kHz sine wave which produces a given sound pressure will sound louder than a 10-kHz sine wave which produces the same sound pressure. In addition, the nature of the ear causes it to produce *harmonic distortion* of sound waves above a certain volume level. Harmonic distortion is the production of harmonics of a waveform which do not exist in the original signal. Thus, the ear can cause a loud 1-kHz wave to be heard as a combination of 1-kHz, 2-kHz, 3-kHz, etc., tones.

Harmonics are very important with respect to musical instruments because their presence and relative intensities in the sound



(A) Played on a violin.



(B) Played on a viola.

Fig. 2-14. Harmonic structure of concert A.

waves produced enable the ear to differentiate between instruments playing the same fundamental tone. For example, a violin has a set of harmonics differing in degree (Fig. 2-14) and intensity from that of a viola. This overtone structure is called the *timbre* of an instrument.

According to Hamm (1973):

"The primary color characteristic of an instrument is determined by the strength of the first few harmonics. Each of the lower harmonics produces its own characteristic effect when it is dominant or it can modify the effect of another dominant harmonic if it is prominent. In the simplest classification, the lower harmonics are divided into two tonal groups. The odd harmonics (third and fifth) produce a 'stopped' or 'covered' sound. The even harmonics (second, fourth, and sixth) produce 'choral' or 'singing' sounds. . . . Musically, the second is an octave above the fundamental and is almost inaudible; yet it adds body to the sound, making it fuller. The third is termed a quint or musical twelfth. It produces a sound many musicians refer to as 'blanketed.' Instead of making the tone fuller, a strong third actually makes the tone softer. Adding a fifth to a strong third gives the sound a metallic quality that gets annoying in character as its amplitude increases. A strong second with a strong third tends to open the 'covered' effect. Adding the fourth and fifth to this changes the sound to an 'open horn' like character. . . . The higher harmonics, above the seventh, give the tone 'edge' or 'bite.' Provided the edge is balanced to the basic musical tone, it tends to reinforce the fundamental, giving the sound a sharp attack quality. Many of the edge harmonics are musically unrelated pitches such as the seventh, ninth, and eleventh. Therefore, too much edge can produce a raspy dissonant quality. Since the ear seems very sensitive to the edge harmonics, controlling their amplitude is of paramount importance. [The study of a trumpet tone] shows that the edge effect is directly related to the loudness of the tone. Playing the same trumpet note loud or soft makes little difference in the amplitude of the fundamental and the lower harmonics. However, harmonics above the sixth increase and decrease in amplitude in almost direct proportion to the loudness. This edge balance is a critically important loudness signal for the human ear." [1][1]

Although the ear may receive the overtone structure of a violin as in Fig. 2-14A, if the listening level is loud enough, the ear will produce additional harmonics and change the perceived timbre of the instrument. This means that sound monitored at very loud levels may sound quite different when played back at low levels. To make things even more difficult, the frequency response of the ear changes with the loudness of perceived signals. The loudness compensation switch found on many hi-fi preamps is an attempt to compensate for the decrease in the ear's sensitivity to low-frequency sounds at low levels.

SOUND-PRESSURE LEVEL

The curves in Fig. 2-15 are the Fletcher-Munson equal-loudness contours, and they indicate average ear response to different frequencies at different levels. The horizontal curves indicate the *sound-pressure levels* (*spls*) at different frequencies which are required to produce the same perceived loudness. Thus to equal the loudness of a 1.5-kHz tone at a level of 110-dB spl, which is the level typically created by a trumpet-type car horn at a distance of three feet, a 40-Hz tone has to be 2 dB greater in sound-pressure level, while a 10-kHz tone must be 8 dB greater than the 1.5-kHz

[1] See references at end of chapter.

tone to be perceived as being as loud. At 50-dB spl, the noise level present in the average private business office, the level of a 30-Hz tone must be 30 dB greater, and a 10-kHz tone must be 14 dB greater than a 1.5-kHz tone to be perceived as being at the same volume. Thus, if a piece of music is monitored so that the signals produce a sound-pressure level of 110 dB, and it sounds well balanced, it will sound both bass and treble deficient when played at a level of 50-dB spl.

The loudness level of a tone can also affect the pitch the ear perceives. For example, if the intensity of a 100-Hz tone is increased from 40- to 100-dB spl, the pitch will decrease by about 10%. At 500 Hz, the pitch changes about 2% for the same increase in sound-pressure level.

As a result of the nonlinearity of the ear, tones can interact with each other within it, rather than being perceived separately. Three types of interaction effects occur: *beats*, *combination tones*, and *masking*.

Beats

Two tones which differ only a little in frequency and have ap-approximately the same amplitude will produce beats at the ear

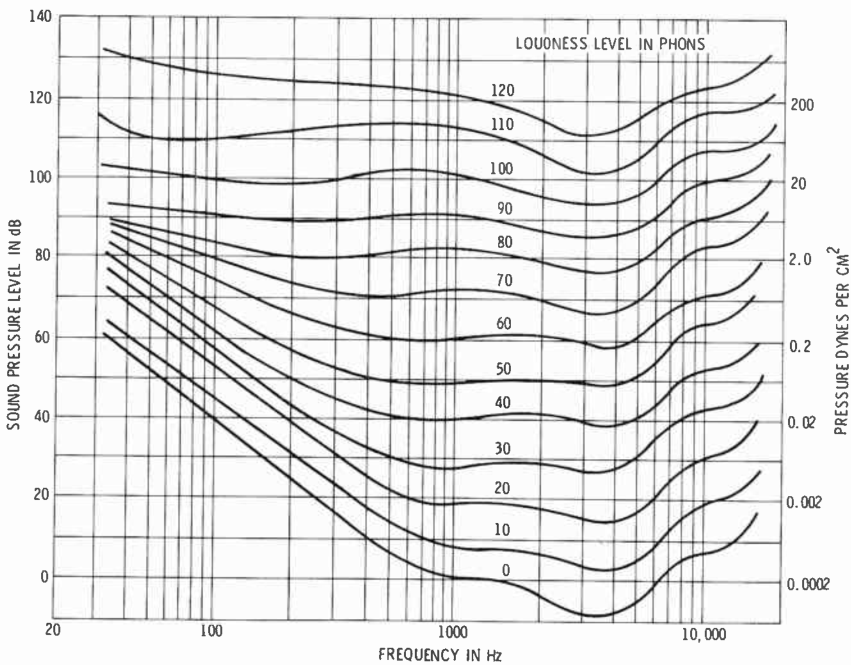


Fig. 2-15. The Fletcher-Munson equal-loudness contours.

equal to the difference between the two frequencies. The phenomenon of beats can be used as an aid in tuning instruments because the beats slow down and stop as the two notes approach and reach the same pitch. The beats are the results of the inability of the ear to separate closely pitched notes. The resulting synthesis of a third wave represents the addition of the two waves when they are in phase and the subtraction of their intensities when they are out of phase.

Combination Tones

Combination tones result when two loud tones differ by more than 50 Hz. The ear produces additional tones equal to both the sum and the difference of the two original tones and also equal to the sum and difference of their harmonics. The formula for computing the tones are: difference tones frequencies $= af_1 - bf_2$; the sum tone frequencies $= af_1 + bf_2$ (a and b are positive integers). The difference tones can be easily heard when they are below the frequency of both of the original tones. For example, 2000 and 2500 Hz produce a difference tone of 500 Hz.

Masking

Masking is the phenomenon by which loud sounds prevent the ear from hearing softer sounds. The greatest masking effect occurs when the frequency of the sound and the frequency of the masking noise are close to each other. For example, a 4-kHz tone will mask a softer 3.5-kHz tone but have little effect on the audibility of a quiet 1000-Hz tone. Masking can also be caused by harmonics of the masking tone, so a 1-kHz tone with a strong 2-kHz harmonic could mask a 1900-Hz tone.

PERCEPTION OF DIRECTION

One ear cannot discern the direction from which a sound comes, but two ears can. This is called the *binaural effect* and is made possible by four cues received by the ears:

(1) relative intensity,
(2) time of incidence of the sound,
(3) phase,
(4) complexity of the waveform.

Relative intensity refers to the fact that a sound coming from the right will reach the right ear at a higher intensity level than the left ear.

There are two reasons for this: (1) the intensity of a sound wave is inversely proportional to the square of the distance between the source and the listener so that the intensity at 6 feet from the source is one-fourth that of the intensity at 3 feet. Since the left ear is farther away from the source than the right, it receives a lower intensity. (2) The left ear is in an *acoustic shadow* cast by the head. The head blocks the direct sound waves and allows only reflected sound from surrounding surfaces to reach the left ear at the blocked frequencies, thus reducing the intensity of the sound perceived by the left ear. This effect is insignificant for low tones, because they bend around the head easily, but it is considerable for tones above 500 Hz because the effect increases as the sound-wave frequency rises. Because the path length to the right ear is shorter than that to the left ear, with respect to a sound source on the right, sudden changes in the sound pressure from the source will be sensed earlier by the right ear. Thus, if a wave begins, changes intensity, changes frequency, changes waveshape, or stops, the right ear detects it before the left ear.

For tones which are continuous, without changes of character, the ears detect the phase difference between them resulting from the time needed for a particular portion of the wave to travel from one side of the head to the other. The phase cue is most accurate in localizing low-frequency tones, where the path length between the ears is a wavelength or less. As the frequency increases, however, the wavelength decreases, and it is possible that the wave may appear in phase at both ears with the far ear actually hearing the wave several complete cycles after the near ear. Thus, phase cues alone are not much help at high frequencies.

The complexity of a waveform acts as a cue for the localization of compound tones and noises. The sound shadow created by the head attenuates the high frequencies, while low frequencies reach the far ear by bending around the head. As a result, the two ears hear a difference in the timbre of the waveform. The farther ear hears fewer high-frequency components.

To summarize: The accuracy of cues for localization varies with the frequency and complexity of the perceived tones. High-frequency tones are located by intensity, low-frequency tones by phase, complex tones and noise by a combination of intensity, time of incidence, and timbre. If there is no difference between what the left and right ears hear, the source appears to be the same distance from each ear, and the only place a single source could fulfill this requirement is directly in front of the listener. This phenomenon allows the recording engineer to place sound not only in the left and right speakers of a stereo system but also between the speakers. By feeding the same signal to both speakers, the ear hears the sound identically in both ears, and the source appears to be directly in front of the listener. By changing the proportions fed to the two speakers, the engineer can create the illusion that the sound source is anywhere between the two speakers and he can make the source appear to move. This technique is called *panning*. The other localization cues can also be used by the engineer to assign the source to locations between the two speakers through the use of electronic time delays, phase shifters, and filters. The ears can distinguish between a source in front of the listener and a source behind, above, or below him, through small movements of the head which provide slightly different perspectives of the sound due to the interference of the outer ear and the features of the face and body with the sound waves.

An interesting phenomenon that arises through the localization cues of the ear in connection with quad sound is that the same signal played through four speakers surrounding a listener produce the effect that the sound is coming from directly above the listener's head. This again is the result of the ear receiving the same cues from all directions and deducing that the only single source that could produce this effect would be one over the listener's head. If it were directly below the listener, his body would cause the sound to bend differently on its way to each ear and the ears would not receive the same cues.

WAVEFORM VALUES

Since waves are constantly varying in amplitude, how can they be said to have a single value, such as a level of 110-dB spl? The maximum value cannot be used because it is reached only for an instant on each side of the zero line during a cycle. In addition, waveforms of different shapes with the same peak amplitudes have different energy content and would not produce the same loudness. Taking the average over a number of complete cycles for any symmetrical wave would result in a value of zero, since the wave would have equal positive and negative excursions. The positive and negative maximums of a wave are called the *peak values*. The difference between the positive and negative peaks is called the *peak-to-peak value*, and the value of the amplitude at any one instant of time is called the *instantaneous value*. Fig. 2-16 shows positive peak A, negative peak B, and instantaneous values C, D, and E.

To get a meaningful average of all these values, the *rms* or *root-mean-square* value was developed. For a sine wave, this value is arrived at by squaring the amplitude of the wave at each point on the waveform, which brings the waveform above the zero line (thus it has a nonzero average) since the square of a negative number is a positive number, and the square of a positive number is also a positive number. The average of the squared waveform is taken by

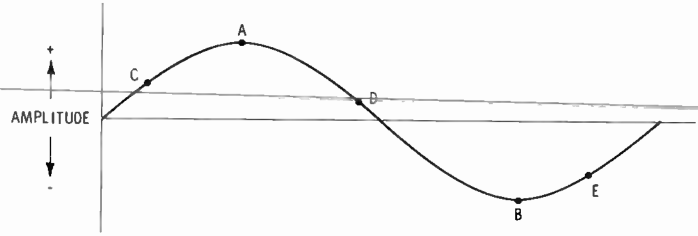
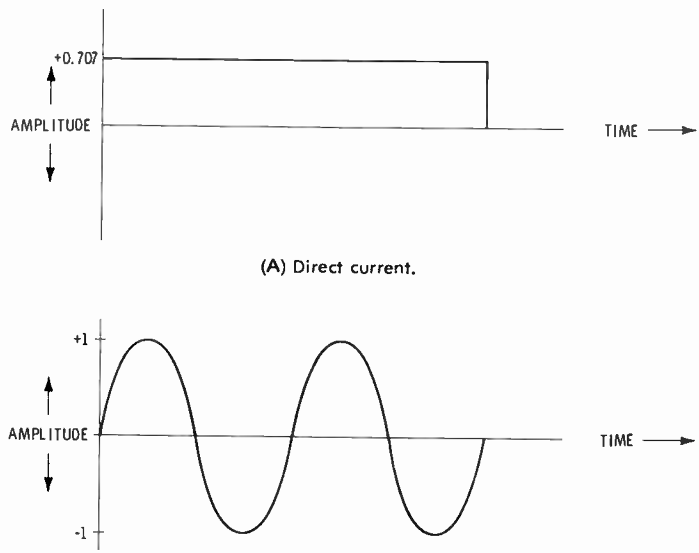


Fig. 2-16. The different values of waveform amplitude.

dividing the peak amplitude by two, and the rms value is found by taking the square root of the result. Beginning, for example, with a peak amplitude of 3 units, the rms value of this wave is the square root of the quantity (3 squared divided by 2) or 0.707 times 3. Thus the rms value of a sine wave is 0.707 times its peak amplitude. The computation of rms values of nonsinusoidal waves is much more difficult requiring that the rms value of each frequency component be computed and summed. All rms values have in common the fact that the rms value of the wave is equal to the value of continuous (i.e., direct current) signal that will produce the same amount of power as the wave in question (Fig. 2-17). For this reason, rms values are also referred to as *continuous* or *effective* values. Unless



(A) Direct current.

(B) Sine wave.

Fig. 2-17. The level of direct current produces the same amount of power as the sine wave.

another value is specified when speaking quantitatively about a wave, this book refers to the rms value.

LOUDNESS LEVELS

Because the ear hears over such a large energy range, a logarithmic scale has been adopted to compress the measurements of sound into more workable figures. The system used is the decibel (dB). The decibel has no units attached to it, rather it expresses the ratio of two powers according to the formula $\text{dB} = 10 \log P_1/P_2$. The answer is positive if P_1 is greater than P_2 and negative if P_2 is greater than P_1. If we have two amplifiers such that $P_1 = 60$ watts and $P_2 = 40$ watts, we can express their power ratio in dB as $10 \log 60/40 = 10 \log 1.5 = 10 \times 0.176 = 1.76$ dB. We can say that the 60-watt amplifier is 1.76 dB more powerful than the 40-watt amplifier. We could also say that the power output of the 40-watt amp is 1.76 dB below that of the 60-watt amp, or that the power of the 40-watt amp is -1.76 dB with respect to the 60-watt amp. Since the ear responds proportionally to the power level in dB, the 60-watt amp would be 1.76 dB louder than the 40-watt amp if they were alternately connected to the same speaker. As mentioned previously, the ear requires equal ratios of power changes to produce equal loudness increases or decreases. Thus a 1.76-dB power increase from 40 to 60 watts increases the loudness by the same amount as an increase from 60 to 90 watts because this is also an increase of 1.76 dB. Under average conditions, the minimum level change that the ear can perceive is about 1.0 dB, although under laboratory or studio conditions of switching sound levels with no appreciable time delay between them, the sensitivity to change can be as low as 0.25 dB at certain frequencies. Thus, an increase from 40 to 60 watts of power does not bring about much of a loudness increase.

The ear does not hear power directly, the power first must be converted into sound waves. The intensity of the sound wave produced is directly proportional to the power which produced it, and therefore the ratio of two intensities is equal to the ratios of the two powers which produced them. The ratio of two powers in dB can be computed from their intensity values by plugging the intensity ratio into the dB formula for power, i.e., $\text{dB} = 10 \log I_1/I_2$. The intensity of sound is usually measured indirectly, through the measurement of sound-pressure levels (spl). Since the intensity of a sound wave is proportional to the square of the sound pressure, the power in the wave is also proportional to the square of the sound pressure. The ratio of the squares of two sound pressures is therefore equal to the ratio of the powers that produced them, i.e., $I_1/I_2 = (\text{spl}_1)^2/(\text{spl}_2)^2 = P_1/P_2$. Power ratios in dB can therefore be expressed by substituting the ratio of the squares of the sound-pressure levels generated into the dB formula for power, thus: $dB = 10 \log P_1/P_2 = 10 \log (spl_1)^2/(spl_2)^2 = 10 \log (spl_1/spl_2)^2 = 20 \log spl_1/spl_2$. The multiplier in front of the formula changes to 20 for spl ratios because squaring a number of doubles the value of its logarithm.

Since the dB system only provides information about the ratio of two powers, it is useful to have a common point, which the number of dB refers to, so that absolute levels can be discussed in terms of dB. For example, if there is a 3-dB difference between two powers, their ratio is two to one. From this information alone, it is impossible to determine whether the power levels in question are 2 watts and 1 watt, or 2000 watts and 1000 watts. If, however, a reference level of 1 watt is chosen and a power level is said to be 3 dB above the reference, its level must be twice the reference power, or 2 watts.

Threshold of Hearing

In the case of spl, a convenient reference-pressure level is that of the *threshold of hearing*, which is the minimum sound pressure that produces the phenomenon of hearing in most people, and is equal to 0.0002 microbar. One microbar is equal to one-millionth of normal atmospheric pressure, so it is apparent that the ear is extremely sensitive. In fact, if the air was any more sensitive, the thermal motion of the molecules of the air would be audible. The use of 0.0002 microbar as the reference level, spl_2, in the dB formula is indicated by expressing a value as a certain number of dB spl. The reference pressure level is called 0-dB spl.

The *threshold of hearing* is defined as the spl for a specific frequency at which the average person can hear only 50% of the time.

Threshold of Feeling

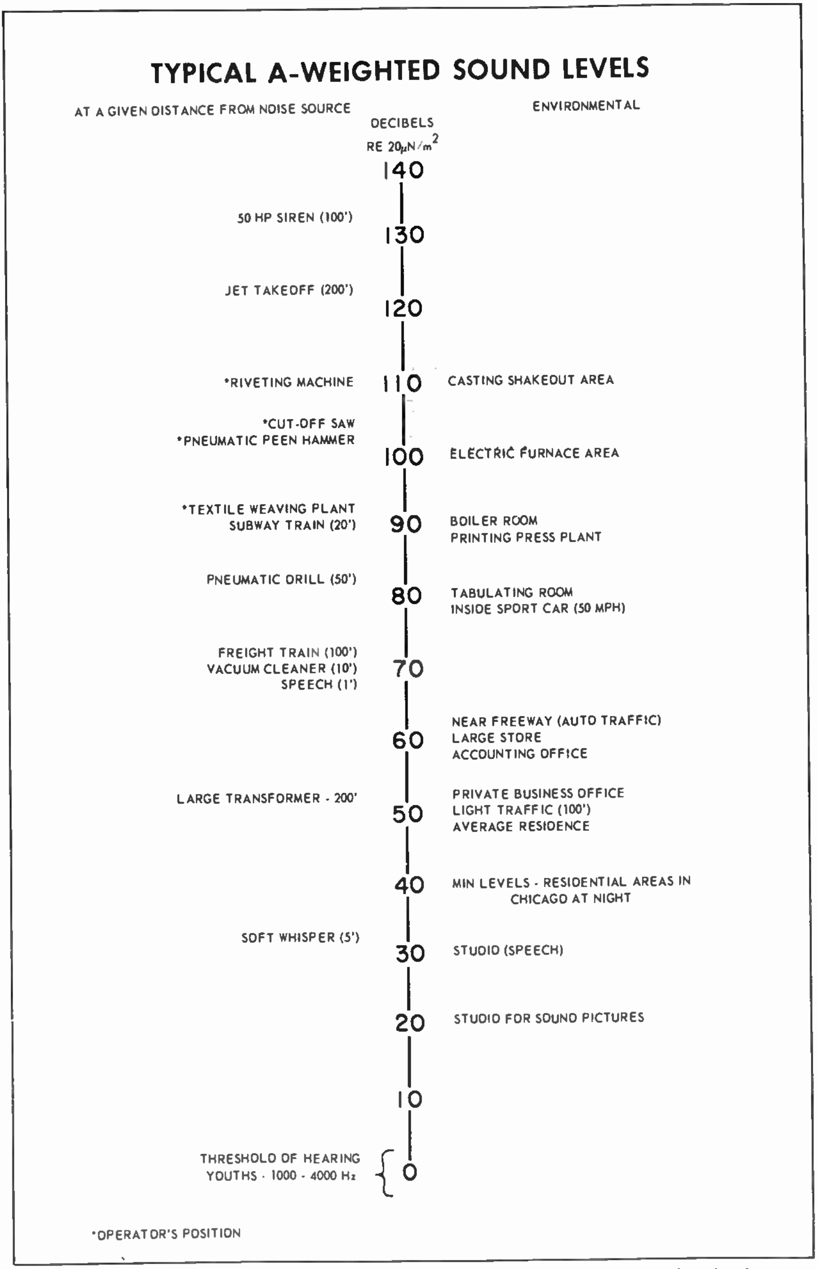
The spl that will cause discomfort in a listener 50% of the time is called the *threshold of feeling* and occurs at a level of about 118-dB spl between 200 Hz and 10 kHz.

Threshold of Pain

The spl which causes pain in a listener 50% of the time is called the *threshold of pain* and corresponds to a spl of 140 dB in the range between 200 Hz and 10 kHz. Fig. 2-18[3] shows typical spls for different sounds. The levels are weighted to take into account the reduced sensitivity of human hearing to low frequencies.

The Phon

The unit of loudness level is called the *phon*. Phons are numerically equal to dB spl at 1 kHz and at other frequencies are related to dB spl by the Fletcher-Munson curves. Thus, the loudness level



Courtesy General Radio Company

Fig. 2-18. Typical spls for common sounds. The reference level of 20 $\mu N/m^2$ is equivalent to 0.0002 microbar (or 0.002 dynes/cm^2).

of a sound in phons is the number of dB spl produced by a 1-kHz sine wave which sounds equally loud. For example, while 40 phons at one frequency is as loud as 40 phons at any other frequency, 40 phons at 1 kHz requires 40-dB spl, while 40 phons at 10 kHz requires 52-dB spl.

The Sone

Since the perception of the ear to loudness is not directly proportional to the sound-pressure level, a scale has been devised to facilitate the discussion of the loudness of one sound compared to that of another. This unit of loudness is the *sone,* and one sone is defined as the loudness of a 1-kHz sine wave at 40-dB spl. A sine wave is specified because the loudness depends on both the frequency and complexity of the waveform. A loudness of two sones is twice as loud as one sone and is equal to a level of 50-dB spl at 1 kHz. In the recording studio, the use of dB spl re 0.0002 microbar is more common than phons or sones. These latter are used more by acoustical engineers in designing and examining rooms for specific acoustic properties.

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CHAPTER 3

Microphones

Three types of microphones are used in recording studios: dynamic, ribbon, and condenser.

THE DYNAMIC MICROPHONE

A *dynamic* or moving-coil mike is a pressure-operated device. It consists of a fine coil of wire attached to a delicate *diaphragm* and suspended in a permanent magnetic field (Fig. 3-1). When sound pressure waves hit the diaphragm, the diaphragm moves the coil in proportion to the wave intensity and causes the coil to cut across the fixed lines of magnetic flux supplied by the permanent magnet. The amplitude of the current flow induced in the coil is proportional to the number of lines of flux the coil cuts (that is, how far it moves from its no-signal position), and the speed at which the coil cuts the

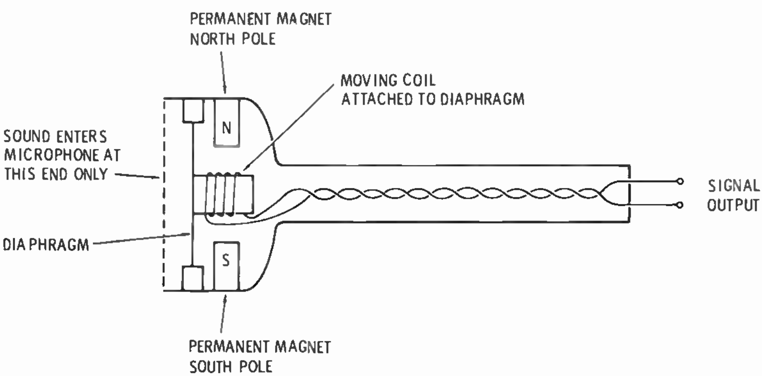


Fig. 3-1. The components of a dynamic microphone.

lines. The frequency of the signal is determined by how often the diaphragm reverses its direction of travel.

Basic pressure mikes are inherently omnidirectional. This means that they are equally sensitive to sound waves coming from any direction (Fig. 3-2). Since signals can only reach the front of the diaphragm, they all add together in the mike. Dynamic mikes can be designed to be directional. This means that they will be more sensitive to signals arriving from one direction than from another. The most common of these directional characteristics is the *cardioid* (Fig. 3-3). Its name arises from its heart-shaped pickup pattern.

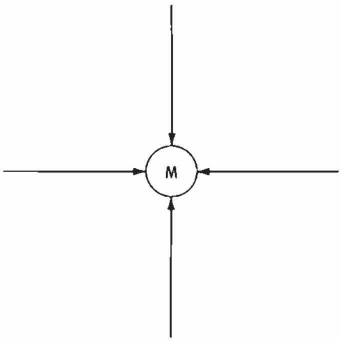


Fig. 3-2. A pressure-operated mike is sensitive to sound waves originating from any direction.

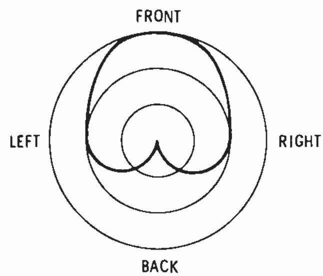
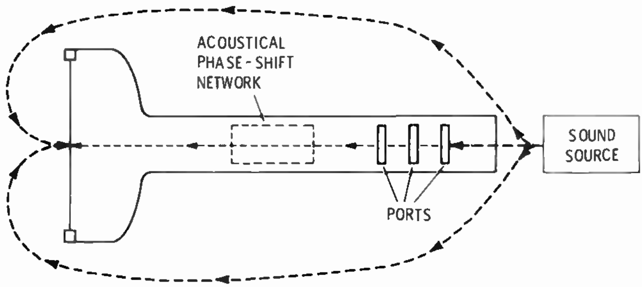


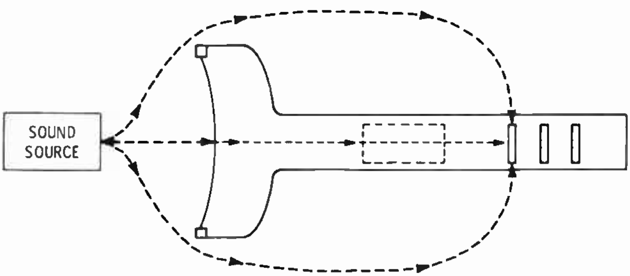
Fig. 3-3. A cardioid pickup pattern.

Signals arriving from the rear of the mike are acoustically phase shifted in the body of the mike and applied to the back of the diaphragm so that a signal arriving from the rear of the mike will be applied equally to both the front and back of the diaphragm and generate no output (Fig. 3-4A). Signals arriving from the front reach the diaphragm without any phase shifting. In Fig. 3-4B, a signal from the front which enters the ports is phase shifted twice: once by the time it takes for the wave to travel the external distance from the diaphragm to the ports and a second time by the phase-shifted network inside the case. When this wave reaches the back of the diaphragm, it is in phase with the wave at the front and reinforces it. Between the front and rear source locations, the signal entering the ports partially cancels and partially reinforces the signal at the front of the diaphragm, producing the cardioid pickup pattern.

The attenuation of an equal signal which is at the same distance from a cardioid mike but 180° off-axis with respect to an on-axis signal is called the *front-to-back discrimination.* The axis of a mike is the imaginary line drawn perpendicular to the plane of the dia-



(A) Signals arrive at front and rear of the diaphragm.



(B) Signal through port is shifted to be in phase with front signal.

Fig. 3-4. Directional operation of the cardioid microphone.

phragm (Fig. 3-5). Other directional designs, called *hypercardioid* and *supercardioid*, sacrifice some front-to-back discrimination in favor of a narrower pickup pattern (Fig. 3-6A). Fig. 3-6B shows the frequency response on-axis, 180° off-axis, and 150° off-axis. Regular cardioids have maximum off-axis rejection at 180°, while supercardioids such as the Electro-Voice RE16 (Fig. 3-7) have maximum rejection at 150°.

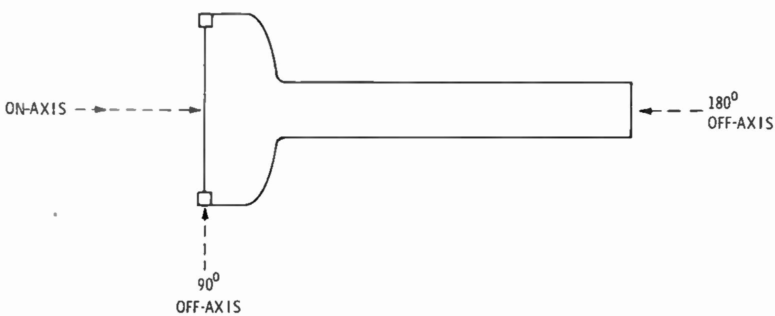
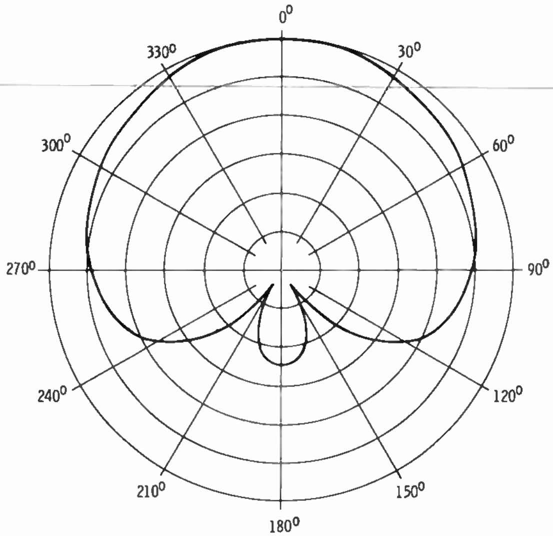
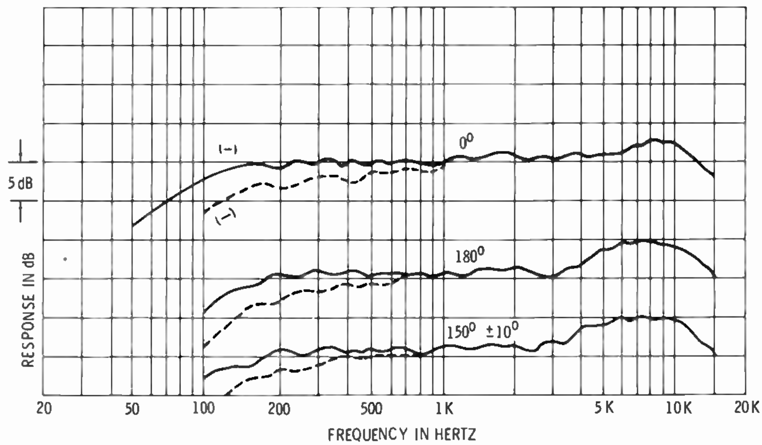


Fig. 3-5. A microphone axis is the line perpendicular to the diaphragm on the side exposed to the sound waves.



SCALE IS 5 dB PER DIVISION

(A) Pickup pattern.



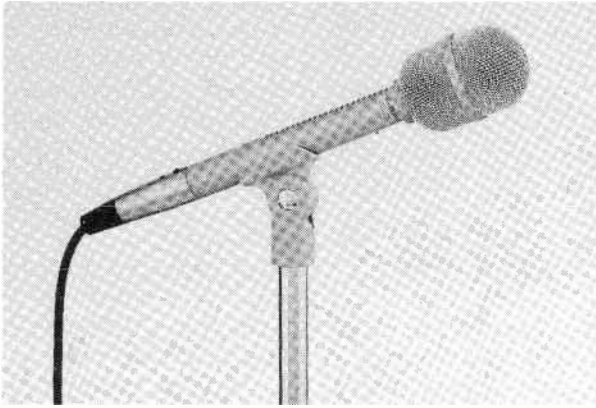
(B) Frequency response.

Courtesy Electro-Voice Inc.

Fig. 3-6. The Electro-Voice RE-16 supercardioid pickup pattern.

THE RIBBON MICROPHONE

The *ribbon* mike, also called a *pressure gradient* or *velocity* mike, utilizes a thin metal ribbon suspended between the poles of a magnet to sense the sound wave (Fig. 3-8). When the ribbon moves, it



Courtesy Electro-Voice Inc.

Fig. 3-7. The Electro-Voice RE-16 dynamic supercardioid microphone.

cuts through the lines of flux generated by the permanent magnet, and this flux induces a voltage in the ribbon. This voltage becomes the signal output. The mike is called pressure gradient because the motion of the ribbon is determined by the difference in pressure between its front and back sides. This difference is proportional to the velocity of the air molecules which make up the wave, and thus, the third name, velocity mike.

Since the ribbon is exposed to sound waves at both its front and rear (Fig. 3-9A), it is equally sensitive to sounds from both directions with sound from the rear producing a voltage 180° out of phase with the voltage due to sound from the front. Sound waves 90° off-axis produce equal, but opposite, pressures at the front and rear of the ribbon (Fig. 3-9B) and cancel. As a result, the ribbon mike is inherently bidirectional, with a figure-eight shaped pickup pattern (Fig. 3-10). Sounds arriving from the sides of the ribbon

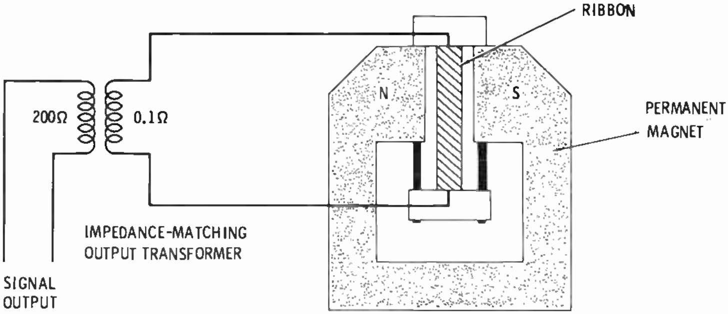
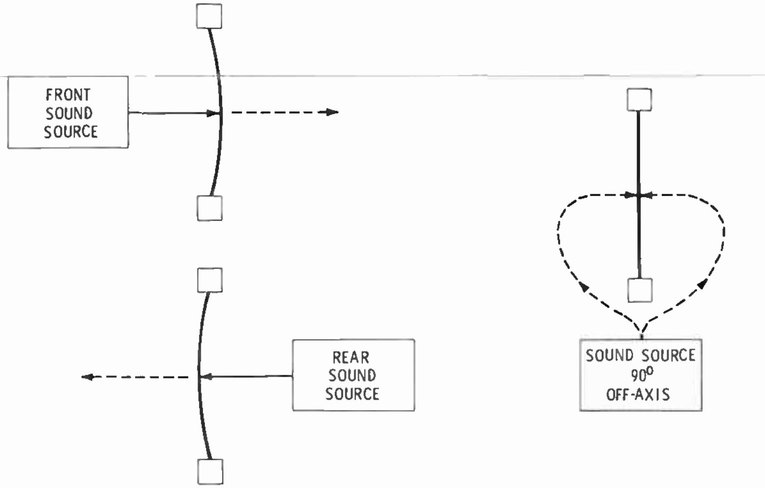


Fig. 3-8. The basic components of a ribbon microphone.



(A) The ribbon is sensitive to sounds at the front and rear.

(B) Sound waves from 90° off-axis.

Fig. 3-9. Sound sources on-axis and 90° off-axis at a ribbon mike.

create equal but opposite pressures on it (no pressure difference) and cancel each other out.

Other directional patterns may be obtained by closing off the acoustic path to the rear of the ribbon (Fig. 3-11). If this path is completely closed, the mike becomes an omnidirectional or nondirectional pressure-operated mike. As the path is gradually opened, the pattern becomes more and more cardioid, then supercardioid, and finally when the path is fully opened, figure-eight.

THE CONDENSER MICROPHONE

Condenser microphones use electrostatic principles rather than the electromagnetic principles used by dynamic and ribbon mikes.

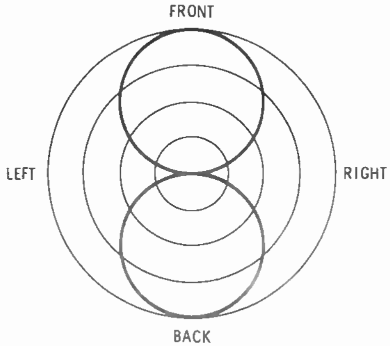
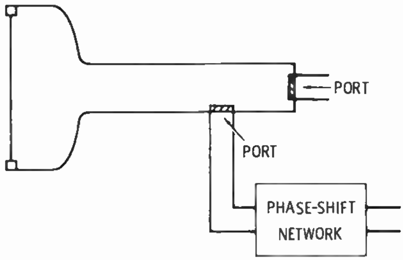
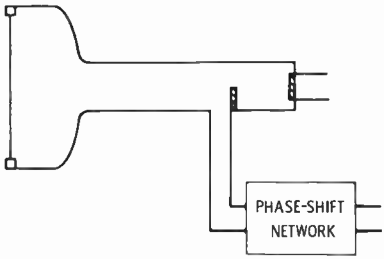


Fig. 3-10. The normal pickup pattern of a ribbon microphone.



(A) Both ports are closed. The ribbon becomes a pressure-operated omnidirectional mike.



(B) Port to phase-shift network is open and door to other port is closed. The pattern becomes cardioid.

(C) Port to phase-shift network is closed and other port is open. The pattern is figure-eight.

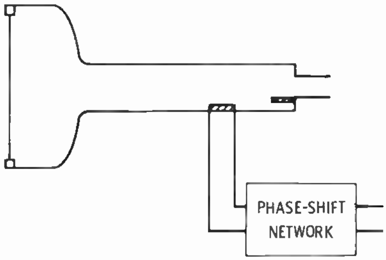


Fig. 3-11. A method of producing variable pickup patterns from a ribbon microphone using ports and an acoustical phase-shift network.

The *head*, or *capsule*, of the mike consists of two very thin plates, one movable and one fixed. These two plates form a *capacitor* (formerly called *condensers*, and hence the name condenser mikes) (Fig. 3-12). A capacitor is an electrical device which is capable of storing an electric charge. The amount of charge a capacitor can store is determined by its value of capacitance and the applied voltage, according to the formula:

$$Q = CV \qquad \text{(Eq. 3-1)}$$

where,

Q is the charge, in coulombs,
C is the capacitance, in farads,
V is the voltage, in volts.

The capacitance of the capsule is determined by the composition and surface area of the plates (which are fixed), the *dielectric* or substance between the plates (which is air and fixed), and the distance between the plates (which varies with sound pressure). The plates of the mike capsule form a sound-pressure-sensitive capacitor.

In the design used by Neumann, the plates are connected to opposite sides of a dc power supply which provides a *polarizing voltage* for the capacitor (Fig. 3-13). Electrons are drawn from the plate

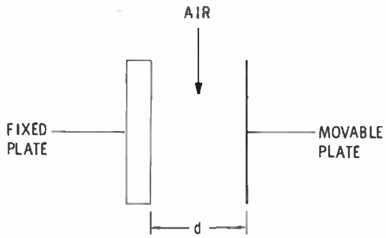


Fig. 3-12. In a condenser microphone capsule the sound pressure varies distance d between plates.

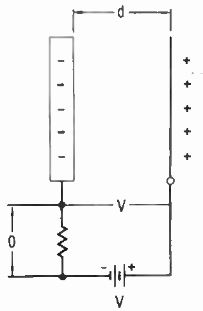


Fig. 3-13. The power supply charges the capacitor through a high-value resistor.

connected to the positive side of the power supply and forced through a high-value resistor onto the plate connected to the negative side of the supply. This continues to occur until the charge on the capsule (that is, the difference between the number of electrons on the positive and negative plates) is equal to the capacitance of the capsule times the polarizing voltage. When this equilibrium is reached, no further appreciable current flows through the resistor. If the mike is fed a sound-pressure wave, the capacitance of the head changes. When the distance between the plates decreases, the capacitance increases, and when the distance increases, the capacitance decreases. According to Equation 3-1, Q, C, and V are interrelated, so a change in C must cause a change in Q and/or V.

The resistor high value, in conjunction with the capacitance of the plates, produces a circuit *time constant* longer than a cycle of

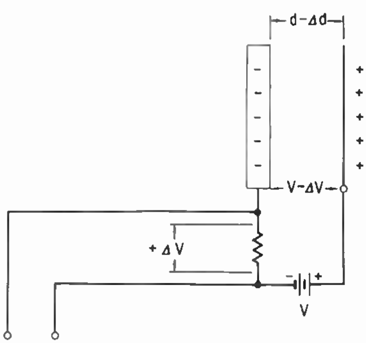


Fig. 3-14. If a sound wave decreases the plate spacing by Δ d, the capacitance increases by Δ C and the voltage across the plates falls by Δ V.

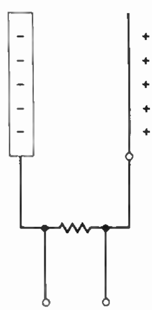


Fig. 3-15. Diagram of an electret condenser microphone. Its circuit is equivalent to that of Fig. 3-13.

an audio frequency. The time constant of a circuit is a measure of the time needed for a capacitor to charge or discharge. Since the resistor prevents the capacitor charge from varying with the rapid changes in capacitance caused by the applied sound pressure, the voltage across the capacitor must change according to $\Delta V = Q/\Delta C$ (Fig. 3-14). The resistor and capacitor are in series with the power supply, so the sum of the voltage drops across them must equal the supply voltage. When the voltage across the capacitor changes, the voltage across the resistor changes equally but in the opposite direction. The voltage across the resistor becomes the output signal. Since it is a low-level, high-impedance signal, it is amplified within the body of the microphone to prevent the hum and noise pickup and signal level losses that would occur (due to the resistance of the cables) if the amplifier was at a distance from the capsule. This amplifier is another reason that condenser mikes need power supplies.

Electret condenser microphones operate on the same principle as the Neumann microphones except that the polarizing voltage is permanently stored on the capsule plates so that the power supply need only supply amplifier power (Fig. 3-15).

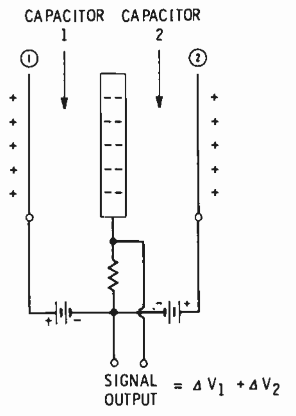


Fig. 3-16. An omnidirectional condenser microphone diagram.

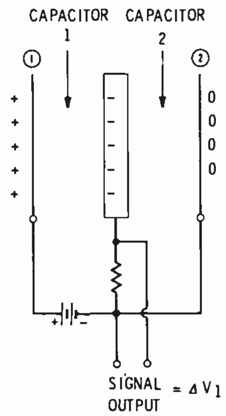


Fig. 3-17. A cardioid condenser microphone.

A different condenser design is used by Sennheiser in their mikes. Rather than applying a polarizing voltage to the capsule and sensing its varying charge-holding capability, the capacitor is used as part of the tuning circuit for a high-frequency oscillator. Variations in sound pressure cause frequency modulation of the oscillator (variations in oscillator frequency proportional to the intensity of the sound waves), resulting in a signal similar to that broadcast by an fm radio station. The high-frequency signal is converted into audio the same way that an fm radio signal is. The lack of polarizing voltage makes the mike less sensitive to physical shock and to changes in humidity or temperature which can cause arcing of the polarizing voltage between the plates.

Condenser mikes are inherently pressure operated, and thus, omnidirectional. Other directional patterns can be achieved through the use of a perforated fixed plate and a second movable plate in the capsule. By placing the two movable plates on opposite sides of the fixed plate, two capacitors are formed. By polarizing the two movable plates to one polarity and polarizing the fixed plate to the opposite polarity (Fig. 3-16), an omnidirectional pickup pattern is achieved. As the intensity of the voltage on one of the movable plates is decreased while holding the polarization intensity of the other plate unchanged (Fig. 3-17), the pickup pattern becomes more and more cardioid. When the movable plates are at opposite polarities and the fixed plate is at ground potential (Fig. 3-18), a figure-eight pattern is achieved.

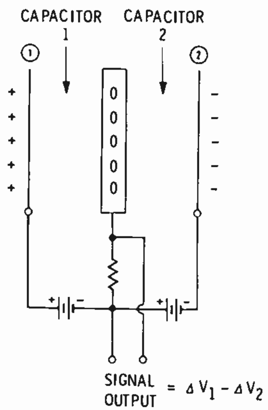


Fig. 3-18. A figure-eight condenser microphone. Operation is the same as in Fig. 3-16 except that capacitors 1 and 2 produce voltages across the resistor that are 180° out of phase with each other.

Another method of obtaining a variable directional characteristic is to use two completely separate microphone elements in the same body, one unit with an omnidirectional characteristic and the other with a figure-eight characteristic. A cardioid pattern is achieved by connecting the two elements in parallel. The voltage due to sound arriving at the rear of the bidirectional element is out of phase with that produced by the omnidirectional element, reducing sensitivity to sounds from the rear. Output from a source 90° off-axis is due to the omnidirectional element only, while the output of the two elements are equal and in phase for a source on-axis and produce twice the output (Fig. 3-19). This method may be used with either electrostatic or electromagnetic mikes. In addition, the elements should be as physically close as possible to prevent phase differences in the pickup of on-axis signals, but if necessary, two discrete microphones can be used and combined in a mixing console. As the ratio of omnidirectional to bidirectional pickup is increased, the directional pattern varies from figure-eight through various degrees of supercardioid, to cardioid, to omnidirectional. Thus, any desired directional pattern can be achieved using two mikes if a mike with the proper pattern is not available.

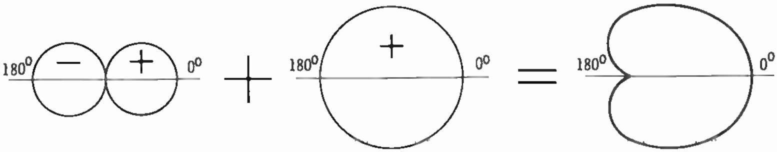


Fig. 3-19. A bidirectional pattern added to an omnidirectional pattern produces a cardioid pattern.

Because condenser mikes have built-in amplifiers, many of them also have built-in attenuation *pads* immediately after the capsule output. This prevents overloading the preamp contained in the mike when exposed to very high sound-pressure levels. If the capsule output level were high enough to overload this amp, the output signal fed to the mike preamp in the control room would be distorted, and no amount of control room preamp padding would solve the problem. The pad within the mike itself avoids this problem and should be used if the signal to be picked up is very loud.

Directional mikes have the property that their bass response increases as the signal source gets closer to the mike. This *proximity effect* is noticeable when the source is closer than about two feet from the mike, and the effect increases as the mike is approached closer. It is caused by the increase in pressure gradients at short mike-to-source distances. At two feet the sound intensity is four times that at four feet from the source, while at one foot the intensity of the sound is sixteen times that at four feet. This increase in pressure gradient occurs throughout the audio spectrum, but there is a greater phase change between the front and rear of the mike at the higher frequencies due to their shorter wavelengths. This causes some of the pressure gradient to be cancelled for high frequencies, producing a bass boost. The boost is somewhat greater in bidirectional mikes than in cardioids. To compensate for this effect, a bass rolloff switch is often provided to reduce the bass response back to flat. In microphones designed for close-up work, the frequency response may be rolled off at the low end, and the proximity effect is used to restore the bass response.

Dynamic mikes tend to be more rugged than condenser or ribbon mikes and also less expensive. Ribbon mikes can be damaged by physical shocks which can break the thin ribbon. The dynamic and ribbon types do not need power supplies to operate and are not as sensitive to changes in humidity or temperature as condenser mikes. Condenser mikes, on the other hand, have better transient response (faster response to rapid changes in sound pressure), flatter frequency response (especially in the high end), and much higher output levels due to their built-in amplifiers. Although ribbon mikes are still in use today, they were more popular during the early days of radio. Only a few new mikes are now designed using ribbons.

There are several other types of microphones, such as the carbon mikes used in telephones and the crystal mikes supplied with inexpensive tape recorders, but these are too noisy and their frequency characteristics are too poor for use in professional recording.

MICROPHONE IMPEDANCE

Microphones are available with different output impedances. The output impedance is a rating used to match the signal-providing capability of one device with the signal-drawing (input impedance) requirements of another device. Impedance is measured in ohms, and its symbol is Z. Commonly used mike output impedances are 50 ohms, 150 to 250 ohms (low) and 20 to 50K (high). Each impedance range has its advantage. In the past, high-impedance mikes were less expensive to use because the input impedance of tube-type amplifiers was high and they required expensive input transformers to be used with low-impedance mikes. All dynamic mikes, however, are low-impedance devices, and those with high-impedance outputs achieve them through the use of a built-in impedance step-up transformer. A disadvantage of high-impedance mikes is the susceptibility of high-impedance mike lines to the pickup of electrostatic noise, such as that caused by fluorescent lights and motors. This makes the use of shielded cable necessary. In addition, the use of a conductor surrounded by a shield creates a capacitor which is in effect connected across the output of the microphone. As the length of the cable increases, the capacitance increases until at about 20 to 25 feet the cable capacitance begins to short out much of the high-frequency information picked up by the mike. Thus, good results with high-impedance mikes are limited to use with cable lengths of 25 feet or less.

Very low-impedance (50 ohms) mikes have the advantage that their mike lines are fairly insensitive to electrostatic pickup. They are sensitive, however, to induced hum pickup from electromagnetic fields, such as those generated by ac power lines. This pickup can be eliminated by using twisted-pair cable. The currents magnetically induced in this cable will flow in opposite directions and cancel each other out in the input transformer of the mike preamp. The 50-ohm lines do not need shielding, for this is effective only against electrostatic, not electromagnetic pickup. Cable lengths are limited to about 100 feet due to signal power losses resulting from cable resistance. These losses, however, do not degrade frequency response, merely signal-to-noise ratio. They can be reduced by using larger conductors which have less resistance.

The 150- to 250-ohm mike lines are less susceptible to electromagnetic pickup than 50-ohm lines but more susceptible than high impedance. They are also more susceptible to electrostatic pickup than 50-ohm lines but less susceptible to this type of pickup than high-impedance lines. As a result, shielded, twisted-pair cable is used, and lowest noise is obtained with the use of *balanced* input circuits. This means that two wires carry the signal voltage and that a shield is wrapped around them and connected to ground. Neither of the two signal lines is grounded. High-impedance lines use *unbalanced* circuits in which one signal lead flows through the center of the cable and is surrounded by a shield which is grounded and used as the second signal lead. Balanced circuits are more expensive than unbalanced ones, due to the cost of balanced transformers which are necessary in the input circuit of the amplifiers they are used with. The 150- to 250-ohm mike lines have low signal losses and can be used with cable lengths up to several thousand feet. The mike lines in most recording studios are 200-ohm balanced lines with the shield grounded at the preamp end only. This is done to prevent *ground loops* which can produce hum in the signal if shields are grounded at more than one point.

The Neumann FET-80 series microphones such as the U47-FET, U87, and KM84, as well as certain transistorized condenser microphones made by Sony and AKG, can be powered by a *phantom* power supply which provides condenser mike power to all mike inputs simultaneously. The phantom powering system does not interfere with the operation of dynamic mikes. The positive side of the power is fed to both sides of each mike line through identical value resistors so that there is no voltage differential between the two signal-carrying leads (Fig. 3-20). The negative side of the supply is connected to the shield. Condenser microphones designed to make use of this voltage no longer need internal batteries, external battery packs, or individual ac-operated power supplies. The resistors used in connecting the phantom supply to the mike inputs provide isolation between the two conductors of each mike line and between mikes. Shorting either of the signal leads to ground on one input does not affect the phantom power delivered to the other inputs.

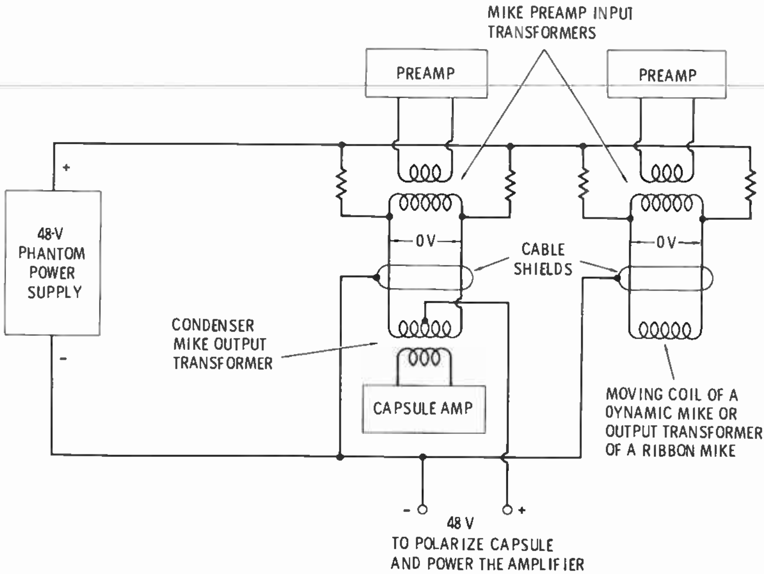


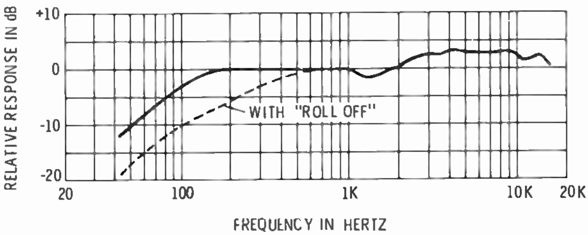
Fig. 3-20. The phantom powering system permits simultaneous operation of phantom-powered condenser microphones and dynamic or ribbon microphones.

However, if two or more inputs are shorted, the phantom voltage may fall too low to be usable. This can happen if mike cables become defective and one or both signal leads short to the shield. The size of the resistors used with the phantom supply depends on the voltage and current requirements of the microphones to be powered. The Neumann U87 contains a meter which indicates when the power supplied to it by either internal batteries or a phantom supply is adequate for proper operation.

SPECIFICATIONS

The *frequency range* of a mike is meaningless because it is the shape of the frequency-response curve that determines how it sounds. Therefore, it is important to see not only how many dB plus or minus the response is, but also to see an on-axis curve of where the pluses and minuses occur (Fig. 3-21).

The next significant piece of data is the directional characteristic of the mike. This is also meaningless without a frequency versus number of degrees off-axis curve. All directional mikes have different frequency response off-axis than they do on-axis, but the better ones maintain the on-axis characteristics over a wider pickup angle. The directional information is given on a circular *polar pattern* (Fig. 3-22), with differently coded lines corresponding to different frequencies. The ideal mike would have the same frequency response at all angles. The diagrams are often divided into two sections and only one-half of the polar plot is shown for each frequency so that more frequencies can be shown without cluttering up the diagram. It may be assumed that the curves are symmetrical around the zero-degree axis.

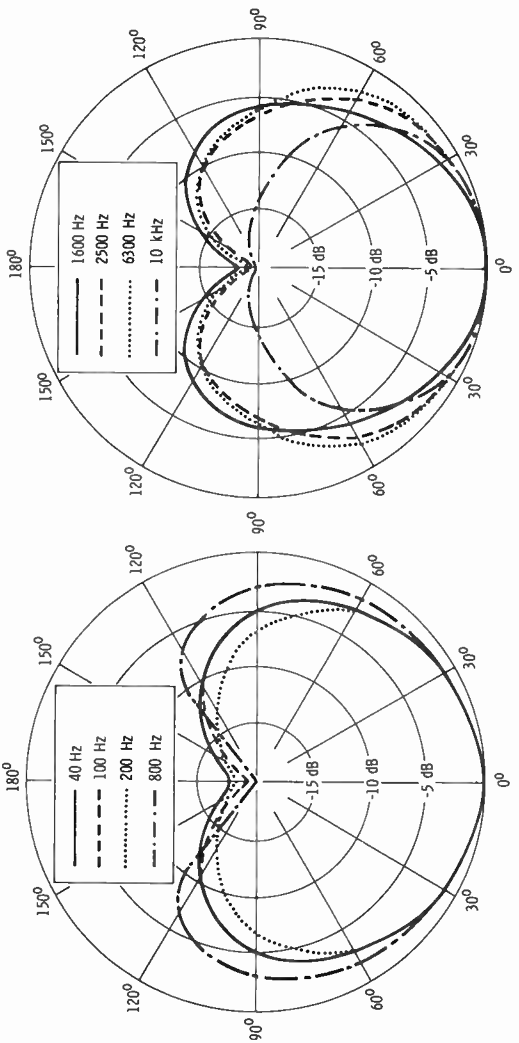


Courtesy Shure Brothers, Inc.

Fig. 3-21. The Shure SM53 dynamic cardioid microphone on-axis frequency-response curve.

The next most significant specification is the *output impedance*. If the mike is not connected to the proper load, the frequency response and directional characteristics shown in the curves will be degraded.

Next in importance are the mike *sensitivity* and *self-noise level*. These figures determine the best signal-to-noise ratio possible with the mike. The higher the output level, the less amplification is needed from the preamp. Using a variable-gain preamp, gain can be set to exactly that needed so as to not unnecessarily amplify thermal noise. Another factor which must be considered is that if a high-output mike is used without a variable-gain preamp, mike pads must be available in order to prevent the preamp from being overloaded on loud passages. So, a high-output mike is not always better than one with lower output. The sensitivity is determined by measuring the mike output into its rated impedance load with a certain sound-pressure level at the diaphragm of the mike. In comparing the sensitivity of two mikes, they must both receive the same sound-pressure level at the same frequency. The most common sound-pressure level used is 10 dynes/cm^2 (10 microbars) at 1 kHz, and the rating is given in dB re 1 milliwatt (dBm). The self-noise level is the noise level generated by the mike when no sound reaches it. In a dynamic or ribbon mike this is the thermal noise caused by electrons moving in the coil or ribbon, while in condenser mikes it is mainly due to noise in the built-in amp. For any given mike, this noise cannot be eliminated, and it places a restraint on how good the signal-to-noise ratio can be.



Courtesy Shure Brothers, Inc.

Fig. 3-22. Polar response curves for the SM53 microphone.

Probably the least important specification for recording use is the level at which the mike begins to create harmonic distortion. This is relatively unimportant because all modern mikes can withstand very intense sound pressure without creating appreciable harmonic distortion. The rating is usually given in dB of sound-pressure level for 0.5% total harmonic distortion.

Typical output levels for dynamic and ribbon mikes are on the order of −55 dBm (re 10 microbars) and −40 dBm for condenser mikes. These levels are too low to send directly to the recording console or tape machine, so specially designed mike preamps are used to raise these levels. These preamps must have a very low *equivalent input noise* (e.i.n.) figure. The lower this figure, the less noise will be added to the signal before it is amplified, and since both signal and noise will be amplified, the lower the noise will be in the final signal. The signal-to-noise ratio is the difference in dB between the level of signal and the level of the noise. For example, Electrodyne LA602s used as mike preamps have an equivalent input noise of −127 dBm and can be set for a fixed gain of 45 dB. If an input signal of −55 dBm is fed into the preamp, the signal-to-noise ratio at the input of the amp is −55-dBm signal to −127-dBm noise, or a signal-to-noise ratio of 72 dB. At the output of the preamp there is a signal level of −10 dBm and a noise level of −82 dBm, so that the signal-to-noise ratio is the same. Thus, it is the level of signal with respect to the noise at the preamp input which sets the signal-to-noise ratio. If a 10-dB pad is used before the preamp input to decrease the mike signal to −65 dBm, the maximum signal-to-noise ratio would be only 62 dB. Therefore, pads should only be used when they are needed to prevent mike preamp overload distortion.

The signal-to-noise ratio at the preamp output can be degraded through poor console-operating procedures. As the signal flows through the console, it passes through several stages of attenuation, each of which is followed by amplification. These stages are distributed rather than lumped together in one fader and one amplifier to prevent noise buildup in the console. As long as the signal level stays high enough so that the equivalent input noise of the next amplifier is far enough below the level of the noise component already present in the signal so that its contribution to the amplifier output is insignificant, attenuation and reamplification does not degrade the signal-to-noise ratio. The maximum amount of attenuation in dB that can be inserted between the output of amplifier 1 and the input of amplifier 2 and cause less than 0.1-dB reduction in the signal-to-noise ratio can be computed by the following formula:

$$\text{dB} = \text{Gain}_1 - 20\ \text{dB} - (\text{e.i.n.}_2 - \text{e.i.n.}_1).$$

As attenuation is increased beyond this amount, both the signal and the noise level fed to amplifier 2 decrease, but the equivalent input noise of amplifier 2 does not. Therefore, e.i.n.$_2$ becomes a more and more significant portion of the amplifier input signal. For example, beginning with a 72-dB signal-to-noise ratio of −10-dBm signal and −82-dBm noise at the output of a 45-dB gain mike preamp, attenuating by 25 dB produces a −35-dBm signal and a −107-dBm noise level. If the equivalent input noise of the next amplifier is equal to that of the mike preamp (−127 dBm), this is a permissible amount of attenuation, and the e.i.n. of the next stage does not contribute significantly to the noise level. If the gain of the next stage is set to 25 dB, the signal and noise levels are brought back up to −10 dBm and −82 dBm, respectively. The signal-to-noise ratio is unchanged throughout the various level changes, remaining at 72 dB (Fig. 3-23). Starting with a mike output level of −55 dBm and an equiva-

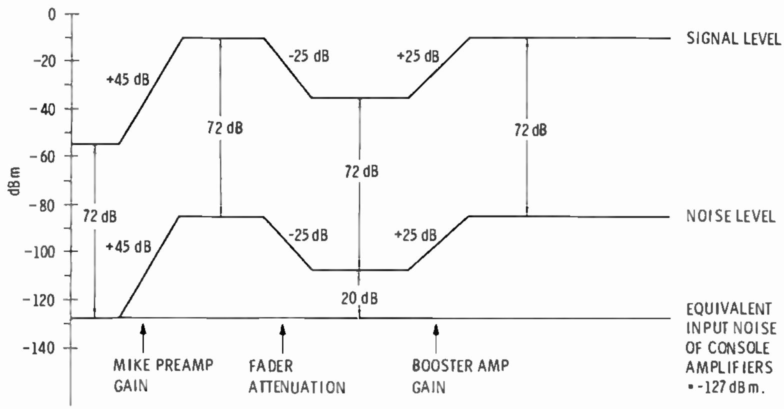


Fig. 3-23. Good console-operating procedures do not degrade the signal-to-noise ratio.

lent input noise in the mike preamp of −127 dBm, there is a signal-to-noise ratio of 72 dB. The mike preamp raises the level of both the signal and the noise by 45 dB, a fader attenuates them by 25 dB, and a booster amp raises them by 25 dB. The 72-dB ratio is maintained throughout. Noise added by the input stage of the booster amp is lower than the noise level at the preamp output and can be disregarded. If, on the other hand, 50 dB of attenuation was used after the mike preamp, the signal would be reduced in level to −60 dBm, while the noise level would fall to only −130.9 dBm (the minimum noise level possible in a circuit with frequency response from 20 Hz to 20 kHz) rather than to −132 dBm. This noise level is added to the −127-dBm e.i.n. of the next stage, producing a total noise level at its input of approximately −125.5 dBm. To obtain the original signal level of −10 dBm, the next amplifier must have 50 dB of gain.

This raises the noise level up to −75.5 dBm, resulting in a signal-to-noise ratio of only 65.5 dB (Fig. 3-24). Attenuating the input signal within 20 dB of the amount of the mike preamp gain lowers the signal-to-noise ratio. The fader attenuates the input signal by 50 dB to −60 dBm and tries to reduce the noise generated by the mike preamp to −132 dBm. The theoretical minimum noise level due to random electron motion in the audio frequency range is approximately −130.9 dBm, so the noise can only drop this far, increasing

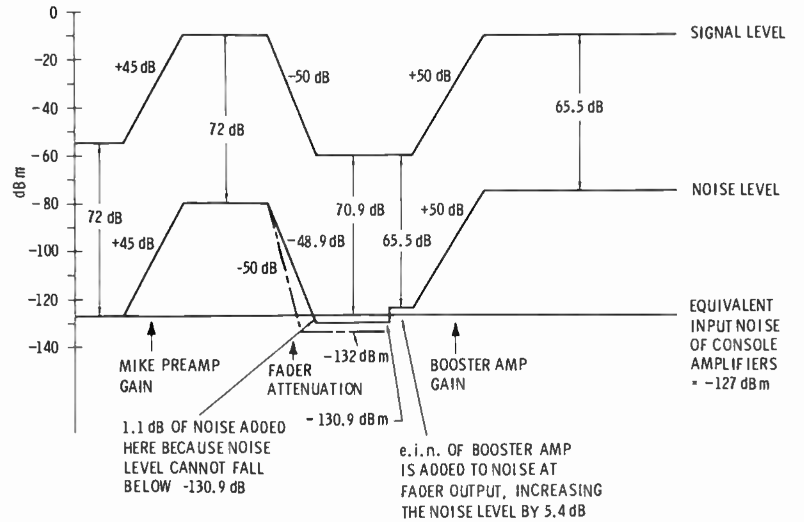


Fig. 3-24. Attenuating the input signal within 20 dB of the amount of the mike preamp gain lowers the signal-to-noise ratio.

the noise level by 1.1 dB over what it should be. The −127-dBm e.i.n. of the booster amp is added to the −130.9-dBm noise at the fader output, producing a −125.5-dBm noise level at the input of the booster amp, an additional 5.4 dB of noise. The ratio at the input of the booster amp is −60-dBm signal to −125.5-dBm noise, or 65.5 dB, a total noise increase of 6.5 dB. There has been 6.5 dB of signal-to-noise ratio lost because the signal was attenuated too much. By being aware of the gain and e.i.n. figures of the amplifiers in the console, the engineer can avoid overattenuating.

At times in a pop session it may be desirable to obtain a more distant sound than that picked up by close miking, so *distant miking* of an instrument is used to pick up some of the *room sound.* Placement of the distant mike at a random height often results in a hollow sound due to *phase cancellation* of the direct sound with the sound reflected from the floor (Fig. 3-25). The sound reflected from the floor travels farther than that which reaches the mike directly. Frequencies for which this extra path length is one-half of a wavelength or an odd integer multiple of one-half a wavelength, arrive 180° out of phase with the direct sound, producing dips in the frequency response of the signal at the mike output. Since the reflected sound is at a lower level than the direct sound due to its travelling farther and losing energy when it hit the floor, the cancellation is not complete.

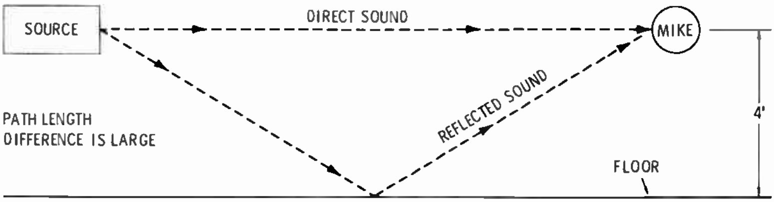


Fig. 3-25. A distant mike picks up both the direct sound from the source, and the reflected sound from nearby surfaces such as the floor.

Dips of 15 dB can be produced, however. Moving the mike closer to the floor reduces the path-length difference (Fig. 3-26) and raises the frequency of cancellation. In the example given by Anderson and Schulein [2], a 2.4-feet path length difference produces cancellations at odd multiples of 233 Hz (i.e., 699 Hz, 1165 Hz, 1631 Hz, 2097 Hz, etc.). Decreasing the path-length difference to 1.2 feet raises the cancellation frequencies to odd multiples of 466 Hz. If the mike was partially sunk into the floor, there would be no reflection, and, as a result, no cancellation. In practice, a mike height of 1/8 to 1/16 of an inch keeps the lowest cancellation frequency above 10 kHz.

Placing the mike close to the reflecting surface also increases its output by 6 dB. Experiments in the reference have shown that can-

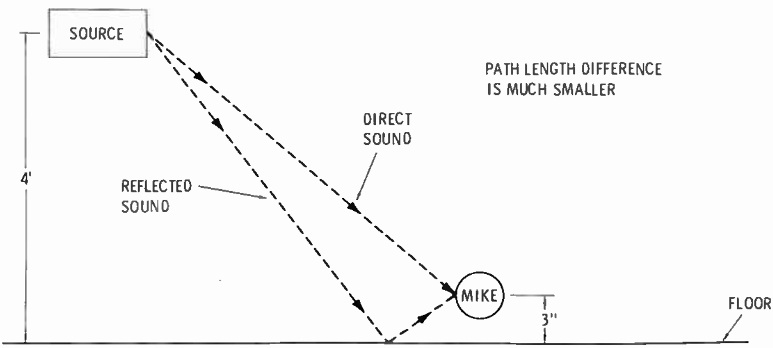


Fig. 3-26. Moving the distant mike toward the reflecting surface decreases the difference in path length and raises the lowest frequncy of cancellation.

cellations are a problem whenever mike-to-source distance is greater than one or two times the distance of the source to the reflecting surface. Surfaces such as a carpeted floor or the absorbent walls of a recording studio reduce the intensity of the reflected sound, thereby reducing the effect of the cancellations.[2]

PHASING

Stereo effects can be produced by using two microphones to pick up the same sound source, with one mike either located slightly farther away than the other or with each mike picking up the different timbres produced at different places on the instrument, such as over the bridge and over the hole of an acoustic guitar. Care must be taken in the use of two or more mikes on the same sound source to be sure that they are in phase with each other. If they are not, frequency cancellations will occur which reduce the volume and change the sound character. In the case of out-of-phase mikes placed left and right in a mix, a mono listener (over a car radio for example) will hear very little of the instrument, while a stereo listener will hear the desired balance. A simple check for phase is to assign the output from both (or all) of the mikes to one speaker and listen for drastic changes in the sound.

If one or more mikes is 180° out of phase with the others, the problem is usually in the miswiring of a mike cable (assuming the studio equipment was properly checked for phase integrity when wired up) and can be solved by either replacing the cable with a good one, using a phase reversal transformer, reversing the phase in the console (sometimes available through a switch on the input module) or if the mike is bidirectional, rotating the mike so that the rear of it picks up the signal. Other phase problems can be caused by using too many mikes near each other, where each picks up some of the sound picked up by the others. The different path lengths from source to mikes result in different phase relationships between the microphone outputs for the same sound. If these are combined at any point, cancellations occur producing severe dips in the frequency response of the signal. A rule of thumb to avoid phase cancellation in stereo miking is to make sure that the distance between either mike and the source is at least three times greater than the distance from mike to mike. Phase cancellation can also occur due to leakage of a sound source into the mike intended for another source if the mike outputs are combined. If the leakage amounts to less than 9 dB, the cancellation will be negligible. To check for this, have sound source 1 stop playing while sound source 2 continues. The level read on the VU meter connected to mike 1 should drop by at least 9 dB. If the distance between the two mikes is made at

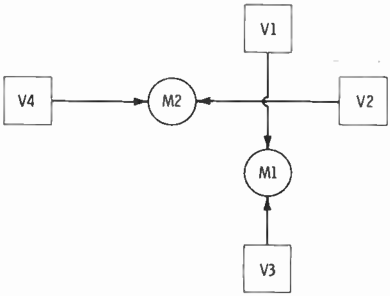


Fig. 3-27. Bidirectional mikes set at 90° angles to each other.

least three times the distance between each mike and its intended source, at least 9 dB of separation will always be achieved.

While cardioid and supercardioid mikes are most commonly used for multitrack recording to achieve separation between instruments, there are times when this goal can be better achieved by a bidirectional mike. While front-to-back discrimination in a cardioid mike may be better than 20 dB at lower frequencies, this may deteriorate to less than 15 dB above 3 or 4 kHz. In a bidirectional mike, however, sounds 90° off-axis are effectively cancelled for all frequencies, maintaining a better than 20-dB figure throughout the audio range. Thus a group of four vocalists can stand in a circle around two bidirectional mikes set at a 90° angle to each other, so that they can see and hear each other well, and the engineer will have control over the level of the two singing into each mike independent of the level of the two singing into the other mike (Fig. 3-27). As another example, the engineer might want to have separate control over the level of two cymbals which are on either side of a tom-tom in a drum set. By placing a bidirectional mike so that its front and rear face the two cymbals and its *dead* side aims at the tom-tom, he can effectively isolate the cymbals.

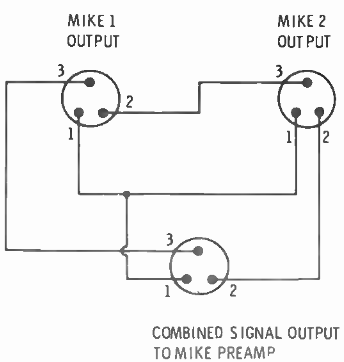
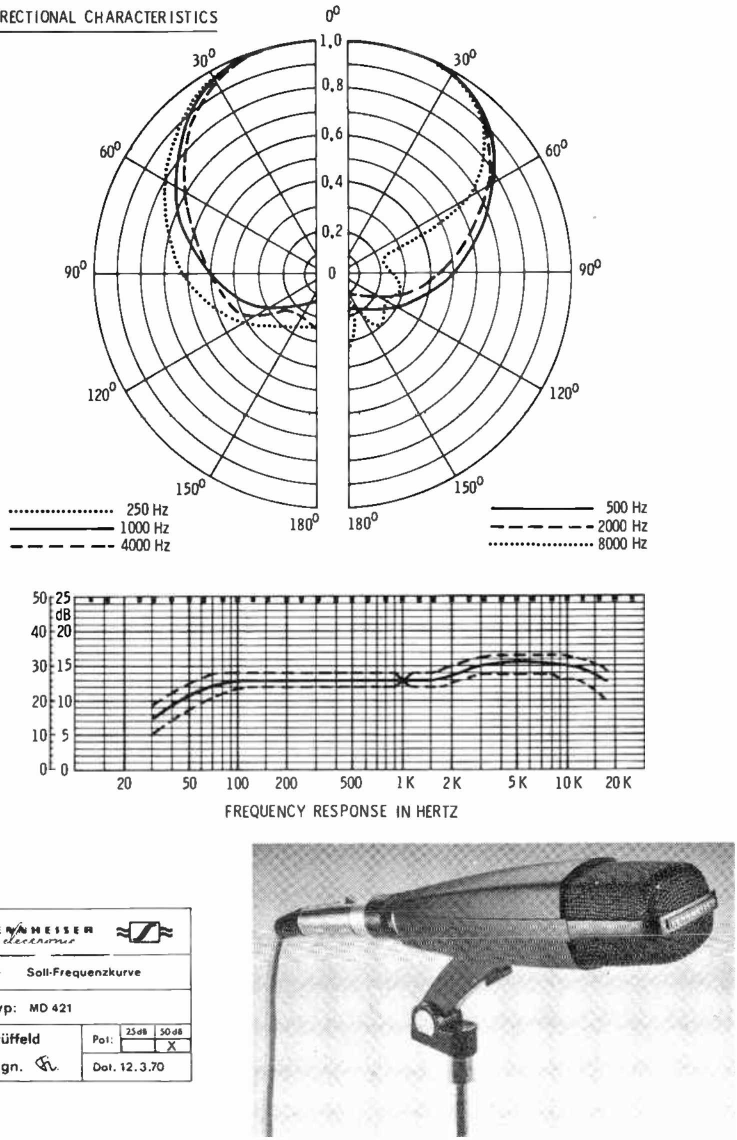


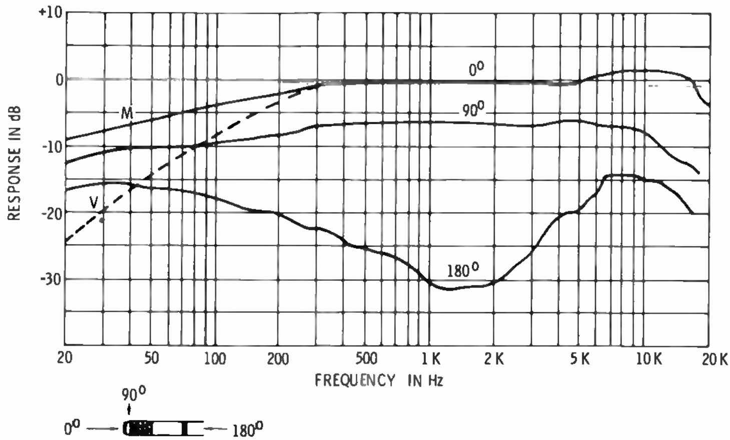
Fig. 3-28. Connecting two microphones in series. Pin 1 is ground, pin 2 is the high signal output, and pin 3 is the low signal output.

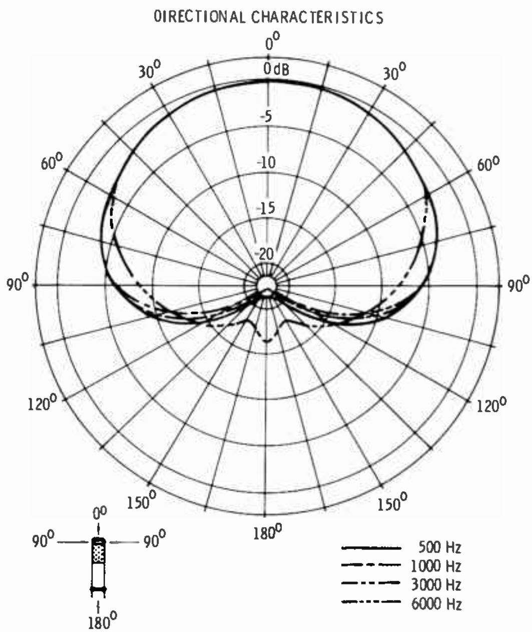


Courtesy Sennheiser Electronic Corp.

Fig. 3-29. The Sennheiser MD421 microphone.

In a large recording session, the engineer may need to use more microphones than the number of inputs provided on his console. This problem can be solved by connecting several microphones in series and treating them as one mike (Fig. 3-28). The input fader





Courtesy Superscope, Inc.

Fig. 3-30. The Sony ECM-22 electret condenser microphone.

will control the level of the mikes equally, so that the balance between them must be set by physically moving them in relation to the instruments. Series connection works best when miking a number of the same instruments, such as violins, using the same model microphone so that the tone from each instrument is the same and

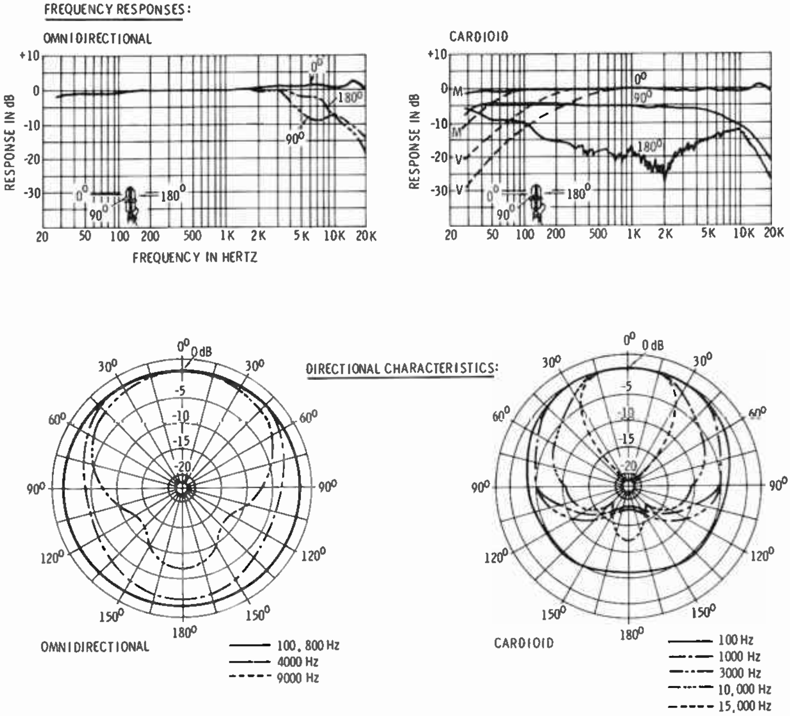
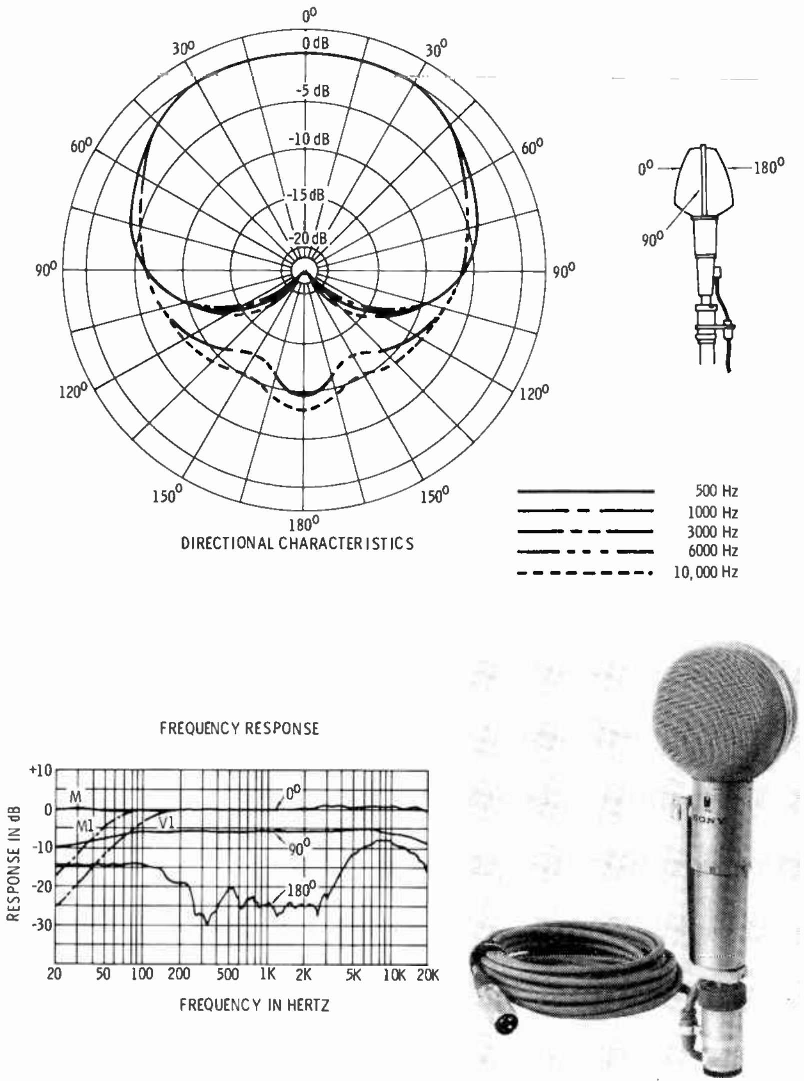


Fig. 3-31. The Sony C37P condenser microphone.

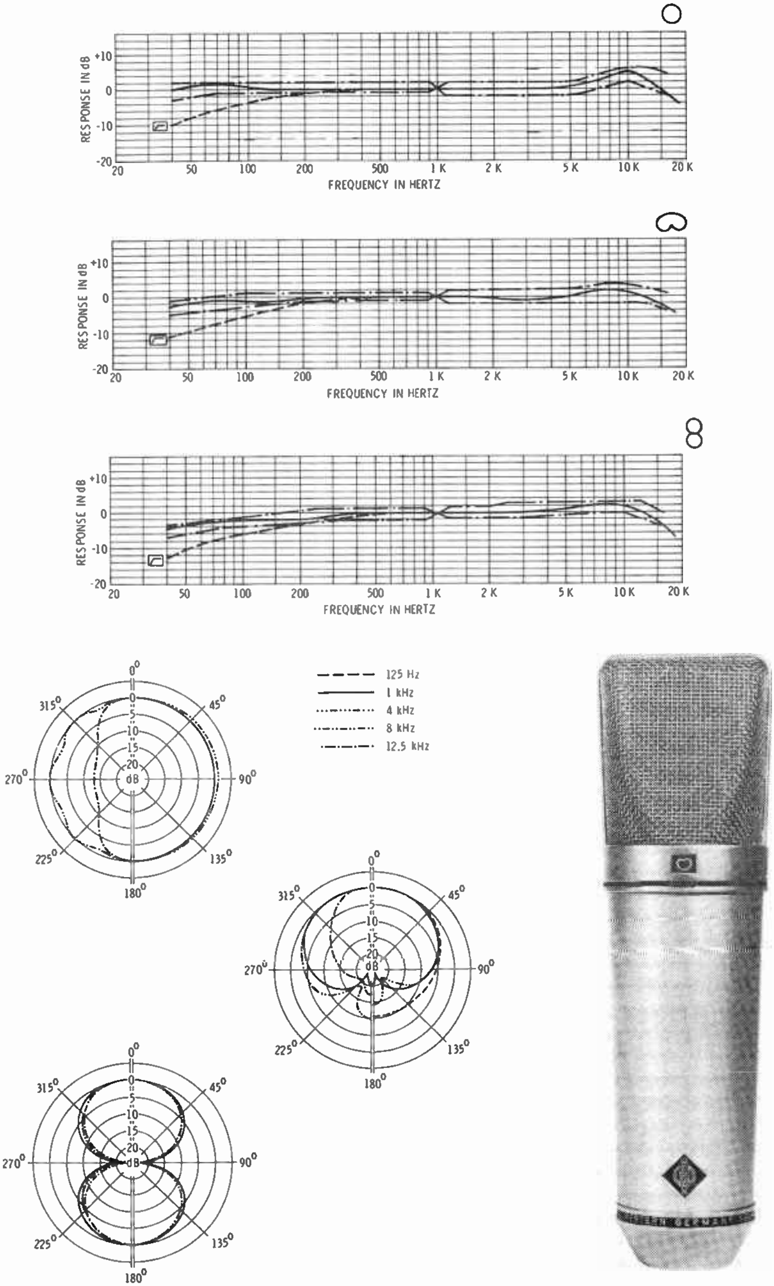
Courtesy Superscope, Inc.



Courtesy Superscope, Inc.

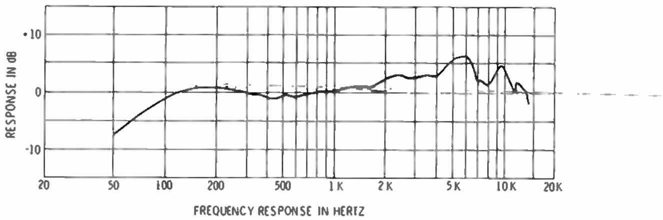
Fig. 3-32. The Sony C-500 condenser microphone.

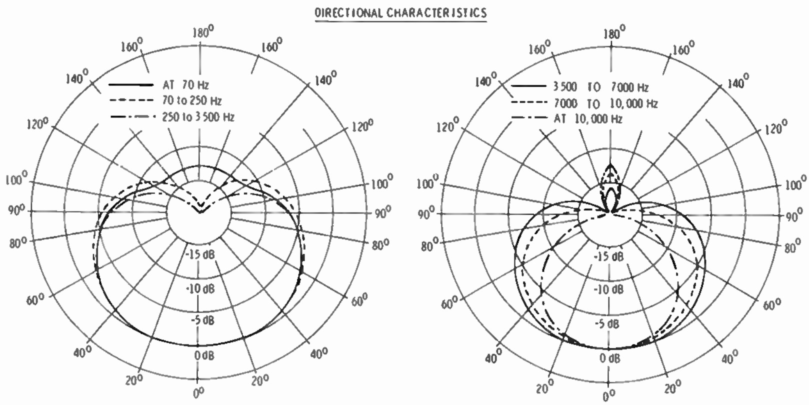
the inability to EQ the mikes separately is not a problem. The mikes must be in phase with one another to be used in series, or severe frequency cancellation will occur if their pickup patterns overlap. Mikes should not be connected in parallel because the low im-



Courtesy Gotham Audio Corp.

Fig. 3-33. The Neumann U87 condenser microphone.





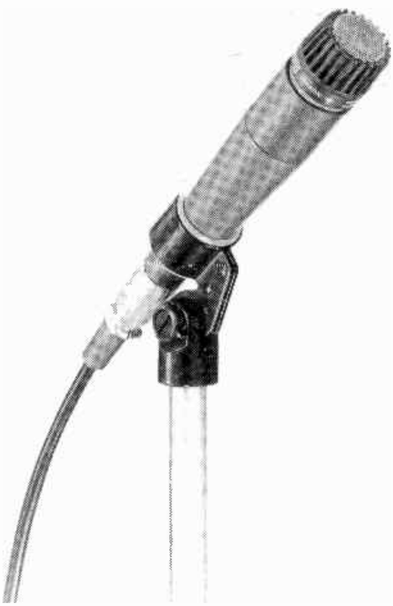
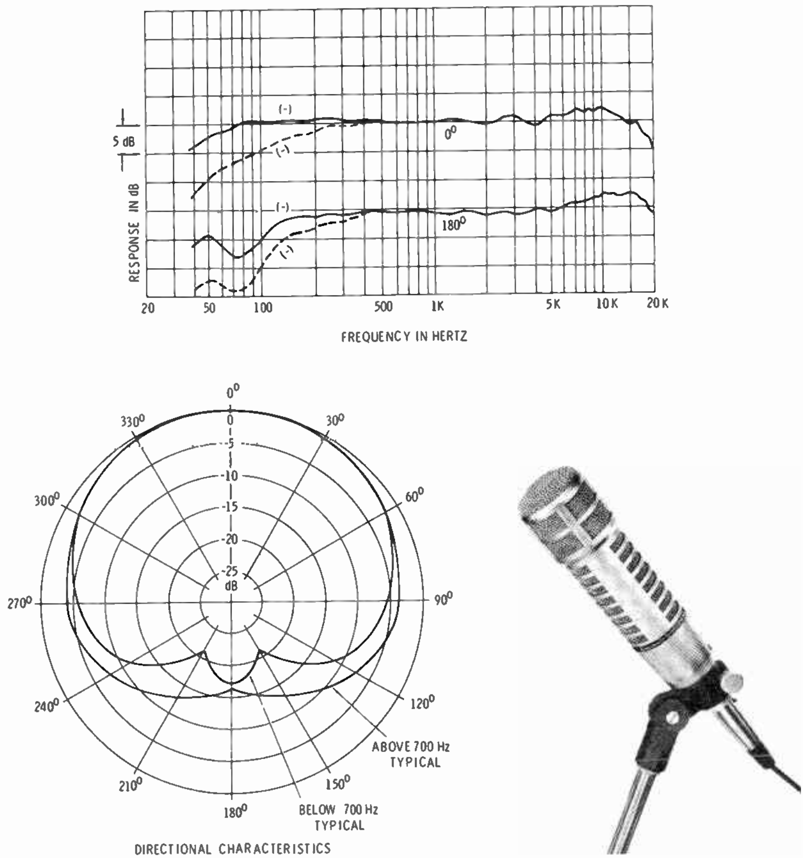


Fig. 3-34. The Shure SM57 dynamic cardioid microphone.

Courtesy Shure Brothers, Inc.



Courtesy Electro-Voice Inc.

Fig. 3-35. The Electro-Voice RE-20 dynamic cardioid microphone.

pedance of one of them would overload the output of the other, causing distortion, loss of level, and poor frequency response.

MICROPHONE CHOICE AND PLACEMENT

Charts 3-1 and 3-2 are presented as initial guidelines to mike choice and placement, but, as this is a matter of individual taste, experimentation and experience will suggest additional approaches. Figs. 3-29 through 3-36 show several of the microphones used in the recording studio and their response curves.

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Chart 3-1. Common Miking Techniques

DRUM SET

1. Two mikes hung over the drums, one on the left side and one on the right side to obtain a stereo effect.
2. Use a separate mike for each drum with the tom-tom mike high enough to pick up the cymbals and the snare mike high enough to pick up the hi hat.
3. Same as item 2 using a separate mike for the hi hat.

BASS DRUM

1. Remove the front head of the drum and place the mike inside the drum, off center. There are more overtones to the sides of the drum than in the center.
2. Place the mike off center at the rear (foot pedal) side being careful that any squeaks from the pedal are not picked up.

SNARE DRUM

1. Aim one mike at the top head.
2. Mike the top and bottom heads with separate mikes. The bottom mike gives the snare sound extra snap.

ACOUSTIC GUITAR

1. Place the mike behind the bridge.
2. Place the mike over the neck where it joins the body of the guitar.
3. Aim the mike into the hole of the guitar.

GRAND PIANO

1. Place the mike near the hammers at the center of the keyboard.
2. Place the mike at the center of the piano where the high and middle register strings are attached.
3. Place the mike at the rear of the piano where the bass strings are attached.
4. Hold the piano lid open using the long stick and place the mike at a distance from the piano aiming at the bottom of the lid.
5. Remove the lid from the piano and suspend the mike directly over the piano center.

UPRIGHT PIANO

1. Place a mike behind the sound board.
2. Place a mike over the open top of the piano.
3. Place a mike inside the piano.

HORNS

1. Place the mike close to the bell watching out for wind noise.
2. Place the mike two to three feet from the bell for a fuller sound.

VIOLIN AND VIOLA

1. Place the mike several feet above the instrument.
2. Place the mike close to the instrument aiming at the F-holes for a scratchier "fiddle" sound.

CELLO AND BASS FIDDLE

1. Place the mike over the bridge for a brighter sound.
2. Aim the mike into the F-hole for a fuller sound.

Chart 3-1. Common Miking Techniques—cont

ELECTRONIC AMPLIFIER

1. Aim the mike at the center of the speaker cone for a bright sound.
2. Mike the speaker off center for a fuller sound.
3. Place the mike on the floor at a distance from the amplifier for a fuller sound.

LESLIE SPEAKER

1. Aim a mike into the top louvres only.
2. Aim separate mikes into the top and bottom louvres.
3. Place the mike about six feet away at the height of the Leslie. The rotating speaker reduces phase cancellation due to reflection from the floor. Better sound is obtained by placing the mike on a side of the Leslie that has louvres rather than the back.

VOCALIST

1. Placing a mike one inch or less from the vocalist's mouth produces a "present" breathy sound.
2. Placing the mike between one and six inches from the vocalist reduces breath noise and crackling from the vocalist's mouth.

VOCAL CHORUS

1. Place a mike six to eight feet from the chorus.
2. Divide the chorus into several small groups and mike each group separately at a distance of one or two feet.

OBOE, CLARINET, AND FLUTE

Place a mike over the finger holes.

TYMPANI

Place a mike eight to twelve feet from the instrument.

PERCUSSION INSTRUMENTS

Place a mike two to six feet away.

ORCHESTRAL BELLS, XYLOPHONE, VIBES

Place a mike four to six feet above the keyboard.

2. Anderson, Roger, and Schulein, Robert. "A Distant Micing Technique." *db, The Sound Engineering Magazine*, Vol. 5, No. 4, April 1971, pp. 29-31.
3. Eargle, John. "How Capacitor Mics Produce Cardioid Patterns." *db, The Sound Engineering Magazine*, Vol. 5, No. 4, April 1971, pp. 32-34.
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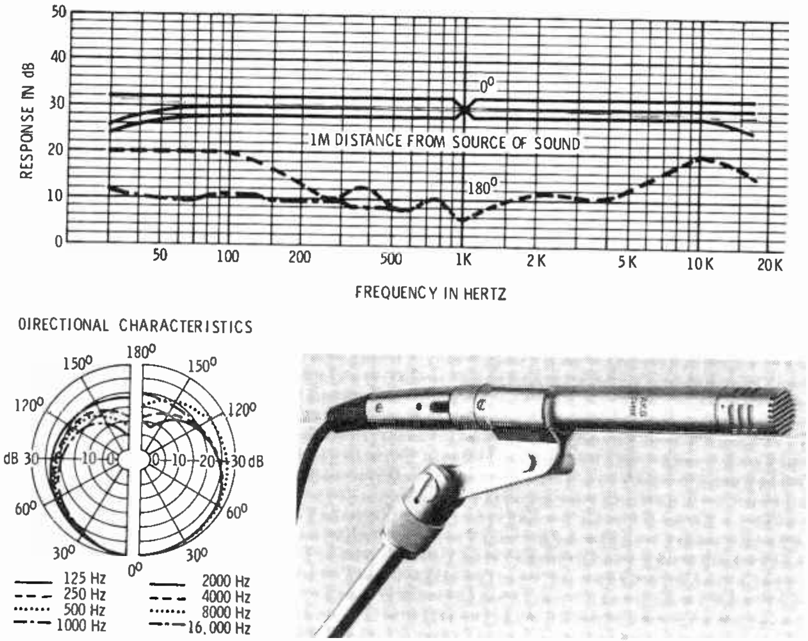


Courtesy North American Philips Corp.

Fig. 3-36. The AKG C451E condenser microphone.

Chart 3-2. Common Mikes Used for Instruments

PIANO: U87, KM84, ECM22, RE15, SM53.
BANJO, GUITAR, DOBRO: U87, KM84, ECM22, C451, SM53, C37.
BASS DRUM: U87, D24, 546, 666, 77-DX.
SNARE DRUM: C451, MD421, KM84, SM53, RE15.
OTHER DRUMS: MD421, SM53, U87, RE15, U67.
CYMBALS: U87, KM84, C451.
HORNS: U87, 77-DX, C37, SM53, U67, U47.
STRINGS: U87, C451, KM84, RE-20, D202.
LESLIES AND AMPLIFIERS: U87, RE15, SM53, 546, MD421, D202.
PERCUSSION: U87, RE15, SM53.
VOCALS: U87, U67, U47, SM53, RE-20, RE-16, D202.
BELLS AND CHIMES: U87, KM84, ECM22, C451.

The above microphones are made by the following manufacturers:

AKG: C451, D202, D24.
Electro-Voice: 666, RE-15, RE-16, RE-20.
Neumann: U47, U67, U87, KM84.
RCA: 77-DX.
Sennheiser: MD421.
Shure: SM53, 546.
Sony: C37, ECM22.

CHAPTER 4

Magnetic Tape Recording

All of the work in the recording studio makes use of the magnetic tape recorder. The ideal recorder would be a memory bank of perfect accuracy with an unlimited capacity for storing separate groups of information in a constant time relationship. The ideal recorder must be able to store the sounds heard in the monitor speakers so that they can be played back at a later date, exactly re-creating the sounds heard from the mikes. Magnetic tape is at present the accepted and the most practical means to approach this ideal. It has advantages over direct recording onto discs in that multitrack formats, editing, and overdubbing are possible.

The theory of tape recording is based on relating physical lengths of magnetic tape to periods of time (Fig. 4-1). By playing back each section of a length of tape at the same speed at which it was recorded, the original rhythm and duration of each sound and the spaces between sounds are preserved. The best way of ensuring that the time spectrum remains unchanged whenever the tape is played is to record and play it back at a constant speed.

PROFESSIONAL AUDIO RECORDERS

All professional audio recorders must perform the same functions and differ only in layout, specifications, and the labeling of their controls. The Ampex AG440 8-track transport is shown in Fig. 4-2. The transports for the 1-, 2-, and 4-track AG440s are identical to this with the exceptions that the reversing idler at the center of the transport is not present, different reel locks are used, and the tape guides are reversible to accommodate either ¼- or ½-inch tape. The features are labeled as follows: (A) Supply reel. (B) Take-up reel. (C) Reel lock. (D) Capstan. (E) Capstan idler. (F) Safety

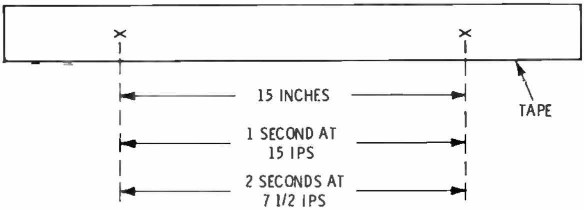
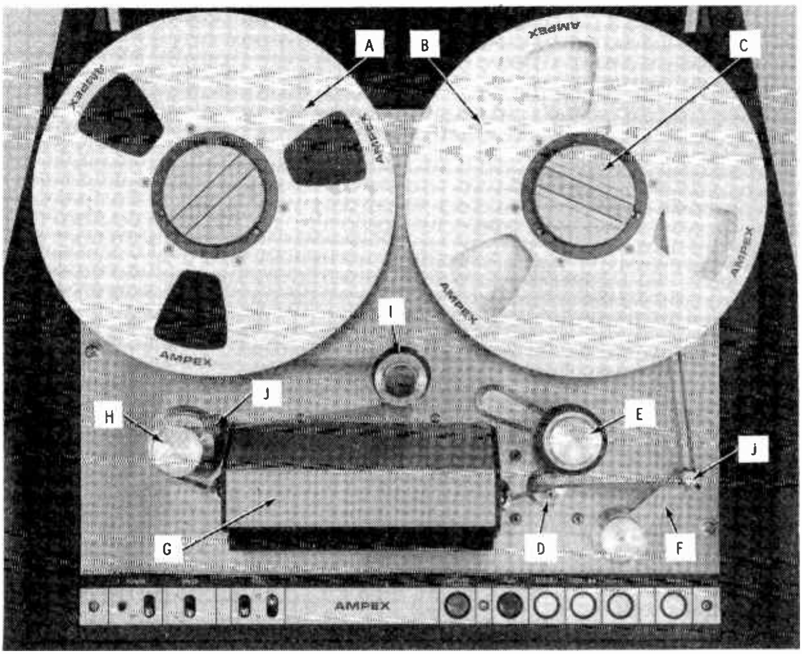


Fig. 4-1. Magnetic tape recording equates lengths of tape to periods of time.

switch. (G) Headblock. (H) Inertia idler. (I) Reversing idler (8-track version only). (J) Tape guide. Moving the tape at a constant speed is the main function of the *transport* which is actuated by pushing the *play button.* The *stop button* stops the motion of the tape by applying the brakes to the left and right turntables and by disengaging the constant-speed drive, so that the tape remains at the spot where it was stopped. The *fast forward* and *rewind buttons* engage the *tape lifters* to lift the tape off the heads and move the tape rapidly in either direction to find particular selections. The *record button* enables signals to be recorded on the tape once the constant-speed drive mechanism is engaged.



Courtesy Ampex Corp.

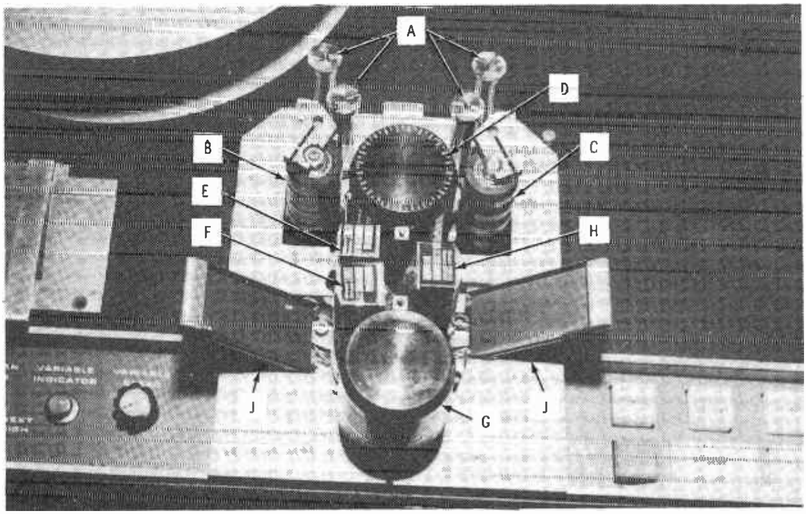
Fig. 4-2. The transport of an Ampex AG440 8-track recorder using one-inch wide tape.

The *edit* button (AG440 only) has three modes of operation: *stop-edit, play-edit,* and *fast-edit.* When the edit button is pushed while the transport is in the stop mode with the safety switch on, the left and right turntable brakes are partially released, and the safety switch is bypassed so that the tape may be moved easily from reel to reel by hand without the safety switch reapplying the brakes if it is allowed to fall. If the edit button is pressed while the transport is in the play mode, the takeup turntable motor is disengaged, and the safety switch is again bypassed. This enables the tape to be played off the transport into a wastebasket to remove unwanted material from a reel, while enabling the operator to hear what is being disposed of. Pressing the edit button in either of the fast forward or rewind modes drops the tape lifters for as long as the button is pressed, in order to allow the operator to hear the tape at fast speed to find the beginning and/or the end of selections.

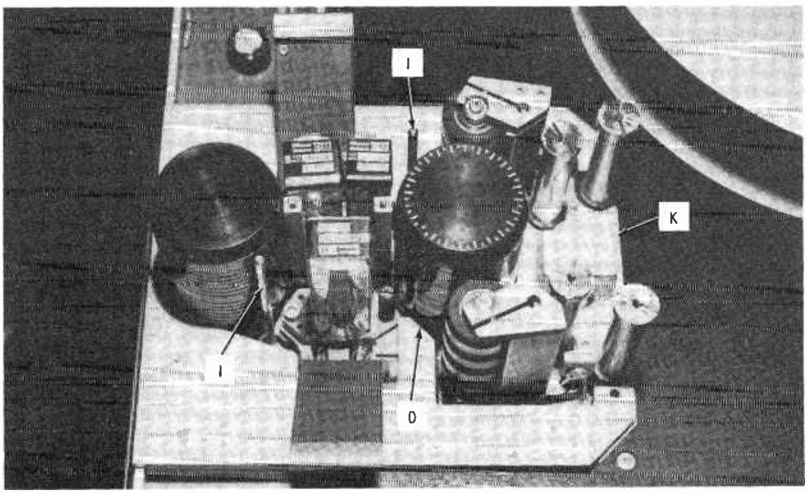
The *safety switch* initiates the stop mode if allowed to fall due to tape running out or breaking, except when defeated by the edit function. It also acts as a tape tension arm to take up the slack when tape motion is initiated.

The *capstan* and the *capstan idler* (often called the *puck* or *pinch roller*) work together to move the tape. The capstan is rotated by a motor which turns at a constant speed. When the play button is pushed, the capstan idler presses the tape against the capstan, and capstan rotation pulls the tape past the heads. The take-up turntable motor is slightly energized so that it will turn and *take up* the tape that is pulled to its side of the capstan. If this was not done, the tape would get tangled around the capstan or fall on the floor. At the same time, the supply turntable exerts a force called *holdback tension,* which acts against the pull provided by the capstan. This provides tension on the tape as it passes the heads to assure good tape-to-head contact. The *inertia idler* to the left of the *headblock* is attached to a heavy flywheel and helps eliminate variations in the tape speed due to friction caused by the tape scraping against the supply-reel flanges or being wound unevenly on the supply reel.

The transport system used by Ampex (also by Scully, Crown, Studer, and others) is called an *open-loop system* and requires holdback tension to maintain tape-to-head contact. Another system used by some manufacturers, such as 3M, is called a *closed-loop* or *differential capstan* drive system. Figs. 4-3A and B show the locations of the drive parts. (A) Tape guide. (B) Incoming capstan idler. (C) Outgoing capstan idler. (D) Capstan. (E) Erase head. (F) Record head. (G) Reversing idler. (H) Playback head. (I) Tape lifter. (J) Head shield. (K) Tape sensor lamp. In this system, the tape is pulled out of the head block faster than it is allowed to enter. The entrance and exit of the tape from the head block is controlled



(A) Front view.

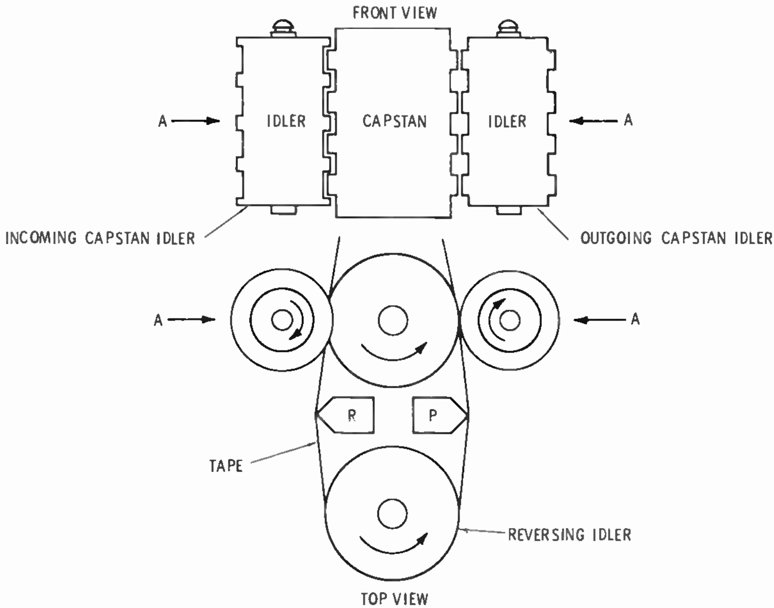


(B) Side view.

Courtesy Mincom Division, 3M Co.

Fig. 4-3. The closed-loop tape drive system used by 3M in their Mincom Division tape machines (dress covers removed).

either by two separate capstans with different diameters or by one grooved capstan with two pressure rollers which mate with opposite grooves in the capstan (Fig. 4-4) so as to provide the same effect as two capstans with different diameters. The phenomenon of removing tape from the head block faster than it is allowed to enter



Courtesy Mincom Division, 3M Co.

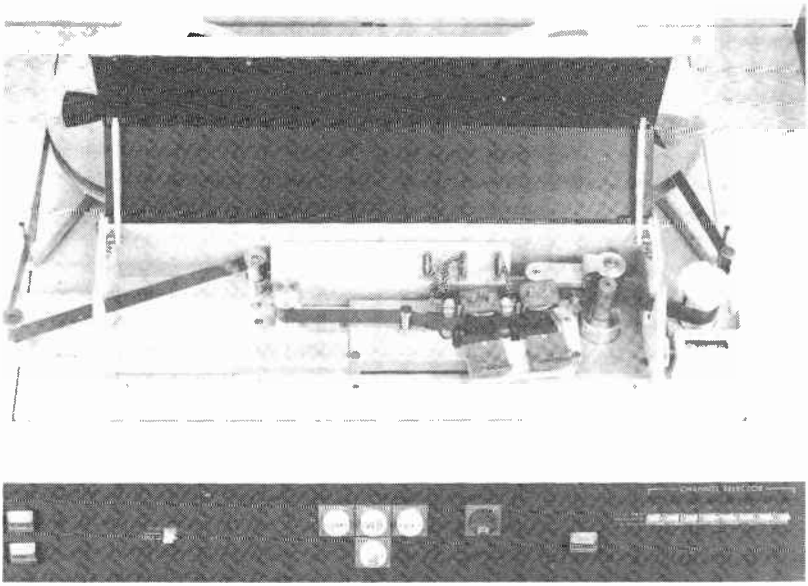
Fig. 4-4. Diagram of the capstan and capstan idlers used in 3M's "Isoloop" tape drive. The grooves make the incoming tape see a smaller capstan diameter than the outgoing tape.

creates the necessary tension of tape against the heads and is possible due to the ability of the tape to stretch by as much as 5% before permanent deformation takes place. The amount of stretch caused by closed-loop designs is well within this tolerance.

Advantages of the closed-loop design are:

1. Tape tension remains constant throughout the reel, whereas in the open-loop system without additional tape tension sensors, holdback tension (and as a result overall tape speed) changes from one end of the reel to the other as the diameter of the tape pack on the supply reel changes. This prohibits the splicing together of a piece of music recorded partially at the head of a reel and partially at the tail, for a pitch change would be noticeable at the splice. Newer machines, such as the Studer A80 and the Ampex MM1100, have tape tension-sensing arms which adjust the takeup and holdback tensions so that tape tension remains constant throughout the reel.
2. The dual-capstan/pinch roller design isolates the tape passing over the heads from the reels better than an inertia idler can.
3. The unsupported tape length in the closed-loop system is shorter, reducing *scrape flutter*.

Scrape flutter is the vibration of a piece of tape in the direction of tape length due to its passing across the head. It is very much like the effect of bowing a violin string except that the bow (head) is stationary and the string (tape) moves. The shorter the unsupported tape length, the higher the flutter frequency until it is above audibility. To reduce the scrape flutter effect, Ampex includes *scrape flutter idlers* (Fig. 4-5) on both sides of the record head to shorten the unsupported tape length to an acceptable value.



Courtesy Ampex Corp.

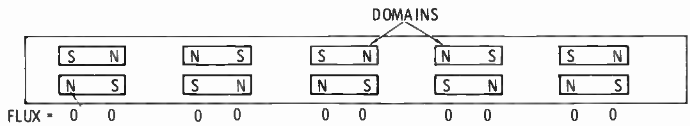
Fig. 4-5. An Ampex MM1000 8-channel recorder illustrating the tape path. One scrape flutter idler can be seen immediately to the right of the erase head, and the other immediately to the right of the record head. The narrow rods to the right of the scrape flutter idlers are the tape lifters.

Almost all professional recorders have three tape heads: the erase head, the record head, and the playback head. The *erase head* wipes out any signals that were previously recorded on the section of tape passing it so that a new signal can be recorded. This allows recording on the same piece of tape many times. The erase head is only energized in the record mode. The *record head* converts the electrical energy of the signal fed to the recorder input into magnetic energy which can be stored on magnetic tape. The *playback* or *reproduce* (*repro*) *head* converts the magnetic energy stored on the tape back into electrical energy which can be amplified and used to drive speakers.

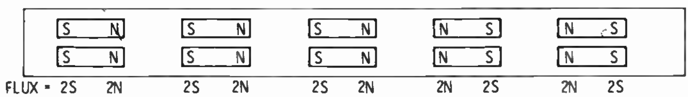
If magnetic tape was a linear medium, the function of recorder electronics would be simple. Unfortunately, there is a discrepancy between the amount of magnetism applied to a piece of tape and the amount of magnetism retained by the tape after the magnetizing force is removed. The magnetism retained is called *remanence* and the greater the remanence of one tape with respect to another, the higher its output will be for the same applied magnetic recording force.

MAGNETIC TAPE

Magnetic tape is composed of a *base material* such as *Mylar* (i.e., polyester) or *acetate*, covered with a magnetic oxide. The base is present merely as a means of holding the magnetic oxide and plays no direct role in the recording process. The molecules of the magnetic oxide form regions called *domains*. These domains are the smallest known permanent magnets. On an unmagnetized tape (Fig. 4-6A), the domains are oriented in random directions so that the



(A) Unmagnetized.



(B) Magnetized.

Fig. 4-6. Fields of domains on unmagnetized and magnetized tape.

north and south poles cancel each other out, leaving an average magnetic force of zero at any point on the tape. The individual magnetic fields are always present, however. When the tape is recorded, the individual domains are lined up in such a manner that their magnetism combines to produce a nonzero average magnetic force at the surface of the tape (Fig. 4-6B).

THE RECORD HEAD

Fig. 4-7 is a diagram of a record head. Current flowing through the coils of wire wrapped around the pole pieces creates a magnetic force which flows through the pole pieces to the *gap*. Magnetic force, like electricity, flows easier through some mediums than others. In electricity the amount of opposition to the flow of electric current is called *resistance*. The magnetic counterpart to current is *flux*, and the amount of opposition to the flow of flux is called *reluctance*. Since the magnetic oxide of the tape offers a lower reluctance path to the flux than the nonmagnetic material in the gap between the pole pieces, the flux flows from one pole piece through the tape to the other pole piece. The electrical signals fed to the record head are alternating currents and are therefore constantly

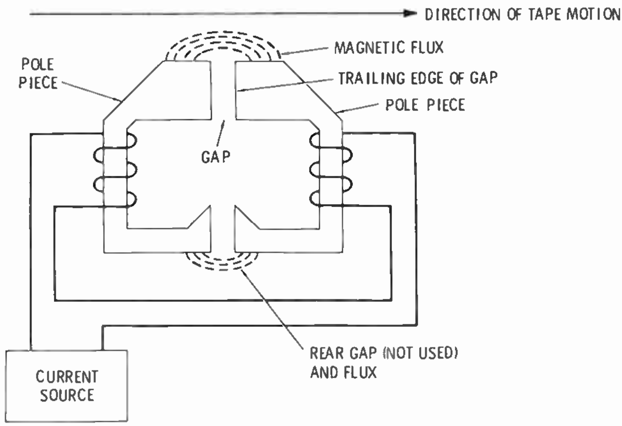


Fig. 4-7. A record head.

changing in amplitude. As a result, the flux produced by the head is constantly changing. Since the recording gap has a certain length (measured in the direction of tape travel), a small section of tape can be magnetized to several different polarities and intensities as it passes the gap. The tape retains the last magnetic polarity and intensity orientation it receives before it leaves the gap. For this reason, the actual recording is said to be done at the *trailing edge* of the record head, with respect to the motion of the tape.

THE PLAYBACK HEAD

The playback head (Fig. 4-8) is constructed similarly to the record head. When the magnetic flux on the tape passes through the gap, it induces a changing magnetic flux in the pole pieces. This flux cuts through the coils and induces a current in them, which can then be amplified. The flux flowing through the pole pieces of the playback head is a function of the average state of magnetization of the tape

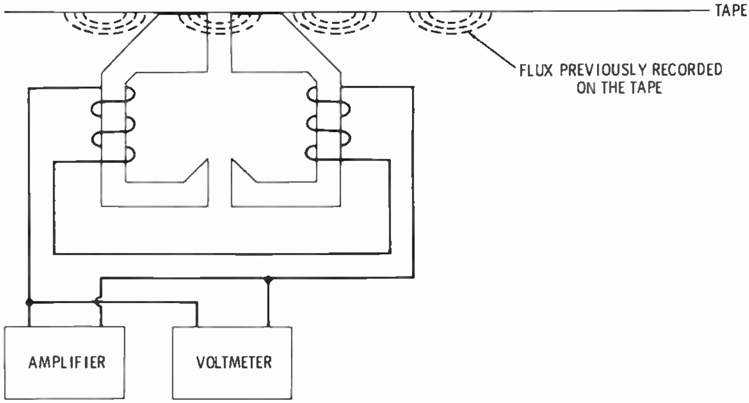


Fig. 4-8. A playback head.

spanning the head gap at any given time. The electrical output of the head is proportional not only to the average magnitude of the flux, but also to its rate of change. If the tape was magnetized in only one direction with the same average magnetism at each point, there would be flux in the pole pieces but no output from the head windings. The rate of change of the magnetism on the tape increases as a direct function of the frequency of the recorded signal, so the playback head output is dependent on the recorded frequency even though the tape is magnetized to the same degree for each frequency. The output voltage of the head is proportional to

$$\frac{\Delta\phi}{\Delta t} \qquad \text{(Eq. 4-1)}$$

where,

$\Delta\phi$ is a change in the average value of gap flux.
Δt is the time interval required for $\Delta\phi$.

Since the playback head output is directly proportional to the frequency recorded, the output of the head doubles for each doubling of the frequency on the tape (Fig. 4-9). In terms of dB, this means that the output of the head rises 6 dB (2 times the voltage where dB $= 20 \log V_1/V_2$) for each octave ($2 \times$ frequency of the recorded signal).

In addition to this change in output with frequency, the size of the playback head gap is a factor in determining the frequency response of the head. The head responds to the average value of magnetization in the gap, but as the frequency increases, more and more of a complete cycle falls inside the gap at any point in time. Since a sine wave has both positive and negative values, the true average magnetic flux in the gap decreases when more than half a cycle is

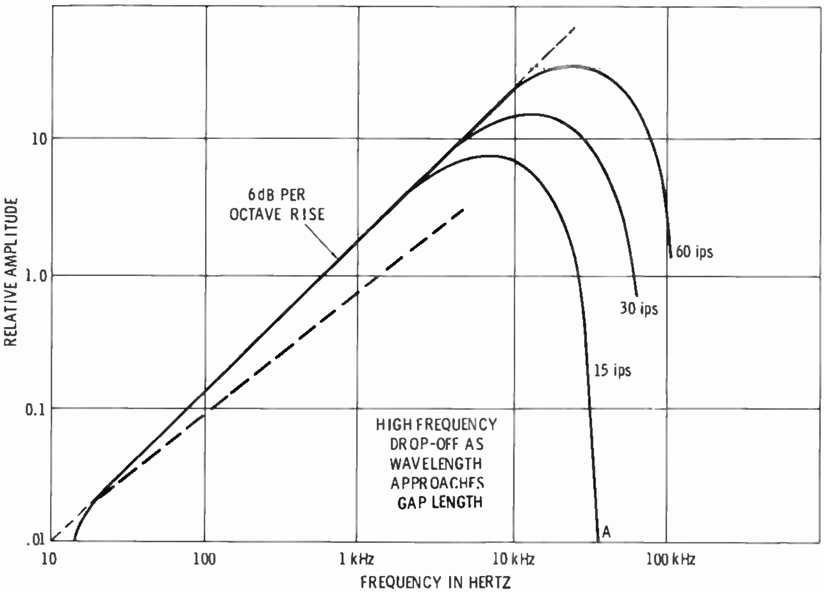


Fig. 4-9. The output of a playback head rises with frequency until the wavelength of the recorded signal approaches the gap length.

spanned. If the gap is spanned by exactly one cycle of a signal (i.e., a north magnetic field and a south magnetic field of equal intensity), the opposing fields cancel, and the average magnetization in the gap is zero, resulting in zero output from the head at that frequency (Fig. 4-10).

The upper frequency limit of the playback head is determined both by the length of the gap and by the speed of the tape. As the wavelength of a signal becomes less than twice the gap length, the output of the head decreases until there is no output at all when the wavelength equals the gap length. The wavelength of a signal on tape is equal to the speed of the tape divided by the frequency of the signal. For example, at 15 ips, one cycle of a 15-kHz tone takes up 0.001 inch of tape, while at 7½ ips it takes up 0.0005 inches. Thus, the faster the tape speed, the higher the upper frequency limit

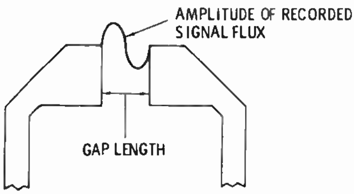


Fig. 4-10. The recorded wavelength is equal to the gap length, so the head produces no output.

of any playback head because the recorded wavelengths get longer at higher speeds. Similarly, the smaller the gap, the higher the frequencies that can be reproduced. Unfortunately, reducing the gap length to extend high-frequency response has the drawback that the average magnetism within the gap eventually decreases to the point where the output from the head is too low for a good signal-to-noise ratio. A compromise must be made between gap length, tape speed, and frequency response. Most present recorders use gaps between 0.00025 inch and 0.000038 inch long.

EQUALIZATION

To achieve flat frequency response with magnetic tape, *equalization* (EQ) is used both in the recording and playback electronics. Equalization is a term used to denote the changing of the relative

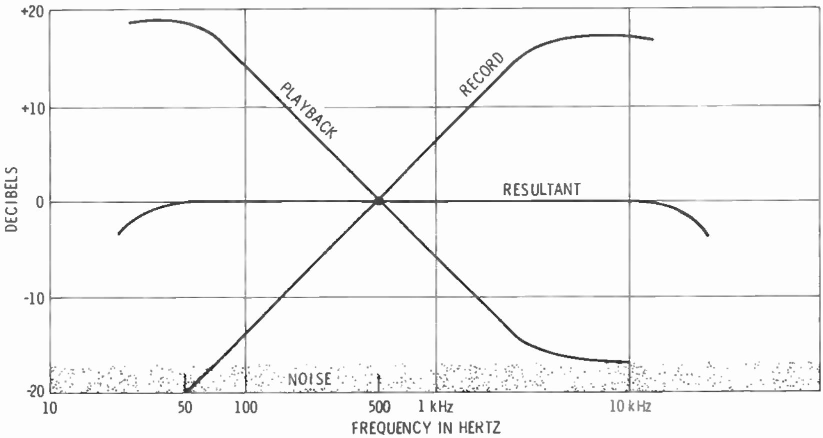


Fig. 4-11. The basic EQ used for magnetic recording.

amplitudes of different frequencies. The tone controls on a hi-fi system are an example of one type of EQ. Tape playback equalization boosts the low frequencies to compensate for the 6 dB per octave decrease of output as the reproduced frequency decreases (Fig. 4-11), while the record equalization boosts the high frequencies to compensate for the loss of some high-frequency energy by *self-erasure*.

Self-Erasure

Self-erasure occurs after the trailing edge of the record head and is the result of the formation of a secondary gap between the tape

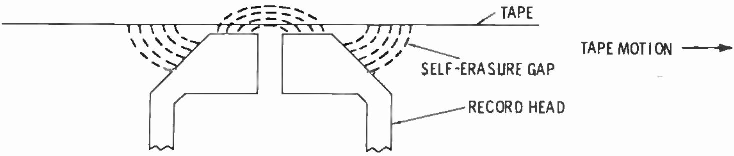


Fig. 4-12. High frequencies are partially erased by the flux induced in the self-erasure gap by high frequencies and by the bias current.

and the head as the tape leaves contact with the head. The flux generated in this gap by high-frequency signals and the bias current erases some of the recorded high frequencies the same way that an erase head erases a signal (Fig. 4-12). The bandwidth of the recording system is limited by the self-erasure of short wavelengths at the high end and by the signal-to-noise ratio at the low-frequency end. As the playback head output becomes very low, more amplification is necessary to bring it up to the same level as the higher frequencies, and, with enough amplification, the noise level of this playback amplifier becomes objectionable.

Bandwidth

The effective bandwidth for magnetic tape recording is about 10 octaves. The *bandwidth* refers to the frequency spectrum between the upper and lower *cutoff frequencies*. These are the frequencies on either side of the spectrum which are 3 dB lower than the center of the frequency spectrum; that is, they have ½ the power of the other frequencies (Fig. 4-13).

Bias Current

As mentioned before, the magnetization curve of magnetic oxide is not linear except between points A and B, and C and D as shown in Fig. 4-14A, thus a method of reducing the distortion is needed. The method used involves mixing a certain amount of high-fre-

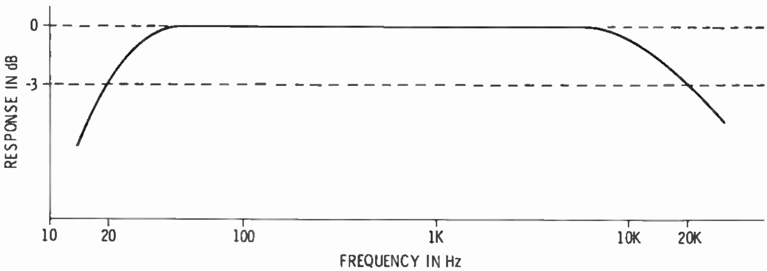
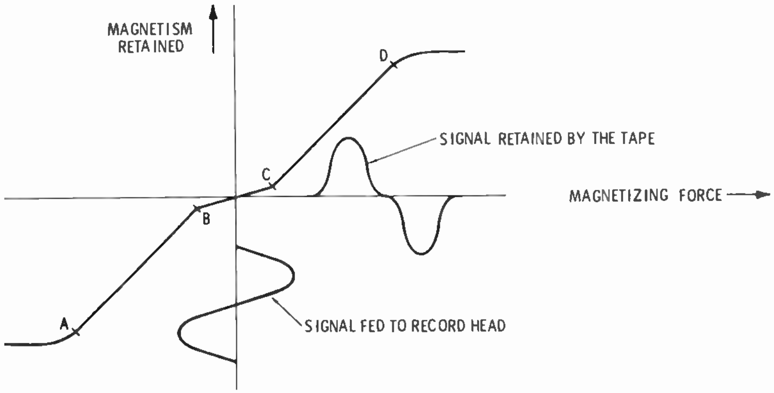
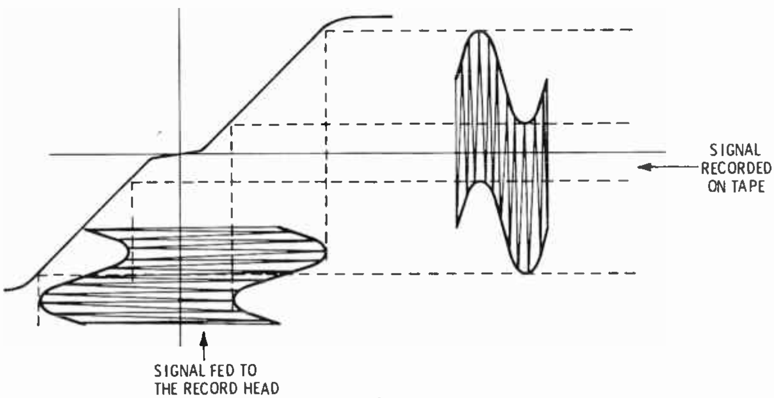


Fig. 4-13. A frequency-response curve with a 20-Hz lower cutoff frequency and a 20-kHz upper cutoff frequency.



(A) Magnetization curve.



(B) After bias.

Fig. 4-14. Illustrating effect of biasing on recording linearity.

quency current with the signal to be recorded. This current is called the *bias current* and must be several times the highest frequency to be recorded to prevent beats from being generated between it and harmonics of the input signal. In the Ampex recorders, the bias frequency is 150 kHz; in other brand machines the frequency is different.

The bias current is mixed linearly with the audio signal so that there is no modulation of either frequency by the other. The bias signal moves the audio signal away from the nonlinear zero crossover point of the magnetization curve onto the two more linear portions of the curve (Fig. 4-14B). The bias signal itself becomes distorted by the magnetic properties of the tape, but since its wavelength is much shorter than the length of the playback head gap, it produces no output from the head.

The amount of bias current used is very crucial and varies with individual record heads and different types of tape. The bias setting affects the sensitivity of tape both at high and low frequencies as well as affecting the overall frequency response, distortion, output level, and signal-to-noise ratio.

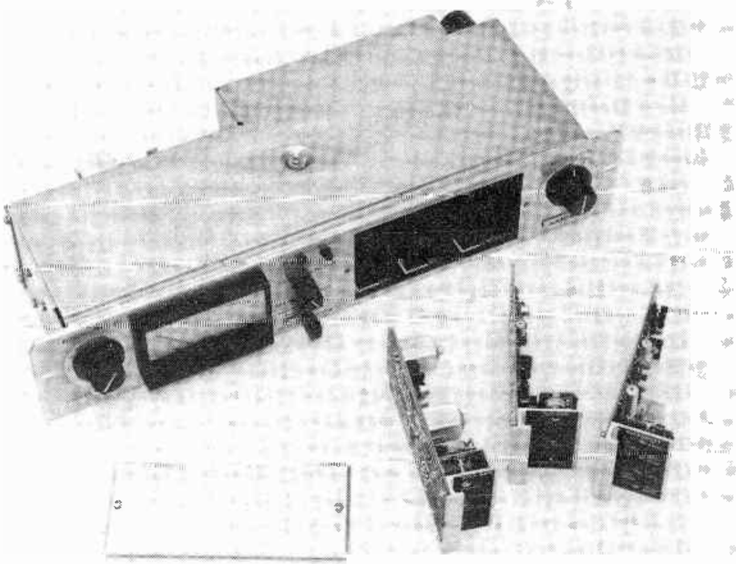
Erasure

The erase head is fed the bias current at a higher level than the record head so that the tape is saturated alternately in both the positive and negative directions. *Saturation* is the point where the tape is completely magnetized in one direction, so that additional increases in magnetizing force do not cause additional increases in the magnetism retained by the tape. This alternate saturation destroys any magnetic pattern that might previously have been on the tape. As the tape moves away from the erase head, the intensity of the magnetic field decreases so that the degree to which the tape is magnetized in each direction also decreases until the domains are left in a random orientation, and the average magnetism is zero.

RECORDING CHANNELS

Each channel of the recorder is identical to the others on that machine, and there are only a few minor changes between the electronics on the AG440 and the MM1000. The VU meter reads the approximate rms amplitude of either the signal being played back from the tape, the signal being applied to the channel input, or the bias current applied to that record head, depending on which position (*repro*, *input*, or *bias*, respectively) the OUTPUT SELECTOR switch is in (Fig. 4-15). In the repro position (Fig. 4-16), the signal played back from the tape is fed to the output of the machine and can be monitored. In either the input or bias position, the signal fed to the input of the machine is passed through the amplifiers and then fed to the output of the machine for monitoring. The reason for this arrangement is to allow the engineer to monitor what is being fed to the machine, both with the VU meter and by listening to the program, without having to record the program. This enables him to set the record level properly so that the signal is neither distorted by driving the tape into the saturation region, nor hidden by tape noise due to being recorded at too low a level.

By switching back and forth between repro and input while recording, the signal just recorded on the tape and played back by the playback head can be compared with the signal being applied to the record amplifier to make sure that the quality has not deteriorated due to any malfunction of the machine or the tape. Switching this control does not affect the recording. On newer MM1000s, the output



Courtesy Ampex Corp.

Fig. 4-15. One channel of electronics for an MM1000 or an AG440 with the plug-in bias, record, and reproduce amplifiers removed.

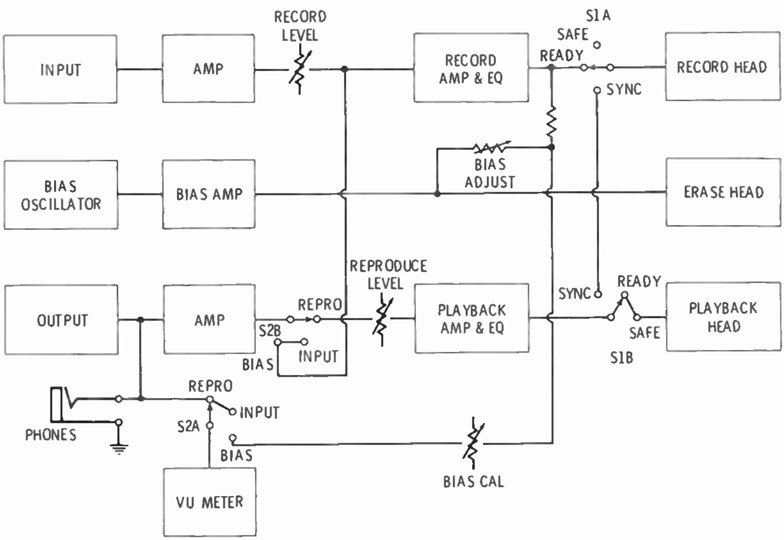


Fig. 4-16. A simplified block diagram of a single channel of AG440 electronics.

selector switch has been relabeled NORMAL, SETUP, and BIAS, with relays added to control output from each channel individually depending on whether that channel is recording or playing back. Table 4-1 illustrates the different modes of these switches and relays.

Table 4-1. MM1000 Monitor Switching

Master Play/Sync Switch on Play				
		Channel Output Signal in		
Channel Selector	Output Selector	Stop Mode	Play Mode	Record Mode
Record	Normal	Input	PB head	Input
Record	Set up	Input	Input	PB head
Nonrecord	Either	PB head	PB head	PB head
Sync	Either	Sync head	Sync head	Sync head

Master Play/Sync Switch on Sync				
		Channel Output Signal in		
Channel Selector	Output Selector	Stop Mode	Play Mode	Record Mode
Record	Normal	Input	Sync head	Input
Record	Set up	Input	Input	PB head
Nonrecord	Either	Sync head	Sync head	Sync head
Sync	Either	Sync head	Sync head	Sync head

The RECORD LEVEL control adjusts the intensity of the signal actually recorded on the tape, while the REPRODUCE LEVEL control adjusts the volume of the playback signal. The SAFE/READY switch prevents that channel from entering the record mode in either the *safe* or the *sel sync* position. In the *ready* position, the channel will go into the record mode (bias applied to the record head and erase head) when the master record button is pushed, assuming that the transport is already in the play mode. The necessity of pushing both the play and the record button to turn on the bias is a safety feature included to prevent accidental erasure of tracks. On the AG440, the master record button will initiate the record mode on all *ready* channels at any time after the play button has been pressed, while on the MM1000, the play and record buttons must be pressed simultaneously to initiate record even though the transport may already be in the play mode. The amber READY LIGHT is a warning light to indicate that that channel is in the READY TO RECORD position. In the sel sync position, the record head is connected to the reproduce amplifiers and, since it is similar in design to the repro head, can be used to play back the tape.

SELECTIVE SYNCHRONIZATION

The use of the sel sync (Ampex trade name for the sync or overdub position, short for selective synchronization) arises from a need to hear previously recorded tracks while simultaneously being able to record another signal in sync with them on the same piece of tape. This process is called *overdubbing*. If sel sync is not used and this is attempted, the previously recorded signal is heard played back by the repro head, while the new signal is recorded on the record head. The object of overdubbing is to be able to play these signals back together so that they sound like they were performed simultaneously. If sel sync is not used and the signals are now played back, the original track would be heard first with the overdubbed track lagging it by a time equal to the distance (d) between the record and playback heads, divided by the tape speed (Fig. 4-17).

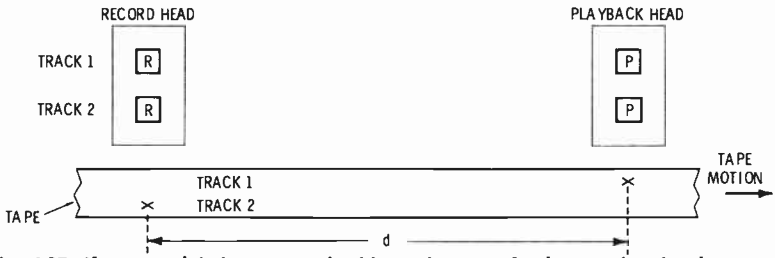


Fig. 4-17. If an overdub is attempted without the use of sel sync, the signal on track two will lag the original signal by a time equal to the distance (d), divided by the tape speed.

To prevent the overdubs from lagging the original tracks, the original channels are put into the sel sync mode so that they are played back through their record heads. When the new signal is recorded, it is recorded directly above or below the basic tracks, and when the entire tape is played back through the playback head, the overdub is in sync with the basic tracks.

On most recorders the frequency response of the signal played back in the sel sync mode is not as good as that played back through the repro head, especially in the low and high frequencies, because each head is optimized for its function. The signal is good enough, however, to be used to cue the overdubs. On the MM1000, an additional circuit has been added to improve the sel sync frequency response so that the difference in frequency response between the record and the reproduce heads is less noticeable. Other recorders, such as the Studer A80, have separate playback equalization for the sync position, and the Ampex MM1100 has record and playback heads of similar design so that sync response closely matches playback response using the same EQ.

On the back panel of the electronics is a SEL SYNC GAIN CONTROL which enables sel sync playback to be set to the same level at 700 Hz as normal playback so that no level changes need be made when switching from one to the other. Below this control is a SEL SYNC BIAS TRAP adjustment. This control is set for each channel so that the bias signal being recorded on a track, which leaks into an adjacent head used for sync playback, is removed before reaching the playback channel output by a tuned LC circuit (Fig. 4-18). The bias leakage would be inaudible, but it can overload amplifiers following the recorder and would cause a false indication on the VU meter of the channel in sync if not removed. These two back-panel controls seldom need adjustment once they are set unless the heads are replaced.

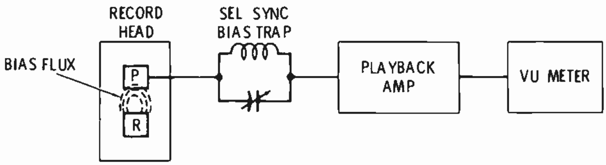


Fig. 4-18. The sel sync bias trap.

CALIBRATION

Since the sensitivity, output level, bias requirements, and frequency response of different brands of tape vary considerably, the record level, playback level, bias current, record EQ, and playback EQ, are all variable. The procedure used to set these controls is called *electronic alignment* or *calibration*. In most studios this is done the first thing in the morning for each machine. The alignment, however, is good only for one particular type of tape and one particular set of heads. If either the type of tape or the head block is changed, the machine must be realigned. As a check on the total performance of the recorder, however, it is good practice to check the alignment of the electronics on each machine to be used just before each session. In this way broken wires, defective components, and other obstacles to good recording are discovered before they ruin a good take. The alignment procedure seems complicated, but once learned it usually takes less than ten minutes to complete.

Alignment is done with reference to an NAB (National Association of Broadcasters) standard playback alignment tape. This tape is available from Ampex, Standard Tape Laboratories, and others, in ¼-inch, ½-inch, 1-inch, and 2-inch tape widths and is recorded *full track*. That is, the signal is recorded on the full width of the tape so that the signal is in phase at all of the gaps on an ideal multitrack playback head.

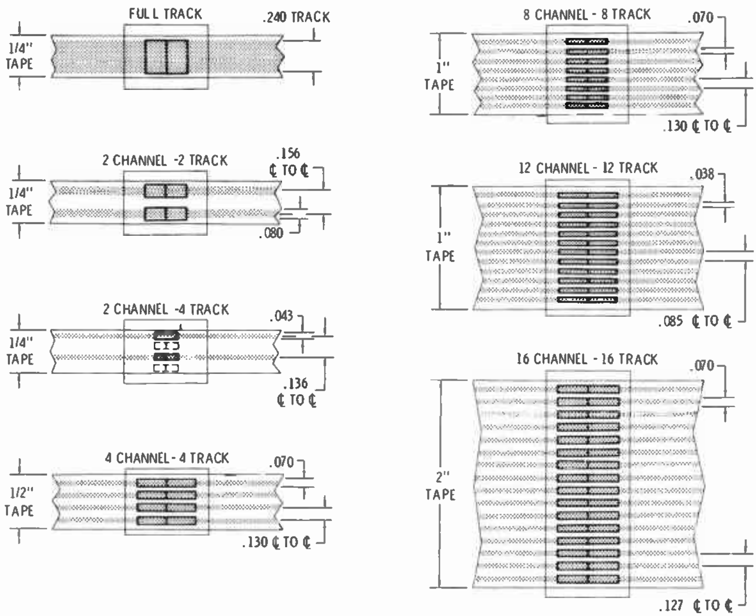


Fig. 4-19. Some track configurations available for magnetic tape recording.

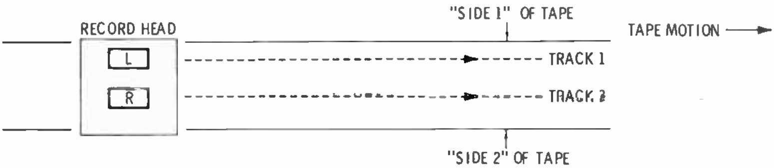
HEAD AND TRACK CONFIGURATIONS

Several head and track configurations are available for each tape width. For ¼-inch tape, the most common ones are full track (mono), two-channel, two tracks (often called *half track* or *half-track stereo*), and two-channel, four tracks (often called *quarter track* or *quarter-track stereo*) (Fig. 4-19).

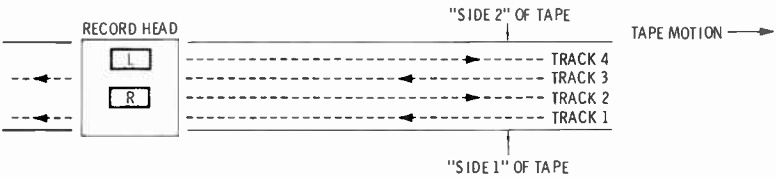
Full- and half-track heads are used for professional work because the greater recorded track widths produced by them enable the tape to retain more magnetism and produce a higher output, resulting in better signal-to-noise ratios. The wide recorded tracks are also less sensitive to dropouts than narrow tracks.

Quarter-track heads are used in consumer reel-to-reel tape recorders because they double the available stereo recording time, as compared to half-track stereo heads, saving the consumer the cost of a second reel of tape. Two tracks are recorded in one direction (Fig. 4-20A), the tape is flipped upside down, and recording is continued on the other two tracks with the tape moving in the opposite direction (Fig. 4-20B). If this were done with half-track stereo or full-track heads, the program recorded in the first pass of the tape would be erased during the second pass.

Since proper playback of the recorded information depends on using heads of the same configuration to record and play back the



(A) Recording tracks 1 and 3.



(B) Recording tracks 2 and 4.

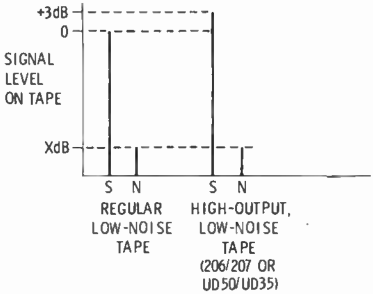
Fig. 4-20. The quarter-track stereo configuration.

tape, studios use quarter-track heads to produce copies of tapes for clients to listen to on consumer-type tape machines. The original full- or half-track tapes are used to produce the discs or tapes that are sold in the record stores.

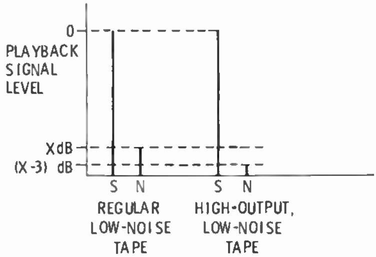
Guard bands of unrecorded tape are left between the tracks to prevent cross talk. The width of the bands is only slightly less than the recorded track width, as can be seen from Fig. 4-19. Track widths are also available for recording three and four channels on ½-inch tape, eight and twelve channels on 1-inch tape, and sixteen channels on 2-inch tape. Twenty-four track heads also use 2-inch tape by reducing the track and guard band widths of the sixteen-channel configuration to the figures used for recording twelve channels on 1-inch tape.

TAPE SPEEDS

Several different tape speeds are used in the studio. Most work is done at 15 ips because this speed allows all audio frequencies to be recorded at full level without saturating the tape and produces a good signal-to-noise ratio. In addition, this speed spreads out the recorded signals far enough apart for easy editing. Consumer hi-fi tape machines move tape at 7½ ips in order to save tape. As a result, most ¼-inch studio recorders operate at both 15 and 7½ ips so that copies can be made for clients at the proper speed. The 7½ ips produces too much of a compromise in signal quality for use in recording multitrack master tapes, so 30 ips is sometimes available as a second speed. Recording at 30 ips produces a signal-to-noise ratio increase of about 3 dB as compared to 15 ips, as well as increasing the resolution of the recording by doubling the length of tape corresponding to a particular time interval. This results in bet-



(A) Recorded signal level.



(B) Playback signal level.

Fig. 4-21. Comparison of high-output, low-noise tape with regular low-noise tape.

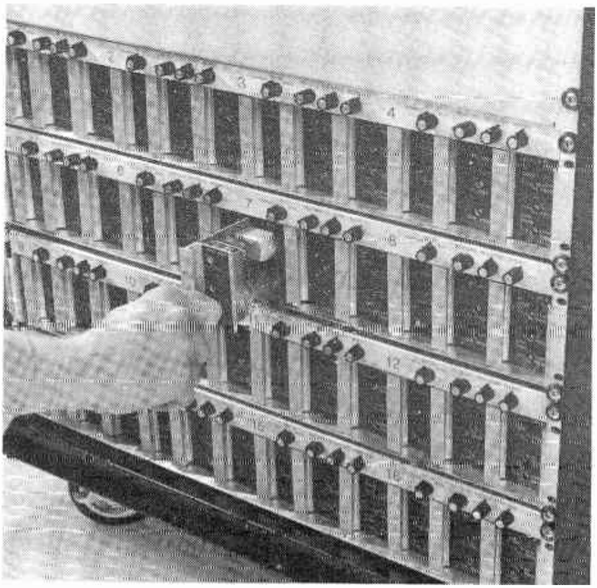
ter transient response. The 15-ips recording is more prevalent than the 30 ips because all professional studios have 15 ips capability, while not all of them have the 30-ips speed, limiting the interchangeability of 30-ips tapes. In addition, many people feel that the improvement in sound quality does not justify the doubled tape costs, especially when some studios offer noise reduction devices at no extra charge which are capable of improving the signal-to-noise ratio of 15-ips tapes by 10 to 30 dB.

Playback alignment tapes are available in several different speeds and track configurations. Each speed has a different set of test tones recorded at specific levels and at specific frequencies. The 15-ips Ampex tape begins with a 700-Hz tone recorded at *standard operating level*. All other tones are recorded in relation to this level so that when played back through a 15-ips NAB equalizer, all tones read the standard operating level. The NAB EQ compensates for the 6 dB per octave decrease in playback head output as frequency falls. The playback level control on each channel is set to produce the reference level at 700 Hz, which is 0 VU for use with regular tape and −3-dB VU for use with high-output, low-noise tapes such as Scotch 206/207 and Maxell UD50/UD35. The playback level control is set 3-dB lower when recording on these tapes because they can retain *3-dB more signal* (Fig. 4-21A) before saturation occurs than regular tapes can. Setting the repro level for −3-dB VU with the alignment tape requires that the record level be boosted 3 dB during the record adjustments to obtain a 0-VU output level. The more playback gain is used, the more tape noise is produced. Because the same output and distortion level are obtainable from these tapes with less playback gain, a better signal-to-noise ratio results (Fig. 4-21B).

After the 700-Hz tone are tones of 15, 12, 10, 7.5, 5, 2.5, 1 kHz, and 500, 250, 100, 50, and 30 Hz. The high-speed, high-frequency playback EQ is set to give the flattest high-frequency response (Fig.

4-22). This usually occurs when the high-frequency reproduce EQ is set so that the 10-kHz tone reads the reference level.

The low-frequency playback EQ cannot be set using the standard full-track alignment tape due to an effect called *fringing*. Fringing occurs when a tape of one configuration is played back with a gap narrower than that which recorded the signal on the tape (Fig. 4-23). The longer wavelength signals (those below about 500 Hz) which are above and below the gap are picked up by the head and added to those within the gap, producing an apparent rise in output at low frequencies. To avoid the fringing effect, the setting of the low-frequency playback EQ is postponed until after the record alignments are done.



Courtesy Ampex Corp.

Fig. 4-22. Close up of electronic adjustments for the Ampex MM1100.

The 7½-ips Ampex playback alignment tape is similar to the 15-ips tape except that it lacks the 30-Hz tone and that all tones are recorded at −10-dB VU except for the last tone, which is a 700-Hz tone recorded at operating level. The lower level is necessary at 7½ ips to prevent saturation of the tape at high frequencies. Naturally, the low-speed, high-frequency playback EQ control is used at 7½ ips.

The alignment tape is allowed to run out in the play mode so that it is wound under constant tension to prevent it from stretching in

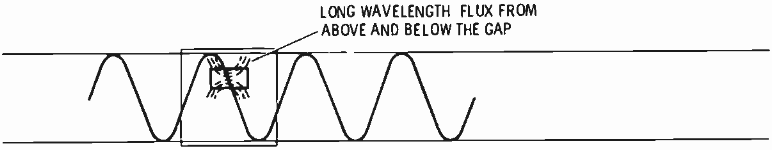


Fig. 4-23. A half-track mono head reporducing a tape recorded full track.

storage. To save time, it is stored *tails out*, i.e., the end of the tape is on the outside of the reel, and it is not rewound to *heads out* until just before it is used again. The tape may be wound back and forth in fast forward and rewind to find the frequencies desired, but it must be played at its proper speed from start to finish before storing.

Although a standard tape exists for setting reproduce levels and EQ for all NAB EQ tapes, the difference in sensitivities to high and low frequencies and the different bias requirements of different types of tape (even those made by the same manufacturer) require that record adjustments be made using the same type of tape to be used for the recording session.

The first step in the record alignment is the adjustment of the *erase adjust* control on the bias amp card. This control is used to tune the erase head circuit so that the bias current is a pure sine wave. If this control were not adjusted properly, the bias would be a distorted sine wave and would cause second-harmonic distortion of the recorded signal (frequencies which are twice that of the input signal would appear on the tape even though they are not present in the input signal). In addition, this bias distortion would cause the record head to become magnetized, erasing high frequencies and adding noise to the tape.

The erase peak adjustment is very stable and does not have to be made more than once a month unless the head block on the machine is changed. The erase adjustment can be made either with or without tape on the machine with the safety switch defeated by holding it up with a hub adapter or some other device. All channels are set in the record mode and all output selector switches in the bias position. The erase peak controls are then set for the highest readings on the individual VU meters. The reading will reach a peak, remain there as the control continues to be turned, and then fall back from the peak. The control should be set so that it is halfway between the point where the meter stopped rising and started falling. This is the proper setting for minimum bias distortion.

The next setting to be made is that of the amount of bias mixed in with the record signal through the BIAS ADJUST CONTROL. The type of tape to be used is threaded on the machine and a signal is fed in at operating level (0 VU on the console) at 1 kHz or 500 Hz, depending on whether you are setting for 15 ips or 7½ ips, respectively. The output selector is put in the reproduce position and the machine is set in the record mode. Since the amount of bias used affects the signal-to-noise ratio, distortion content, signal output, and frequency response of the recording, all these factors should be optimized when adjusting bias. As bias is increased to a certain point, all of these factors improve. As bias increases past this point, low-frequency sensitivity, signal-to-noise ratio, and distortion content figures continue to improve, but high-frequency sensitivity begins to fall, thus reducing output at high frequencies. As bias continues to increase, signal-to-noise ratio and distortion figures improve further, but both low- and high-frequency sensitivity decrease so that the overall output is lower. Continuing in this same direction, the signal-to-noise ratio eventually begins to deteriorate because of the great amount of amplification needed to recover the ever-smaller signal recorded on the tape. The best compromise occurs near the level of bias which gives peak output of the signals in the 500- to 1000-Hz range. A little high-frequency sensitivity is sacrificed in favor of better signal-to-noise ratio and distortion figures, for this loss of sensitivity can be overcome with the record pre-emphasis (EQ).

The bias level is set by turning the control clockwise (increasing the bias) until the output of the tape rises to a peak. Bias is then increased further until the output level falls 1 dB. This is called overbiasing the tape by 1 dB and with Scotch 206/207 produces the lowest tape modulation noise. Overbiasing also decreases the sensitivity of the tape to *dropouts* which will be discussed later in this chapter. With some types of tapes, however, trying to restore the lost high-frequency sensitivity by boosting the record pre-emphasis causes a hump in the frequency-response curve around 6 to 8 kHz. For these tapes, the bias can be set for a lower amount of overbias but never less than the amount which produced the peak in the 500- to 1000-Hz range.

The BIAS CAL control enables a reference level to be set for the bias in order to check it easily. After bias is set to the proper level, as described above, the output selector switch is set to *bias*, and the *bias cal* control is adjusted so that the meter reads 0 VU. Once adjusted, whenever the meter reads 0 VU in the bias position of the output selector switch, the bias is set properly. Since the proper bias control setting varies with the electrical characteristics of the heads, this setting is only good for one head block. This control is very stable and need only be checked once a month.

After setting the bias, the record level and pre-emphasis controls must be set so that all tapes recorded will play back according to the NAB standards. This assures interchangeability between studios. The gain and EQ of the playback head were previously set for flat frequency response when reproducing the NAB standard tape, so if the record circuits of the machine are now adjusted so that the signals recorded have flat frequency response when played back, the tape will produce flat frequency response on all NAB standard machines.

First, a 700-Hz tone is recorded on tape (the same type of tape for which we just set the bias) with the output selector switch in the repro position, so that what has been recorded on the tape and played back through NAB reproduce EQ can be monitored. The record level control is adjusted so that the recorded signal plays back at a level of 0 VU. Next, the output selector switch is set to the input position, and the record calibrate (REC CAL) control is set so that the meter again reads 0 VU. This equalizes the gain in the *input* and *repro* positions of the output selector so that the signal fed the machine and the playback of the signal being recorded on the tape can be compared at equal volumes to check its quality, and so that the engineer can see the level at which the signal will be recorded without having to actually record the signal.

The next adjustment is the record pre-emphasis control. This is set by feeding in a high-frequency tone, recording it on tape, and reading its playback level on the VU meter (output selector in the repro position). The control for the appropriate tape speed is set for the flattest high-frequency response. On most recorders this occurs when 10 kHz reads the same level as the 700-Hz tone. At 15 ips this level is 0 VU, but at 7½ ips the high-frequency record adjustments must be made at a record level 10 dB below 0 VU. This is due to the large amount of EQ necessary to record high frequencies properly at low speeds. If the level were not reduced, the EQ would boost the high-frequency signals to the point that the tape would saturate. Once the tape reaches saturation, additional increases of magnetizing force cause no increase in the signal recorded on the tape. Thus, a 10-kHz tone cannot be recorded at operating level at 7½ ips. This does not affect music reproduction very much because the high frequencies are usually a very small proportion of instrument output.

After the record EQ is set, the low-frequency playback EQ can be set. This level could not be set with the full-track alignment tape because of the fringing effects discussed before. If we record a low-frequency tone now, however, it can be played back with the same head configuration that was used to record it, eliminating the fringing effect. A 50-Hz tone is recorded at operating level, and the low-frequency playback EQ for the appropriate tape speed is set so that the tone reads 0 VU on the meter with the output selector switch set on repro. This will now complete the record alignment procedure.

A step-by-step procedure for different speeds and different tapes is given as Chart 4-1, while the NAB curves for record and playback EQ at different speeds are given in Fig. 4-24; this also shows the CCIR record and playback curves. These curves are the European tape equalization standard which is used instead of the NAB standard in many parts of the world.

Additional adjustments not available on Ampex audio recorders, but provided by 3M and Scully are noise balance and linearization. The NOISE BALANCE control passes a dc current through the record head to compensate for magnetism not removed during routine demagnetization. Any residual magnetism can cause noise and second-harmonic distortion. Since the magnetism is of fixed polarity, the polarity and amount of dc passing through the head can be adjusted to produce a magnetic field which exactly cancels the magnetism retained by the head.

This control is set by recording a 1-kHz tone, monitoring the playback head output with a wave analyzer set to measure the second harmonic, and adjusting the noise balance control for a minimum reading on the analyzer. If a wave analyzer is not available, the recorder can be set in the record mode with no input signal and

Chart 4-1. Playback and Record Alignment Procedures

PLAYBACK ALIGNMENT

Thread the playback alignment tape on the machine to be aligned. If the tape was stored tails out, rewind to the head.

A. ***15-ips playback alignment for low-noise, high-output tape, using a standard full-track 15-ips alignment tape.***
   1. **Set repro level for −3-dB VU at 700 Hz.**
   2. **Set high-frequency 15-ips playback EQ for −3-dB VU at 10 kHz.**
   3. **Do not adjust low-frequency playback EQ until after record adjustments.**

B. ***15-ips playback alignment for regular tape using a standard full-track 15-ips alignment tape.***
   1. **Follow steps A1 through A3, but adjust for 0-VU readings.**

C. ***7½-ips playback alignment for low-noise, high-output tape, using a standard full-track 7½-ips alignment tape.***
   1. **Set repro level control so that the 700-Hz tone recorded 10 dB below operating level reads 0 VU.**
   2. **Set 7½-ips high-frequency playback EQ so that the 10-kHz tone reads 0 VU.**
   3. **Do not adjust low-frequency EQ until after record adjustments.**
   4. **Set repro level control so that the 700-Hz tone recorded at operating level reads −3-dB VU.**

D. ***7½-ips playback alignment for regular tape, using a standard full-track 7½-ips alignment tape.***
   1. **Follow steps C1 through C3.**
   2. **Set repro level control so that the 700-Hz tone recorded at operating level reads 0 VU.**

RECORD ALIGNMENT

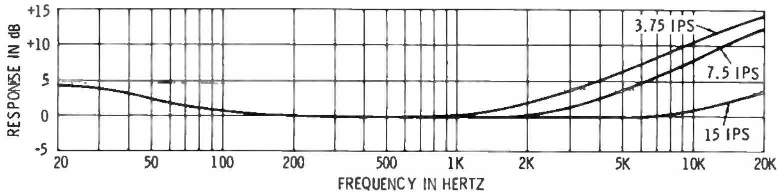
Thread the tape to be recorded on the machine.

A. *For all tapes at 15 ips.*

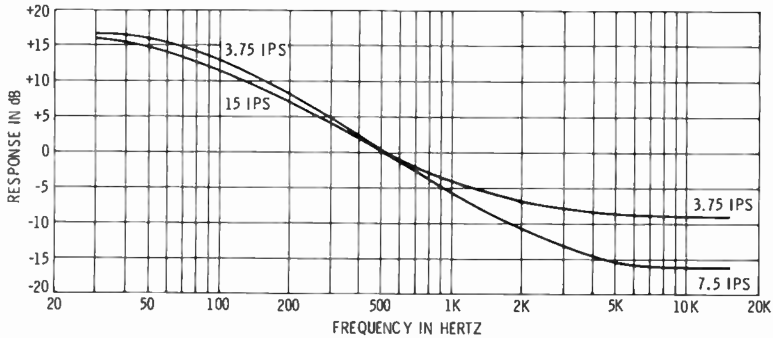
1. Set output selector to the bias position, and set the machine into the record mode on all tracks.
2. Adjust the erase peak control for maximum meter reading.
3. Feed a 1000-Hz tone into the machine inputs.
4. Set the output selector to repro, and beginning with a low bias setting, increase the amount of bias until the meter reading rises to a maximum. Continue increasing the bias level until the meter reading drops by 1 dB.
5. Feed a 700-Hz tone into all machine inputs at 0-VU level from the console, and set the record level control so that the meter on the recorder reads 0 VU.
6. Set the output selector to the input position and adjust the record cal control for a 0-VU meter reading.
7. Set the output selector to repro and feed a 10-kHz tone into all machine inputs at 0-VU level from the console, and set the 15-ips record EQ so that the meter reads 0 VU (at the peak of the meter swing if the needle is unsteady).
8. Feed a 50-Hz tone into the machine inputs at 0-VU level. Adjust the 15-ips low-frequency playback EQ for a 0-VU meter reading.

B. *For all tapes at 7½ ips.*

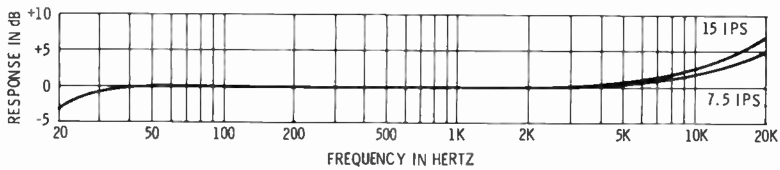
1. Follow steps A1 and A2 above.
2. Set the output selector to repro and feed a 500-Hz tone to all the machine inputs. Set all tracks into the record mode and increase bias level to obtain maximum meter reading. Increase bias so that meter reading drops slightly below the peak reading.
3. Feed a 700-Hz tone at 0 VU to all machine inputs and set the record level control for a 0-VU reading on the meter.
4. Set the output selector to input and set the record cal control for a 0-VU reading on the meter.
5. Reduce the setting of the record level control so that the meter reads −10-dB VU.
6. Set the output selector to repro and adjust the repro level control for a 0-VU reading on the meter.
7. Feed a 10-kHz tone into the machine inputs at 0 VU and adjust the 7½-ips record EQ to obtain a meter reading of 0 VU.
8. Feed a 50-Hz tone into the machine inputs at 0 VU and adjust the 7½-ips low-frequency playback EQ for a 0-VU meter reading.
9. Feed a 700-Hz tone into the machine inputs at 0 VU, set the output selector switch to input and adjust the record level control for a reading of 0 VU on the meter.
10. Reduce the repro level control to approximately the position set with the operating level tone on the playback alignment tape, then switch the output selector to the repro position and adjust the repro level control for a 0-VU meter reading. (Follow step 10 in the order stated to prevent pinning and possibly damaging the VU meters on the tape machine.)



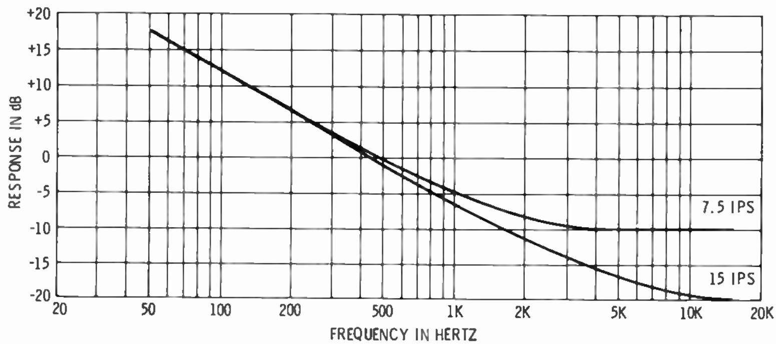
(A) Typical recording pre-equalization for ¼-inch tape recorders using NAB characteristics.



(B) Typical post-equalization for ¼-inch tape recorders using NAB characteristics.



(C) Typical pre-equalization characteristics for ¼-inch tape recorders running 7.5- and 15-ips, using CCIR (DIN) standard.



(D) Typical post-equalization curves for ¼-inch recorders using CCIR characteristics at 7.5 and 15 ips.

Fig. 4-24. Pre- and post-equalization for the NAB and CCIR characteristics.

adjusted for minimum snapping and crackling noise when listening to the playback head output.

Linearization

A *linearizer* is a circuit which compensates for the nonlinearity of magnetic tape. Tape tends to produce a certain type of distortion (mostly third harmonic) when overdriven before reaching saturation. The linearizer reduces this distortion by adding a small amount of third-harmonic distortion to high-level signals that are out of phase with the distortion generated by the tape. This cancels out the distortion that would otherwise occur on signal peaks.

Adjustment of this control entails recording a 1-kHz tone on the particular type of tape to be used and increasing the record level until 3% third-harmonic distortion is produced on the tape with the linearizer switched off. The linearizer is then turned on and adjusted for minimum third harmonic. Distortion should be reduced to below 1%. This adjustment must be rechecked whenever a different type of tape is used.

Print-Through

After the recording is made, the engineer must take steps to prevent the signals stored on the tape from being altered inadvertently. One type of deterioration which can occur is called *print-through*. Print-through is the unwanted transfer of a signal from one layer of tape to another by magnetic induction. Its effect is more pronounced when excessively high levels are recorded and is increased when the tape is stored in or exposed to a magnetic field. The transfer from layer to layer decreases by about 2 dB for each 1-dB decrease in the record level but increases with the time of storage and increased storage temperature. The amount of print-through depends on the physical separation of the section carrying the printing signal from the tape to be printed on, so tapes with thinner backings and with high-output oxide coatings tend to print-through more.

Magnetization components are recorded both in the direction of tape length, and perpendicular to its surface. According to E. D. Daniel [2], these two components print onto the adjacent layers of tape where they create external fields which are in phase at the surface of the outer layer and out of phase at the surface of the inner layer. In playback, the fields add for the outer layer and partially cancel for the inner layer (the fields are of unequal strengths), producing less audible print-through for the inner layer than for the outer (Fig. 4-25).

If the tape is stored heads out, the print-through will be heard before the main signal occurs. If the tape is stored tails out, the louder print will come after the signal occurs and will be masked somewhat by the desired signal. Print-through is not always permanent. By storing tapes tails out and rewinding just before playback, the printed signal intensity is reduced, and there is usually not enough time for a new signal to be printed before the tape is played back. Storing tape outside the temperature range of 60 to 80 °F or outside the range of 40 to 60% relative humidity can shorten the life

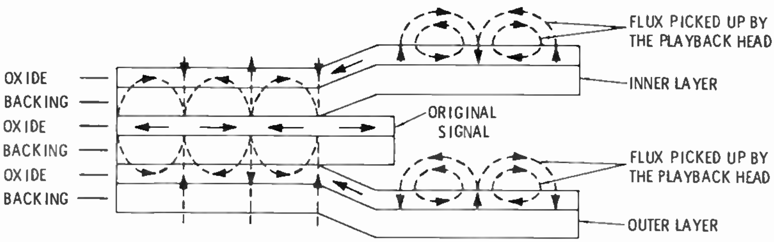


Fig. 4-25. The difference between print-through to the inner and outer layers of tape.

of both virgin and recorded tapes due to stretching and contracting of the base material.

Degaussing

Tape recorders must be clean both physically and magnetically to give optimum performance. If parts of the machine over which the tape passes are magnetized, erasure of the high frequencies can occur. If the record head is magnetized, the signal on the tape will have second-harmonic distortion.

Magnetism can be removed through the use of a *degausser* or *demagnetizer*. The degausser acts in the same way that the erase head of a recorder works, except that it operates at the power-line frequency of 50 or 60 Hz instead of above 100 kHz. This lower frequency is effective only if the degausser is moved past the magnetized object very slowly. The degausser works by subjecting the magnetized object to a magnetic field of sufficient strength to overcome that already present in the object. Once that occurs, the alternating field causes the domains of the object to be alternately magnetized one way and then the other. If the magnetic field is then slowly decreased, the domains are left in a random pattern such that the overall magnetism is zero. This is accomplished by moving the degausser away from the object at a speed no greater than one or two inches per second. The degausser must not be energized or deenergized within 3 feet of the tape recorder, or flux surges may create magnetic charges larger than that which the degausser can remove. Each object in the tape path must be slowly approached, saturated magnetically, and slowly moved away from one-by-one for proper demagnetization.

A *magnetometer* or *gaussmeter* is a device which measures the magnetic charge on an object. It is useful for indicating when an object is charged and needs degaussing, but because it is sensitive to the average of the magnetic charges over a rather large area, the object being measured may be charged even though it causes no reading on the meter. For example, the two pole pieces of a playback head could be magnetically charged, one north and one south. Since the sensitive part of the magnetometer covers both pole pieces at once, the fields cancel, and it gives no reading. The tape passing over the playback head, however, would receive the charges individually and possibly be partially erased. For this reason, degaussing is done every morning.

Cleanliness

The tape recorder also accumulates dirt due to *oxide shed.* This is oxide which falls off the magnetic tape and can accumulate on the heads, guides, and pinch roller. If it is allowed to build up on the heads, it can cause *dropouts,* which are momentary drops in the amplitude of a signal. This is due to the *separation loss* which occurs when the tape moves away from the *intimate* or close contact with the tape heads provided by the holdback tension of the supply reel. This loss can be computed as follows: separation loss (in dB) equals 55 times the separation distance, divided by the wavelength of the signal being recorded or reproduced. A separation distance equal to the thickness of a piece of cellophane from a cigarette pack (0.001 inch) can cause a dropout of 55 dB at 15 kHz (the wavelength of a 15-kHz signal recorded at 15 ips is 0.001 inch). Thus, any dirt which pushes the tape away from the head can cause severe problems.

Similarly, if dirt is allowed to accumulate on guides or on the pressure roller where it might be transferred to the tape and push the tape away from the head, dropouts can occur. The machines should always be inspected for cleanliness before recording or playing back tapes. Any dirt, oxide shed, or pieces of tape resulting from editing or tape breakage should be removed before beginning a session. Sensitivity to dropouts occurring during recording is decreased by overbiasing the tape, for as the dropout occurs and the oxide coating moves away from the head, the bias put on the tape decreases, causing the output of the tape to increase.

Head Alignment

Another very important factor affecting the quality of a recording is the physical head alignment. The record and playback heads in the head block have five adjustments: height, zenith, wrap, rack, and azimuth.

The *height* determines where across the width of the tape the signal will be recorded. If the track is recorded and reproduced on heads with different height settings, not all of the recorded signal will be reproduced, resulting in a poorer signal-to-noise ratio, and if the tape is multitrack, cross talk will occur between the tracks.

Zenith refers to the tilt of the head toward and away from the tape. The zenith must be adjusted so that the tape contacts the top and bottom of the head with the same force, otherwise the tape will tend to *skew*.

Skewing is the riding up or down of a piece of tape on a head or guide, so that its edges are no longer parallel to the top plate of the transport. This causes variations in the effective height, azimuth, and tape speed.

Wrap refers to the angle at which the tape bends around the head and the location of the gap in the angle. The wrap determines the intimacy of the tape-to-head contact and thus controls the sensitivity of the head to dropouts.

Rack refers to how far forward the head is and determines the pressure of the tape against the head. The farther forward the head is, the greater the pressure.

Azimuth refers to the tilt of the head in the plane parallel to tape motion. The head gaps should be perpendicular to the tape, so that all of the gaps are in phase with each other and so that the signal is in phase at all points within each gap.

(A) Correct. (B) Bottom of head too far forward. (C) Top of head too far forward.

Fig. 4-26. Grease pencil wear patterns for checking zenith adjustment.

Height can be adjusted either optically or by using a test tape of the proper track configuration and adjusting the head height for maximum output at 1 or 3 kHz. Zenith is tested by covering the pole pieces with white grease pencil and playing a piece of tape to observe the pattern formed as the grease pencil wears off (Fig. 4-26). Naturally this tape cannot be used for recording afterwards since the grease pencil rubs off onto the oxide. The edges of the wear pattern should be parallel. If they are not, the zenith must be adjusted using the screws on the head block.

Wrap angle is checked at the same time as zenith by making sure that the wear pattern is centered around the gap. Need for rack adjustment is indicated if the grease pencil wear pattern is wider on the record head than the playback head, or vice versa.

Azimuth can be tested by deliberately skewing the tape across the heads (by pushing up and down on the edges of the tape right in front of the head), while reproducing the 15-kHz band of a standard alignment tape. If the output increases, the azimuth needs to be adjusted. It can be adjusted by either of two methods, both of which use a full-track standard alignment tape.

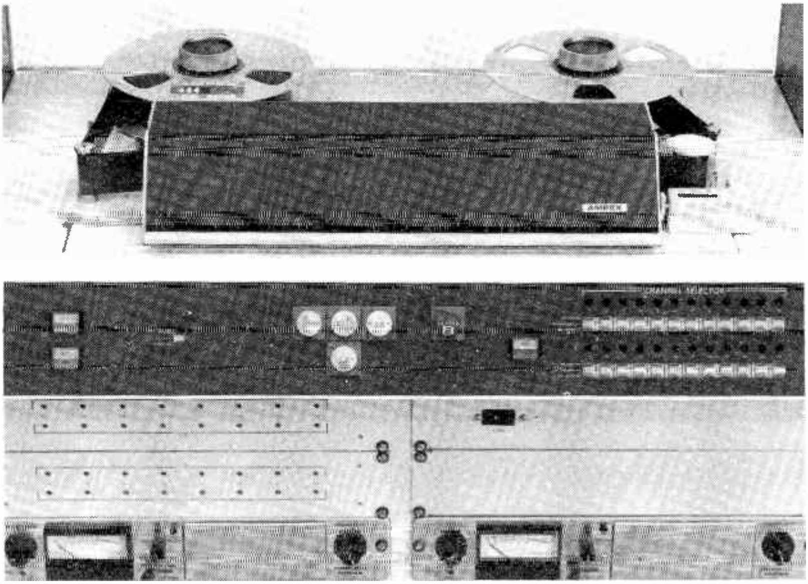
The first method is to play the 15-kHz section of the tape and adjust the azimuth for the highest output on all channels. This will have to be a compromise because some channels will rise as others fall on either side of the correct setting. The peak at the correct setting is fairly sharp with smaller broader peaks to either side of it. In order to be sure of finding the proper peak, the head gap should be perpendicular to the tape path before the adjustment is attempted.

The second method uses the phase of the 12- or 15-kHz signal to find the correct setting. After finding the peak, as in the first method, the output of the top channel of the head is fed into the vertical input of an oscilloscope, and the bottom channel output is fed into the horizontal input of the scope. The pattern resulting on the face of the scope represents the relative phase of the two channels. A straight line sloping up 45° to the right indicates that the two channels are in phase, a circle indicates that they are 90° out of phase, and a straight line sloping upwards 45° to the left indicates that the two channels are 180° out of phase. The azimuth should be adjusted so that the two channels are in phase.

On a multitrack head, it is not possible to get all of the gaps in phase at the same time. This is due to the *gap scatter* which occurs in manufacturing, so the phase adjustment is made for the outer tracks. The inner tracks will then usually be less than 60° out of phase with one another. The record-head azimuth can be set in the same way as the playback-head azimuth by playing the test tape back in the sync mode. Erase head azimuth is not very critical because its gap is very wide. It is correct as long as it is approximately at right angles to the tape path.

Although virtually all 2-, 4-, 8-, and many 16-track heads have adjustable alignment, the Ampex 16-track heads do not. Ampex feels that 16-track head alignment cannot be properly done by most studios and that the heads can be locked in place tight enough after alignment at the factory to withstand any shocks they might encounter. Head alignment need not be done too frequently. Its necessity is indicated by the deterioration of performance of the tape machine as indicated by the inability to align the electronics to their normal response characteristics. It should be checked periodically by the studio technicians.

Operationally, the MM1000 is very similar to the AG440, but the MM1000 has several extra features. For ease of operation, the safe/ready/sel sync switches are duplicated on the MM1000 front panel (Fig. 4-27). In addition, a master play/sync switch enables the operator to place all channels in sync with one control. The front-panel mode controls are in series with the individual channel safe/ready switches so that the switches on the electronics units must be left in the ready mode at all times. If they are put into the safe mode, the machine cannot be put into the record mode from the front-



Courtesy Ampex Corp.

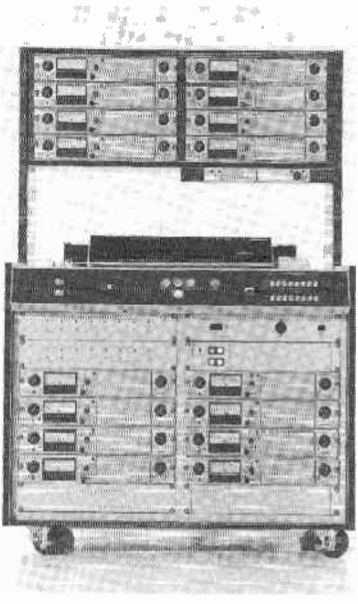
Fig. 4-27. The front panel of an Ampex MM1000 24-track recorder.

panel controls. If they are put in the sync position, the channels will remain in the sync mode regardless of the front-panel mode switch settings. The MM1000 also provides switching so that in the master sync mode tracks to be recorded on will play back in the sync mode until the record button is pressed, at which point they go into the record mode.

On the AG440 each track playing back in the sync mode has to be switched to the ready mode before it can be put into the record mode. When not recording, all tracks on the MM1000 play back in sync with one another without the operator doing any switching, but on the AG440 the operator must switch all tracks to either safe/ready or sel sync in order for all tracks to play back in sync with each other. The MM1000 has a counter which can be used to locate

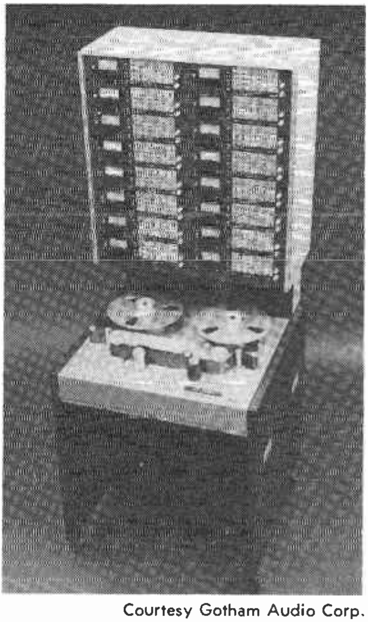
Courtesy Ampex Corp.

Fig. 4-28. Ampex AG440 2- and 4-track recorders.



Courtesy Ampex Corp.

Fig. 4-29. An Ampex MM1000 16-track recorder.

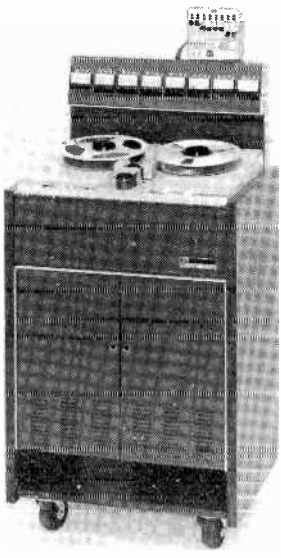


Courtesy Gotham Audio Corp.

Fig. 4-30. The Studer A80 16-track recorder.

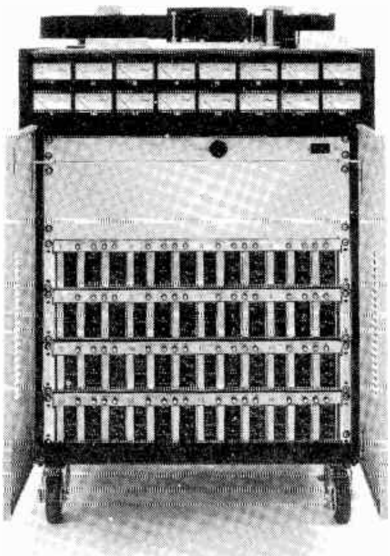
Courtesy Dictaphone Corp.

Fig. 4-31. The Scully Model 100 16-track recorder.



Courtesy Mincom Division, 3M Co.

Fig. 4-32. The 3M M56 8-track recorder with remote control unit.

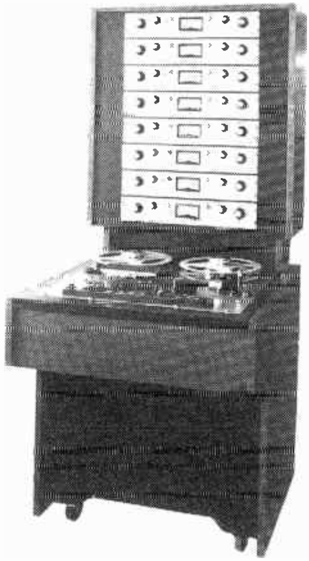


Courtesy Ampex Corp.

Fig. 4-33. The Ampex MM1100 16-track recorder.

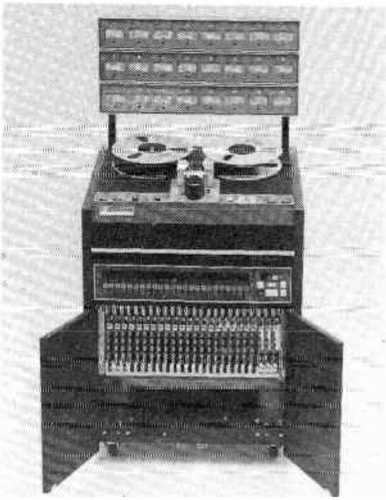
Courtesy Dictaphone Corp.

Fig. 4-34. The Scully Model 280B 2-track recorder.



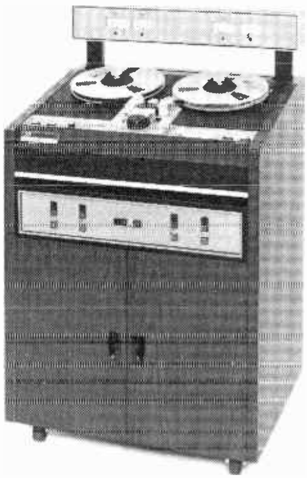
Courtesy Dictaphone Corp.

Fig. 4-35. The Scully Model 284 8-track recorder.



Courtesy Mincom Division, 3M Co.

Fig. 4-36. The 3M M79/24, 24-track recorder. This machine is also available in 8- and 16-track versions.



Courtesy Mincom Division, 3M Co.

Fig. 4-37. The 3M M79/4 4-track recorder.

selections on a reel or, since it is calibrated in minutes and seconds (at 15 ips), it can be used to time selections at high speed.

The tape-tension switch for the MM1000 is located inside the machine and is accessible by opening the end panel of the machine, while the tension switches on the AG440 are located on the front panel of the machine. The MM1000's switch simultaneously adjusts the take-up and holdback tensions for use with either 1- or 2-inch tape, while the switches on the AG440 adjust take up and holdback individually and are set on the basis of the reel hub size used: high tension for the large NAB hubs used on 10½-inch reels and low tension for the small EIA hubs used on 5- and 7-inch reels.

Special circuits are included in professional recorders to prevent clicks and pops from being recorded on the tape when the record mode is initiated or stopped. This makes it possible to overdub onto tracks which are partially acceptable by *punching in* to the record mode on the original track at the point that the performance becomes unacceptable and *punching out* of record when the unacceptable section has been repaired, retaining the rest of the original performance. This saves time in overdubbing in that only the faulty part of the performance need be repeated. Time is also saved in mixing since the entire part is recorded on one track rather than on several which must be combined. Since the pops are suppressed by turning the bias on and off slowly, a certain amount of time must exist between the end of the material to be retained and the beginning of the material to be punched in so that the bias can come up to full strength before the new material is performed. If there is not enough space, part of the new material may not be recorded. Similarly, space must exist between the end of the punched-in section and

Courtesy Mincom Division, 3M Co.

Fig. 4-38. The 3M Selectake unit provides a numercial readout of tape position when used with a 3M recorder.

the rest of the original performance to prevent part of the original from being erased.

Since the musician must hear his original performance up to the point he is to begin overdubbing, the channel in question must provide him with sync-head playback in his headphones. When he begins recording, however, he must hear what he is currently playing, with no time delay. This means that either the recorder output selector must be switched to the input mode, or that the console must be switched from the tape playback to the program mode. To eliminate possible confusion on the part of the engineer arising from simultaneously punching into record and switching the monitoring from tape to program, most 16- and 24-track recorders have an overdub mode in which tracks punched into record automatically switch to feed their input signal to the channel output for monitoring, while nonrecording tracks feed tape playback to their outputs. The console can be left in the tape playback mode, and all monitoring can be done through the tape machine.

Figs. 4-28 through 4-38 are some of the tape machines currently in use in recording studios.

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2. Daniel, Eric D. "Tape Noise in Audio Recording." *Journal of the Audio Engineering Society*, Vol. 20, No. 2, March 1972, pp. 92-99.
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CHAPTER 5

Signal-Processing Equipment

EQUALIZERS

One of the most important signal-processing devices used in the multitrack studio is the frequency equalizer. This device gives the engineer control over the harmonic balance or *timbre* of instruments heard by the listener and can be used to compensate somewhat for deficiencies in microphone frequency response or for deficiencies in the sound of an instrument. The frequency equalizer has several uses: to make the sounds from several mikes or several tape tracks blend better than they would otherwise, to match the sound of an over-dubbed instrument to the same instrument recorded with a different mike or in a different place, to make an instrument sound completely different from the way it normally does for a particular effect, and to increase the separation between instruments by rolling off the leakage frequencies.

Equalization (*EQ*) refers to altering the frequency response of an amplifier so that certain frequencies are more or less pronounced than others. It is specified as plus or minus a certain number of dB at a certain frequency. Although only one frequency is specified at a time, the frequency set on equalizers used in the studio actually refers to a curve, so signals at 4 kHz and 6 kHz are also boosted somewhat by adding EQ of +4 dB at 5 kHz (Fig. 5-1). The amount of boost or cut at frequencies other than the one named is determined by whether the curve is peaking or shelving, by the *Q* or bandwidth of the curve, and by the amount of boost or cut at the named frequency.

The equalization available in many older design equalizers such as those by Lang and Pultec is performed by passive filters, followed

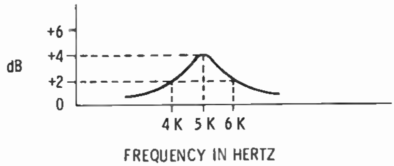


Fig. 5-1. A 5-kHz peak boost curve.

by an amplifier used to eliminate the insertion loss of the filters. *Insertion loss* refers to the signal level difference in dB between having the filters in the circuit and having them out of the circuit and is due to power losses that occur as a result of the manner in which the equalizer works. With all boost and cut controls set to zero, the filters attenuate all frequencies the same amount, and the amplifier brings the filter outputs back up to the level of the input signal. Cutting the signal at a certain frequency increases the attenuation above the preset level at the cut frequency, while boosting a signal decreases the attenuation at that frequency.

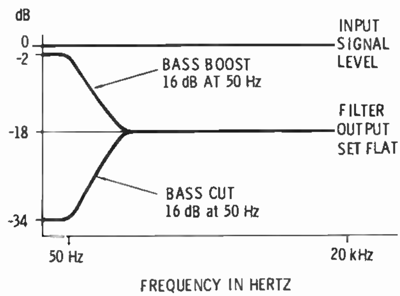


Fig. 5-2. Operational curve of a passive equalizer.

Operation of the filter is illustrated in Fig. 5-2. The input signal is attenuated by 18 dB by the filter when it is set for flat response. Boosting 50 Hz by 16 dB decreases filter attenuation at 50 Hz to 2 dB. Cutting 50 Hz by 16 dB increases filter attenuation to 34 dB at 50 Hz. Fig. 5-3 is a block diagram showing typical signal levels in a passive equalizer set for flat response. The block diagram of Fig. 5-4 shows levels in the equalizer in Fig. 5-3 with the filter set for 16-dB boost at 50 Hz.

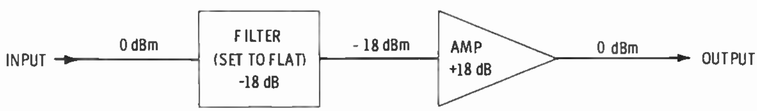


Fig. 5-3. Block diagram showing typical signal levels in a passive equalizer set for flat response.

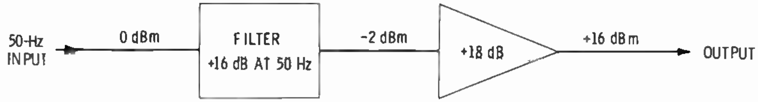
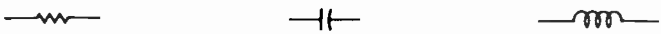


Fig. 5-4. Block diagram showing levels in the equalizer in Fig. 5-3 with the filter set for 16-dB boost at 50 Hz.

The filter uses only three types of components (aside from switches): resistors, capacitors, and inductors (Fig. 5-5). *Resistors* present equal opposition to the flow of signals of all frequencies. This opposition is called *resistance* (R) (Fig. 5-6). *Capacitors* present a frequency-dependent type of resistance to signal flow called



(A) Resistor. (B) Capacitor. (C) Inductor.

Fig. 5-5. Schematic drawings of electronic components.

capacitive reactance (X_c). Capacitive reactance increases as the frequency of the signal decreases and is computed by the formula:

$$X_c = \frac{1}{2\pi fC} \qquad \text{(Eq. 5-1)}$$

where,

f is the frequency in hertz,
C is the capacitance in farads.

Thus a capacitor passes high frequencies better than low frequencies (Eq. 5-1) and will not pass direct current (a signal of zero frequency) as shown in Fig. 5-7. *Inductors* (coils of wire) also present a frequency-dependent type of resistance called *inductive reactance* (X_L) to the signal. This reactance increases as the frequency of the signal increases and is computed by the formula

$$X_L = 2\pi fL \qquad \text{(Eq. 5-2)}$$

where,

f is the frequency in hertz,
L is the inductance in henrys.

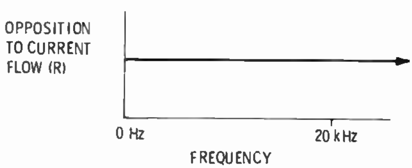


Fig. 5-6. The frequency response of a resistor is flat.

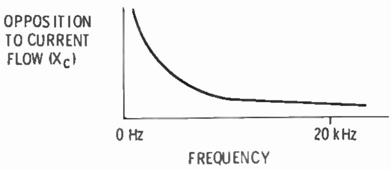


Fig. 5-7. Capacitive reactance (X_C) is inversely proportional to frequency.

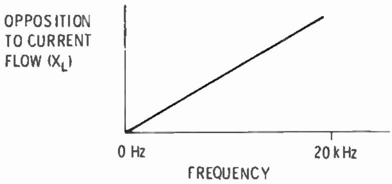
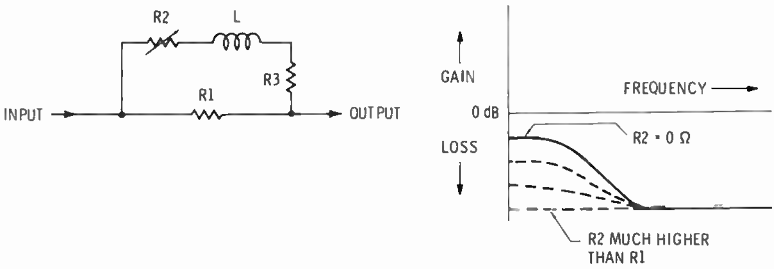


Fig. 5-8. Inductive reactance (X_L) is proportional to frequency.

Therefore, an inductor passes low frequencies better than high frequencies (Fig. 5-8). In fact, it passes direct current with no opposition at all except for the small resistance of the wire of which it is made (all conductors have some resistance). Inductors and capacitors do not dissipate power through their reactance, rather they store power. Resistors dissipate power by converting it into heat.

Low-Frequency Shelf Boost

Low-frequency shelf boost is achieved by feeding the signal to the junction of fixed resistor R1 and the series connection of variable resistor R2, inductor L, and fixed resistor R3 (Fig. 5-9A). The low frequencies find less opposition in the path through R2, L, and R3 which leads directly to the output of the filter, and thus they take



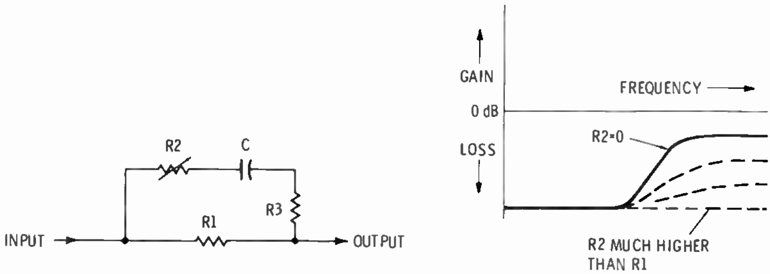
(A) Filter. (B) Loss curves.

Fig. 5-9. A low-frequency shelf boost filter and loss curves.

this route. The higher frequencies find less opposition to their flow through R1. They take this path and are dissipated a certain amount. The lows are dissipated in proportion to the setting of variable resistor R2 which is the front-panel boost control. The different boost frequencies are selected by switching different value inductors into the circuit. Thus, it is not the amount of boost of the low frequencies that is varied, rather the amount of their attenuation. Fig. 5-9B shows loss curves for the network.

High-Frequency Shelf Boost

A high-frequency shelf boost can be achieved similarly to the low-frequency boost. To do this a capacitor is substituted for the inductor so that the high frequencies find the path through R2, C, and R3 easier than flowing through R1 (Fig. 5-10A). Changing the capacitor changes the frequency of equalization. Fig. 5-10B shows the loss curves for the network.

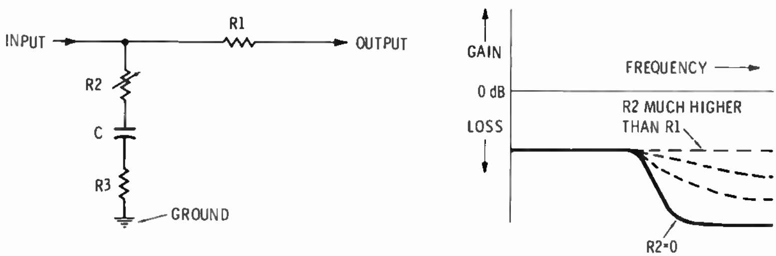


(A) Filter. (B) Loss curves.

Fig. 5-10. A high-frequency shelf boost filter and loss curves.

High-Frequency Shelf Cut

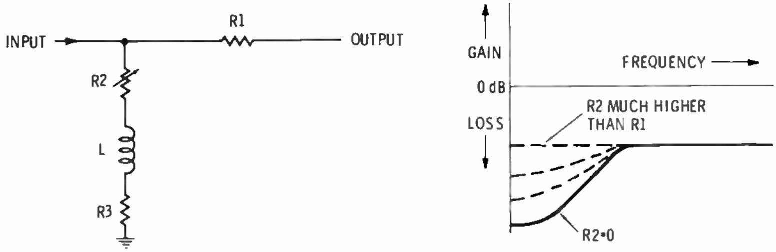
In the high-frequency shelf cut filter, the signal is fed to the junction of R1 and the series connection of R2, C, and R3 which is grounded at the other end (Fig. 5-11A). High-frequency signals



(A) Filter. (B) Loss curves.

Fig. 5-11. A high-frequency shelf cut filter and loss curves.

find it easier to be dissipated through R2, C, and R3 to ground than to pass through R1. The setting of the variable resistor determines the proportion of highs that are thus removed from the signal. The low frequencies follow the path through R1. Fig. 5-11B shows loss curves for the network.



(A) Filter. (B) Loss curves.

Fig. 5-12. A low-frequency shelf cut filter and loss curves.

Low-frequency shelf cut is achieved simply by substituting an inductor for the capacitor (Fig. 5-12A). Fig. 4-12B shows loss curves for the network.

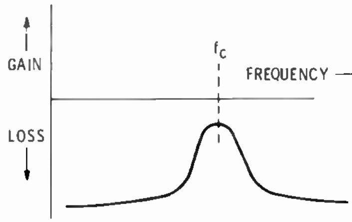
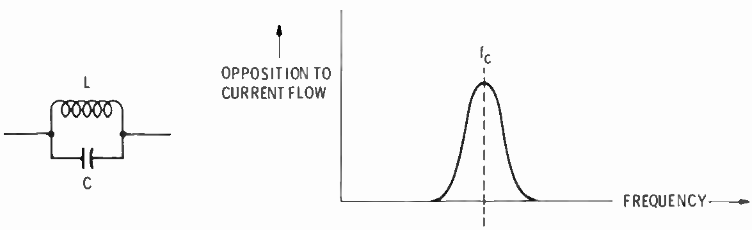


Fig. 5-13. A peaking curve.

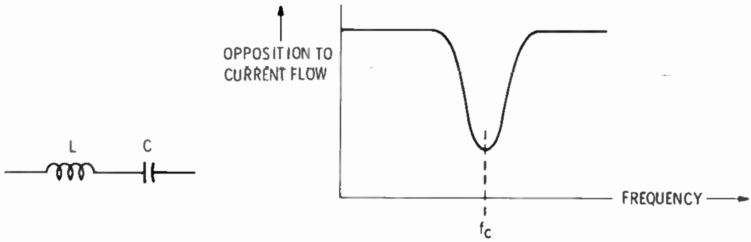
PEAKING FILTERS

The preceding filters all produce shelving EQ. That is, the attenuation increases (or decreases) with frequency at a particular rate to a point where it levels off and remains constant even though the frequency continues to change in the same direction. The high- and middle-frequency boost filters in the Lang and Pultec program and midrange equalizers, however, provide a peaking curve. That is, the attenuation decreases as frequency rises to the boost frequency,



(A) LC circuit (B) Frequency-response curve.

Fig. 5-14. A parallel LC circuit and frequency-response curve.

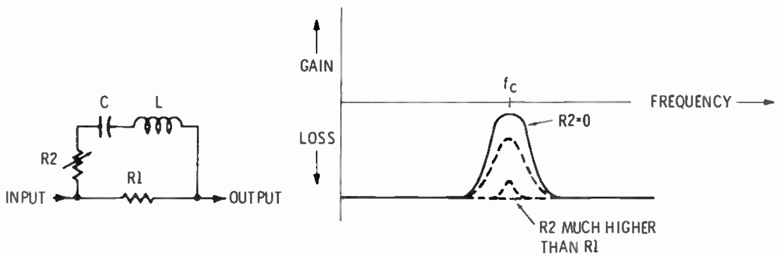


(A) LC circuit.

(B) Frequency-response curve.

Fig. 5-15. A series LC circuit and frequency-response curve.

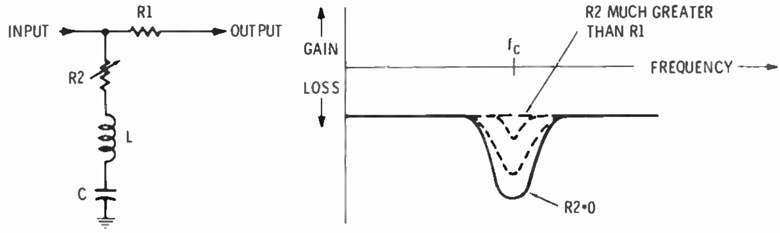
reaches a minimum at that frequency, and increases as signal frequency continues to increase (Fig. 5-13). This type of curve is created by a combination of a capacitor and an inductor. If these two reactors are connected in parallel, they provide a large opposition to signal flow at one band of frequencies and less opposition at all other frequencies (Fig. 5-14A). Fig. 5-14B shows the frequency-response curve for the circuit. If they are connected in series, they provide little opposition to one band of frequencies and much opposition at all other frequencies (Fig. 5-15A). Fig. 5-15B shows the frequency-response curve for the circuit. The frequency at which the parallel circuit provides most opposition and the series circuit provides least opposition is called the *center* or *resonant* frequency and occurs at the frequency where the capacitive reactance equals the inductive reactance. Thus, a peaking boost can be obtained by feeding the signal to the junction of a fixed resistor in parallel with a series combination of a variable resistor and a series LC (inductor-capacitor) circuit (Fig. 5-16A). The variable resistor controls how much of the signal near the resonant frequency is allowed to flow through the LC circuit and avoid the attenuation of the fixed resistor. Signals far removed from the resonant frequency find less opposition through the fixed resistor and they take that route, losing power



(A) Filter.

(B) Frequency-response curves.

Fig. 5-16. A peak boost filter and frequency-response curves.



(A) Filter. (B) Frequency-response curves.

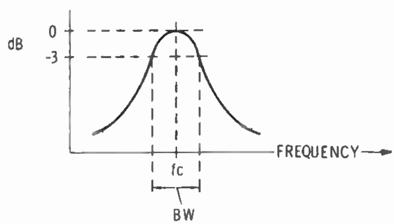
Fig. 5-17. A dip cut filter and frequency-response curves.

as they do. At the far side of the fixed resistor, the signals following the two different paths are combined, and the resonant frequency has a higher level since it has been attenuated less. Fig. 5-16B shows the frequency-response curves.

The LC circuit can also produce a dip in frequency response (Fig. 5-17A). To achieve this, the far end of the circuit is connected to ground so that the resonant frequency is bypassed to ground to an extent determined by the variable resistor. Other frequencies find it easier to take the route through the fixed resistors only. Fig. 5-17B shows the frequency-response curves.

The shape of an LC filter response is determined by the Q of the circuit, which depends on the amount of resistance in the inductor winding. The lower the resistance for the same value of inductance, the higher the Q of the LC circuit and the sharper the peak or dip that can be produced. The *bandwidth* of the circuit is the number of hertz between the frequencies above and below the center fre-

Fig. 5-18. The bandwidth (bw) of a peaking filter is the number of hertz between the points that are 3-dB down from the center frequency.



quency which are 3 dB different in amplitude from the level of the center frequency (Fig. 5-18). The Q of a circuit is computed by dividing the center frequency in Hz by the bandwidth in Hz. By rearranging this formula, it can be seen that the bandwidth can be changed by varying the circuit Q while holding the center frequency constant. This can be easily done by adding resistance to the LC circuit. The bandwidth controls of the Lang and Pultec equalizers do just that. As the control is rotated from the sharp to the broad position, the amount of resistance is increased, decreasing the Q and increasing the bandwidth. In the sharp position, only frequencies close to resonance are affected by the filter, but as the bandwidth is increased, frequencies farther from resonance are also affected.

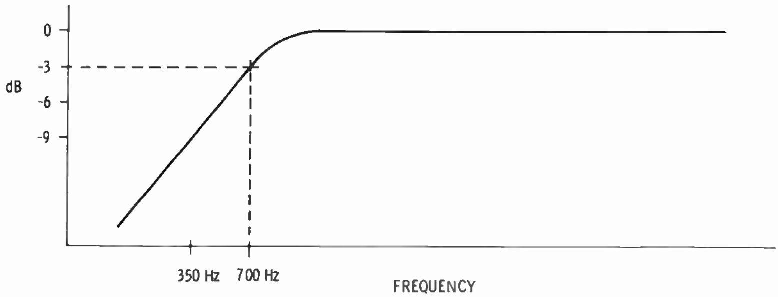


Fig. 5-19. A 700-Hz, high-pass filter with a slope of 6 dB per octave.

HIGH- AND LOW-PASS FILTERS

Other types of equalizers are the *high-pass* and *low-pass* filters. As their names imply, certain frequencies are passed at full level, while others are attenuated. Frequencies which are attenuated less than 3 dB are said to be inside the *passband*, while those attenuated by more than 3 dB are in the *stop band*. The frequency at which the signal is attenuated exactly 3 dB is called the *turnover* or *cutoff frequency* and is used to name the filter. Ideally, attenuation would become infinite immediately outside the passband, but in practice this is not attainable. In the simplest case, attenuation increases at a rate of 6 dB per octave. This rate is called the slope of the filter. Other slopes in common use are 12 and 18 dB per octave. For example, Fig. 5-19 shows a 700-Hz high-pass filter response curve with a slope of 6 dB per octave, while Fig. 5-20 shows a 700-Hz low-pass filter response curve with a slope of 12 dB per octave.

High- and low-pass filters differ from shelving EQ in that their attenuation does not level off outside the passband, rather it con-

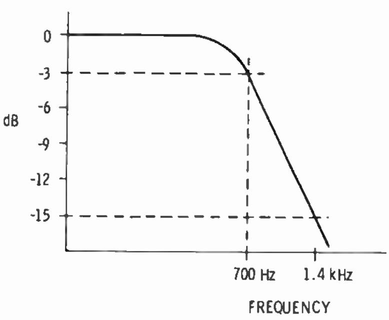


Fig. 5-20. A 700-Hz, low-pass filter with a slope of 12 dB per octave.

tinues to increase. A high-pass filter in combination with a low-pass filter can be used to create a bandpass filter with the bandwidth of the passband controlled by their turnover frequencies and the Q controlled by their slopes. A bandpass filter of this type has a flatter response curve (Fig. 5-21) in the passband than an LC bandpass filter.

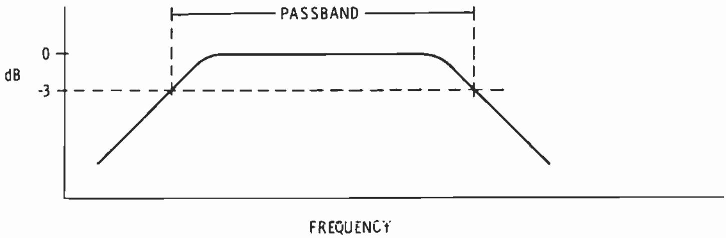


Fig. 5-21. A bandpass filter created by combining high- and low-pass filters with different cutoff frequencies.

Graphic equalizers utilize LC circuits similar to those described earlier. The main difference between them and other equalizers is that the same knob controls boost and cut, and the relative positions of the knobs on the unit provide a visual display of the frequency response curve of the unit. Graphics provide control over many frequencies simultaneously, while other equalizers divide the frequency spectrum into three or four bands and provide control over one selectable frequency in each band.

The equalizer circuits described above are called *passive equalizers* because the circuitry which changes the frequency balance does no amplification. The amplifiers merely increase the level of the filter output to compensate for filter insertion loss. All of the response curves generated by passive equalizers can also be achieved through the use of *active filters,* which eliminate the need for inductors, thereby greatly reducing the size of the equalizer units. Most new consoles have extensive equalization available in the input module so that it can be inserted into the circuit and varied without the engineer having to leave his seat or, even more important, without having to change his position with respect to the monitor speakers. These active circuits perform better than passive ones because inductors can cause hum as well as harmonic and intermodulation distortion.

In addition to its use in modifying sound in tape recorders (NAB pre and post-EQ) and discs (RIAA pre- and post-EQ), equalizers can be used to remove hum and other undesired discrete-frequency noises. A *notch filter* is used for this purpose. This filter can be tuned to attenuate a particular frequency and has a very narrow bandwidth so that it has little effect on the rest of the program (Fig. 5-22). Notch filters are used more in film-location sound than in studio recording because the problems encountered in location work which make use of notch filters necessary are usually not present in a well-designed studio. A tunable notch filter, however, can be useful in removing resonant rings from certain instruments.

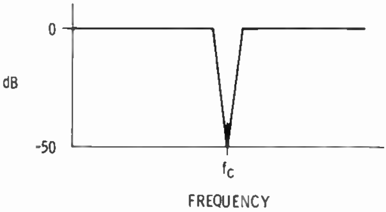


Fig. 5-22. Frequency response of a notch filter.

Equalization must be done by ear, but it is helpful to have an idea of what frequencies will give the desired effect. Useful EQ frequencies for some common instruments are provided in Table 5-1, and the frequency ranges of instruments are illustrated in Fig. 5-23.

Table 5-1. Useful EQ Frequencies for Common Instruments

Instrument	Frequency Ranges of Interest
Bass guitar	Attack or pluck is increased at 700 or 1000 Hz, bottom added at 60 or 80 Hz, string noise at 2.5 kHz.
Bass drum	Slap at 2.5 kHz, bottom at 60 or 80 Hz.
Snare	Fatness at 240 Hz, crispness at 5 kHz.
Hi hat and cymbals	Shimmer at 7.5 kHz to 10 kHz. Clank or gong sound at 200 Hz.
Tom toms	Attack at 5 kHz, fullness at 240 Hz.
Floor toms	Attack at 5 kHz, fullness at 80 or 120 Hz.
Electric guitar	Full at 240 Hz, bite at 2.5 kHz.
Acoustic guitar	Body at 240 Hz, clarity at 2.5 kHz, 3.75 kHz, and 5 kHz, with the sound thinning out as frequency rises. Bass strings at 80 or 120 Hz.
Organ	Bass 80 to 120 Hz, presence 2.5 kHz, body 240 Hz.
Piano	Bass 80 to 120 Hz, presence 2.5 kHz to 5 kHz, thinning as frequency is raised. Honky-tonk sound at 2.5 kHz as bandwidth is sharpened. Resonance and echo at 25 to 50 Hz.
Horns	Fullness at 120 to 240 Hz, shrill at 7.5 or 5 kHz.
Voice	Presence at 5 kHz, sibilance at 7.5 kHz to 10 kHz, boominess at 200 to 240 Hz, fullness at 120 Hz.
Strings	Scratchiness at 7.5 to 10 kHz, fullness at 240 Hz.
Harmonica	Fat at 240 Hz, electric at 2.5 kHz, acoustic at 5 kHz.
Conga	Resonant ring at 200 to 240 Hz. Presence and slap at 5 kHz.

In summary, the frequency spectrum can be divided up into six important sections, following de Gar Kulka's description [5]:

1. The very low bass between 16 and 60 Hz which encompasses sounds which are often felt more than heard, such as thunder in the distance. These frequencies give the music a sense of power even if they occur infrequently. Too much emphasis on this range makes the music sound muddy.

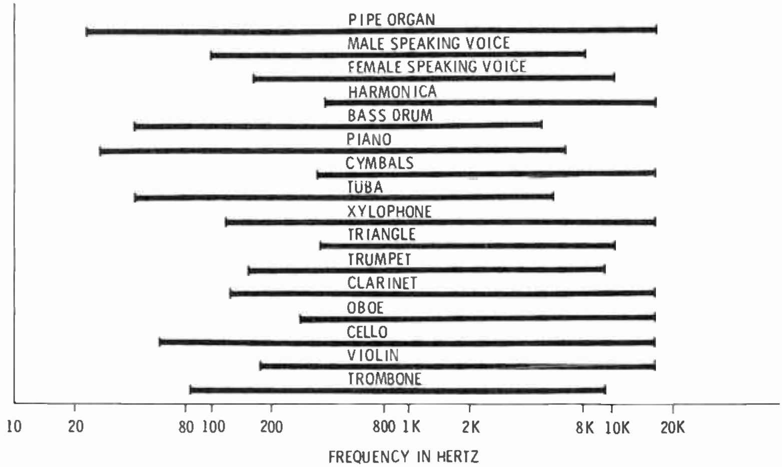


Fig. 5-23. Frequency ranges of instruments.

2. The bass between 60 and 250 Hz contains the fundamental notes of the rhythm section, so EQing this range can change the musical balance, making it *fat* or *thin*. Too much boost in this range can make the music sound *boomy*.
3. The midrange between 250 and 2000 Hz contains the low-order harmonics of most musical instruments and can introduce a telephone-like quality to music if boosted too much. Boosting the 500- to 1000-Hz octave makes the instruments sound horn-like, while boosting the 1- to 2-kHz octave makes them sound tinny. Excess output in this range can cause listening fatigue.
4. The upper midrange between 2 and 4 kHz can mask the important speech recognition sounds if boosted, introducing a lisping quality into a voice and making sounds formed with the lips such as "m," "b," and "v" indistinguishable. Too much boost in this range, especially at 3 kHz, can also cause listening fatigue. Dipping the 3-kHz range on instrumental backgrounds and slightly peaking 3-kHz on vocals can make the vocals audible without having to decrease the instrumental level in mixes where the voice would otherwise seem buried.

5. The presence range between 4 and 6 kHz is responsible for the clarity and definition of voices and instruments. Boosting this range can make the music seem closer to the listener. Adding 6 dB of boost at 5 kHz makes a mix sound as if the overall level has been increased by 3 dB. As a result of this effect, many record companies and mastering engineers make a practice of adding a few dB of boost at 5 kHz to make their product sound louder. Reducing the 5-kHz content of a mix makes the sound more distant and transparent.
6. The 6- to 16-kHz range controls the brilliance and clarity of sounds. Too much emphasis in this range, however, can produce sibilance on vocals.

The best way to learn to use an equalizer is to set the amount of boost to a maximum and change the boost frequency until the desired range of the instrument to be EQed is found. The amount of boost can then be decreased until the desired effect is obtained. Attenuation of a frequency range can be achieved in a similar manner. Drawing a curve of the effect introduced by the equalizer when the desired sound is attained can often aid in future equalizing and in visualizing what the controls actually do.

If boosting one range of an instrument creates the need to boost the other ranges, and they are then boosted, the effect achieved is simply that the overall level has been raised. This is more easily done with the input fader. If the increased fader level does not make the sound satisfactory, it may be that one range of frequencies is too dominant and requires attenuation. Just because the ear can hear from 20 Hz to 20 kHz does not mean that these frequencies should exist on every record. There are many times that the presence of these frequencies may detract rather than add to the desired effect.

As far as recording with EQ goes, there are differing opinions. If an engineer other than the one that records a multitrack tape is to mix it, he may have a very different idea of how the instruments should sound and may have to work very hard to counteract the EQ recorded by the original engineer. If everything is recorded flat, however, the producer and artists have to strain while they are trying to pass judgment on a performance to imagine how the instruments will sound later. It is also important to know how the instruments will sound with EQ during overdubbing so that the producer and artists can decide when a song has been "sweetened" enough. When several mikes are to be combined onto one channel of the tape, they can be EQed individually only before recording, so recording flat as a strict rule will prevent optimizing the sounds picked up by each mike. In addition, while recording with EQ does not change the perceived noise level, playing back with EQ does. EQ added on playback is also added to the residual tape noise of the track. So boosting highs, for example during playback, would make the tape hiss of that channel more audible than if the highs were boosted before recording. If the same engineer is to record and mixdown the tape, recording with EQ is usually not a problem. In any event, unless a special effect is desired, EQ should be used moderately, and microphone selection should be used to obtain good sound on an instrument. If an instrument is recorded wrong in the initial recording session, it can rarely be corrected later during mixing.

METERS, COMPRESSORS, LIMITERS, AND EXPANDERS

Since amplifiers and magnetic tape are limited in the range of signals they can pass without distortion, audio engineers need a means of determining whether the signals they are working with will be stored or transmitted without distortion.

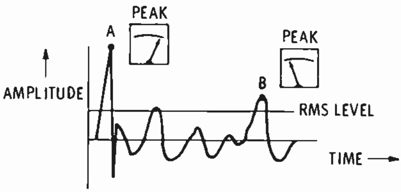


Fig. 5-24. A peak meter reads higher at point A than at point B, even though the loudness is the same.

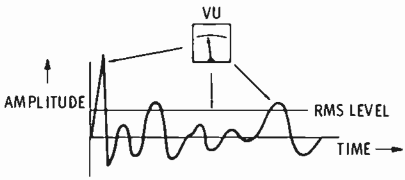


Fig. 5-25. A VU meter reads the rms level and ignores the peaks that do not contribute to loudness.

The most convenient method of indicating this is through the use of a visual device such as a meter. If preventing distortion on the tape were the only concern, peak-indicating meters could be used to display the maximum amplitude fluctuations of a waveform. The human perception of loudness, however, does not have much relationship to the peak level of signals. The meter may read higher at a certain point in the program, but the program may not sound any louder (Fig. 5-24). If the meter is to be used to set or maintain a certain volume level, a peak indication is of no use.

Since the perception of the ear to loudness is proportional to the rms value of a signal, a meter was designed that could read this level so that volume and meter indication would coincide (Fig. 5-25). The scale chosen for the meter was calibrated in volume units, and hence the name *VU meter* (Fig. 5-26). Zero VU is considered the standard operating level. Although VU meters do the job of indicating volume level, they ignore the short term peaks which can overload tape. These peaks can be from 8 to 14 dB higher than the

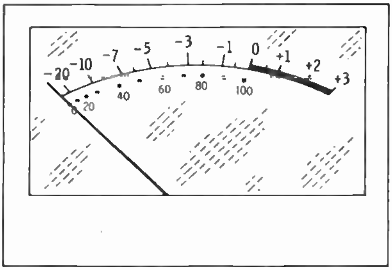


Fig. 5-26. A VU meter. The upper scale is calibrated in volume units for use in recording. The lower scale is percentage modulation for use in broadcasting.

rms values indicated. This means that the electronics must be designed so that unacceptable distortion does not occur until at least 14 dB above 0 VU (Fig. 5-27).

The difference between the maximum level which can be handled without excess distortion and the average operating level of the system is called *headroom*. Some studio-quality amplifiers are capable of outputs as high as 26 dB above 0 VU and thus have 26 dB of headroom. Magnetic tape, however, has limited headroom, for its dynamic range is such that providing the headroom necessary to prevent distortion of the peaks would make tape noise too audible during the rest of the program. The 3% distortion level for magnetic tape recorded on a tape machine that does not use a linearizer is only 6 dB above 0 VU, while the console amplifiers have distortion of less than 0.4% at this level. Typical VU meter specifications are given in Chart 5-1.

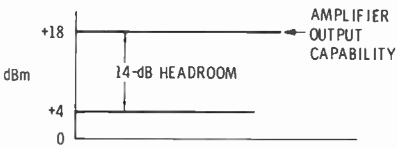


Fig. 5-27. If the console is to have 14 dB of headroom above 0 VU (+4 dBm), the amplifiers must be capable of output levels of at least +18 dBm.

Chart 5-2. VU Meter Specifications

Sensitivity:	Reads 0 VU when connected across a +4-dBm signal (1.228 volts in a 600-ohm circuit).
Frequency Response:	±0.2 dB from 35 Hz to 10 kHz. ±0.5 dB from 25 Hz to 16 kHz.
Overload Capability:	Can withstand ten times 0-VU level (+24 dBm) for 0.5 second, and five times 0 VU (approximately +18 dBm) continuously.

Volume units indicated on the meter are equal to dB for sine waves, but for other waves or for complex signals, the VU meter actually reads between the rms and peak values of the signal. For these waves volume units are larger than dB.

Proper record level for most program material is 0 VU although higher levels would be possible if it was certain that there would be no peaks in the program that would cause distortion. Since the VU meter would not read those peaks even if they existed, enough headroom must be provided for them to occur, unless the dynamic range of the system is changed. Thus, certain instruments with high peak-to-rms level ratios, such as the piano, should be recorded at less than 0 VU. If they are recorded at 0 VU, the 14-dB higher peaks would be 8 dB above the 3% distortion level of the tape. A −5- to −8-VU level would produce a less distorted recording.

Several meters are now available using *light emitting diodes* (*LEDs*) to provide level indication through the illumination of lights corresponding to different levels, rather than through the use of a meter pointer. These units follow peaks better than any meter can, giving virtually instantaneous display of the signal level. These units have the same defect as other peak meters, however, in that they do not indicate the loudness of the program. The ideal level monitor would be a combination of peak-responding lights with a VU meter so that both volume and dangerous peaks would be indicated.

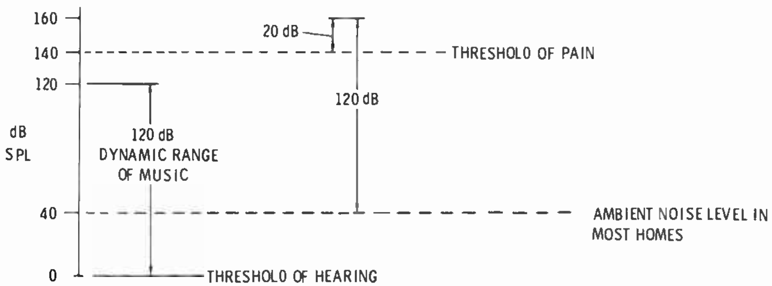


Fig. 5-28. The ambient noise level of most homes would obscure the lower 40 dB of a 120-dB dynamic range program. If the volume of the program is raised so that the softest portion is above the ambient noise, the loudest portion would be 20 dB above the threshold of pain.

The dynamic range of music is on the order of 120 dB, while the dynamic range of magnetic tape is only on the order of 60 dB, excluding the use of noise reduction systems which can add another 20 to 50 dB but still come short of the 120 dB of music. The dynamic range of discs is about 70 dB. Even if a recording and reproducing system with a 120-dB dynamic range were available, unless it was used in a noise-free environment, either the quiet passages would be lost in the ambient noise of the listening area (35- to 45-dB spl for the average home), or the loud passages would be too loud to bear. Similarly, if a 120-dB dynamic range recording was reproduced through a medium with a narrow dynamic range such as a-m radio (20 to 30 dB) or fm radio (40 to 50 dB), a lot of information would be lost in background noise (Fig. 5-28). To prevent these problems, the dynamic range is reduced to a level that is appropriate both for the medium through which it is to be reproduced and for comfortable listening in the average home. This reduction is accomplished by a combination of the engineer *riding the faders* and the use of a device called a *compressor.*

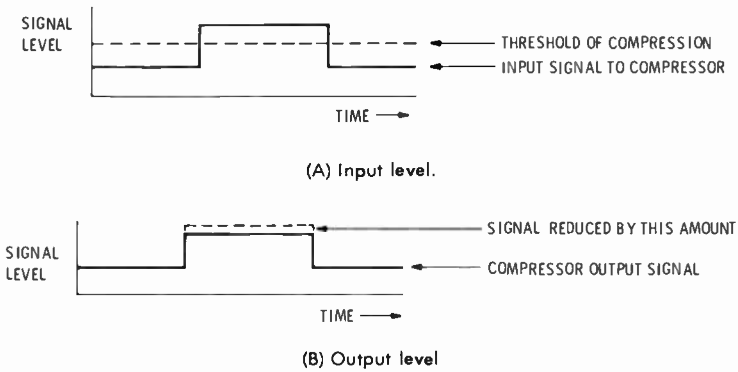


Fig. 5-29. The compressor reduces the level of the portion of the signal that exceeds the threshold.

A compressor is, in effect, an automatic fader. When the input signal exceeds a predetermined level (Fig. 5-29A), called the *threshold,* compressor gain is reduced and the signal is attenuated (Fig. 5-29B). The number of dB increase of input signal needed to cause a 1-dB increase in the output signal of the compressor is called the *compression ratio* or the *slope* of the compression curve. Thus, for a ratio of 4:1, an 8-dB increase of input produces a 2-dB increase in output. Since the signals generated by music vary in loudness and therefore may be above the threshold one instant and below it the next, the speed with which the gain is to be reduced after the signal exceeds the threshold and is to be restored after the input signal falls below the threshold must be determined. These speeds are determined by the *attack* and *release* times, respectively.

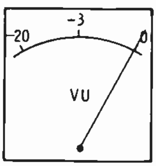
As stated earlier, the perception of the ear to the loudness of a signal is proportional to rms value, so large, short duration peaks do not noticeably increase the loudness of a signal. What is desired is to allow the signal volume to rise and fall but to lesser extents than the volume would if it were not controlled. If the peaks of the waveform were permitted to trigger *gain reduction,* the volume would actually decrease, rather than increase, by a smaller amount.

This would change the dynamics of the program noticeably and in an unacceptable way. To avoid the triggering of compression by these peaks, the attack time is set so that the waveform must exceed the threshold level for a time long enough to constitute an increase in the average level, and gain reduction will not decrease the overall volume. The attack time is defined as the time it takes for the gain to decrease by a certain amount, usually to 63% of its final value.

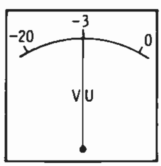
If the release time were set too short for the program material, i.e., if full gain were restored each time the signal fell below the threshold; *thumps*, *pumping*, and *breathing* would be heard due to the rapid rise of background noise as the gain is increased. Also, if a rapid succession of peaks were fed into the device, the program gain would be restored after each one, and the level of the program would be heard to rise after each peak. Since the level-sensing mechanism is sensitive to both positive and negative waveform excursions, extremely short release times could cause gain reduction twice each cycle, introducing harmonic distortion into the signal.

To eliminate these effects, longer release times are used so that repeated waveform excursions past the threshold cause gain reduction only once. The gain remains reduced through all of these excursions and returns to normal gradually. This makes the increase in background noise level less obvious, as well as making any gain changes that may be required shortly thereafter less drastic. If the release time is too long, however, a loud section of the program may cause gain reduction that persists through a soft section, making the soft section inaudible. The release time is defined as the time needed for the gain to return to a certain percentage of its no-gain reduction value (usually 63%).

Compressors usually have a built-in VU meter to allow monitoring of the amount of gain reduction taking place. The meter usually sits at 0 VU when the input signal is below the threshold and falls to the left to indicate the number of dB of gain reduction when the input signals exceeds the threshold (Figs. 5-30A and B). Some compressors use meters that read gain reduction directly, resting at the left



(A) The input signal to the compressor is below the threshold, and compressor gain is normal.



(B) The input signal exceeds the threshold causing 3 dB of gain reduction. The meter indicates that the gain is 3 dB lower than normal.

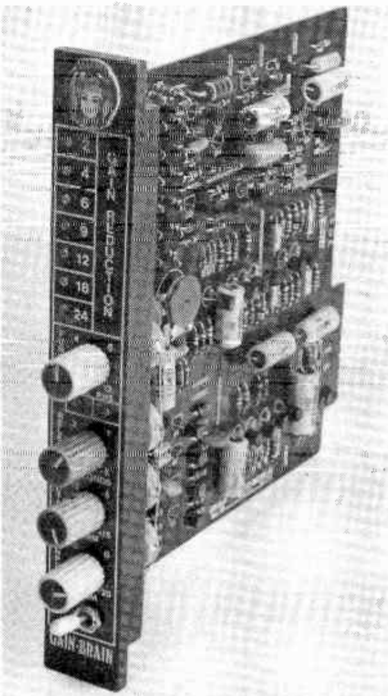
Fig. 5-30. The gain-reduction meter on a compressor.

(A) No gain reduction. (B) Gain reduction of 3 dB.

Fig. 5-31. The gain-reduction meters on some compressors are designed to read gain reduction directly.

side of the scale until the signal exceeds the threshold and then moving upscale to show gain reduction in dB (Figs. 5-31A and B). Still other units use indicator lights which are faster than meters to indicate the instantaneous gain reduction (Fig. 5-32). On some compressors, the meter is switchable to read either gain reduction or output level.

If the compression ratio is made large enough, the compressor becomes a *limiter* (Fig. 5-33). A limiter is used to prevent signal peaks from exceeding a certain level in order to prevent overloading amplifiers, tapes, or discs. An extreme case of a limiter is a *clipper* which



Courtesy Allison Research, Inc.

Fig. 5-32. The Gain Brain limiter uses light emitting diodes to indicate gain reduction.

chops off the top of any waveform exceeding the threshold level. A clipper could be said to have an infinite compression ratio. Most limiters have ratios of 10:1 or 20:1, although they are available with ratios up to 100:1. Since such a large increase in the input signal is needed to produce an increase in the output of a limiter, the likelihood of overloading the equipment following the limiter is greatly reduced, and the threshold of limiting can usually be set about 8 dB higher than the threshold of compression.

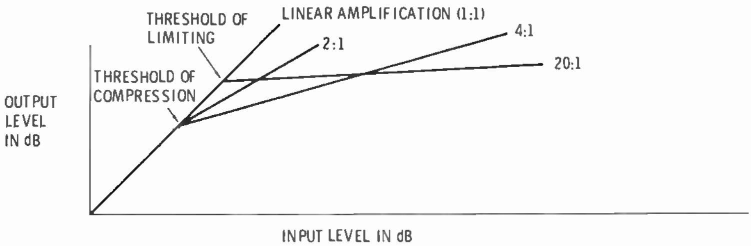


Fig. 5-33. The output of a compressor is linear below the threshold point and follows the slope of the selected compression or limiting curve above the appropriate threshold.

Limiting is most often used in recording to prevent the short-term peaks (which add little information to the program in proportion to the distortion they would cause if they saturated the tape, or the noise they would allow to enter the system if the signal were recorded at a level low enough so that the peaks would not distort) from reaching their full amplitude. Extremely short attack and release times are used so that the ear cannot hear the gain being reduced and brought back up. Limiting is used to remove only occasional peaks, for gain reduction on many successive peaks would be noticeable. If the program contains many peaks, the threshold should be raised and the gain reduced manually so that only occasional extreme peaks are limited.

Expansion is the process of decreasing the gain of a signal as its level falls and/or increasing the gain as the level rises. Thus, when the signal level is low (below the expansion threshold), gain is low and program loudness is reduced. When the signal level increases above the threshold, the gain is increased. Expanders increase the dynamic range of a program by making loud signals louder and soft signals softer. They can also be used as noise-reduction devices by adjusting them so that the noise to be removed is below the threshold level, while the desired signal is above the threshold as shown in Fig. 5-34.

The selection of proper attack and release times and degrees of compression, expansion, or limiting depends on the program material. In multitrack recording, dynamic range modifications usually deal with only single instruments or groups of instruments. In radio, television, and disc cutting, entire songs are compressed and the problem of using the proper parameters is more critical.

Limiting is usually only used for recording speech or instruments with *transients* (momentary high-level peaks) so that the signal can be recorded at a high level without overloading the tape. Compression is used for several reasons:

1. It minimizes the change in volume which occurs when an instrumentalist or vocalist momentarily changes his distance from the mike.
2. It can make the volume of the different ranges on an instrument the same. For example, some bass guitar strings are usually louder than the others on the same guitar, and the use of compression produces a smoother bass line by matching the volumes of the different notes. As another example, some instruments such as horns are louder in some registers than in others due to the amount of effort required to produce the notes. Compression would equalize the volume levels of the different registers.
3. Compression enables a signal to be made significantly louder in a mix, while increasing the overall signal-level reading on the meter only slightly, by increasing the ratio of average to peak levels.
4. Compression can be used to reduce sibilance in a voice by inserting a filter in the compressor circuit which causes it to trigger compression when an excess of high-frequency signal is present. A compressor used in this manner is often called a *de-esser*.

Producers often strive to cut their records as *hot* as possible, i.e., they want the recorded levels to be as far above the normal operating level as possible without blatant distortion. They talk about *competitive levels*, for they feel that louder records will stand out and

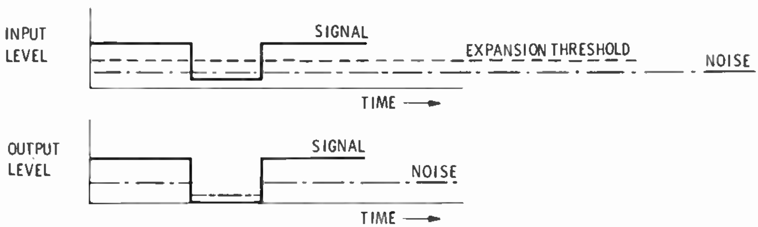


Fig. 5-34. When signal is above expansion threshold, gain is unity and both signal and noise appear at the output. When signal is below threshold, the gain decreases and reduces the level signal and the noise levels in the output.

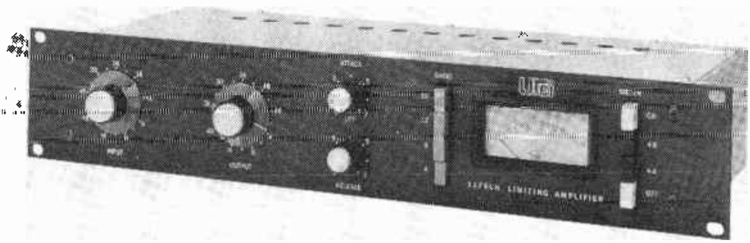
sound better than soft ones when a stack is being played on a record changer. In fact, a record that is one or two dB louder than another will sound better due to the Fletcher-Munson effect which will make the louder record appear to have more bass and more highs. To achieve these hot levels without distortion, compressors and limiters are often used in disc cutting to remove peaks and raise the average level of the program so that the disc will be louder than it otherwise would be.

Compressing a mono mix is done in the same manner as compressing a single instrument, except that the adjustment of the threshold, attack, release, and ratio controls is more critical to prevent pumping due to instruments that are prominent in the mix. Compressing a stereo mix gives rise to an additional problem. If two individual compressors are used, a peak in one channel will reduce the gain on that channel causing anything that was centered between the two speakers to jump toward the channel not undergoing compression. To avoid this *center shifting*, most compressors have provision for connecting them in stereo with a second compressor of the same make and model. This connection mixes the outputs of the signal-level sensing circuits of the two units, so that a signal which causes gain reduction in one channel will cause equal gain reduction in the other channel, preventing the center information from moving.

Expanders can be used to shut off mike channels when the instrument to be picked up by a particular mike is not playing, in order to prevent leakage from headphones or other instruments from being recorded on the tape. In mixing, expanders can be used as noise-reduction devices to turn off tracks when nothing is recorded on them, thus removing the tape noise that would otherwise be added to the mix from them. When the expander detects a signal above the expansion threshold, it turns the track back on. Used in conjunction with a compressor, an expander can eliminate the rush of hiss that would otherwise be produced when the signal stops and compressor gain increases.

Several examples of compressors and limiters in common use are the Universal Audio 1176LN, the Teletronix LA-2A and LA-3A, the Pye compressor, and the Spectra Sonics 610.

The Universal Audio 1176LN (Fig. 5-35) has compression ratios of 4:1 and 8:1 as well as two limiting ratios of 12:1 and 20:1. It has adjustable attack and release times and adjustable output level after the gain reduction section. The amount of gain reduction is selected by setting the meter selector switch to GR and advancing the input level control until the desired gain reduction is either heard or indicated on the meter. The output level control is then adjusted so that the desired level is presented to the input section of the console. The output level can be monitored by depressing either the +4 or +8 meter selector switch. The +4 switch calibrates the VU meter so that 0 VU equals +4 dBm, while with the +8 switch, 0 VU equals +8 dBm. It is good practice for the output level of all compressors to be set so that their output does not exceed 0 VU on the highest meter calibration scale to prevent overloading their output amplifiers.



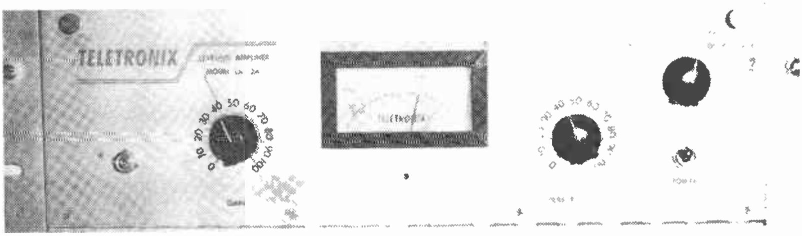
Courtesy United Recording Electronic Industries

Fig. 5-35. The Universal Audio 1176LN.

Since very fast release times (from 50 milliseconds to 1.1 seconds) are available with the 1176LN, precautions must be taken when compressing or limiting low-frequency signals. If the release time is set too short, the compressor action can follow the individual cycles of the signal waveform, causing harmonic distortion. As a guideline when compressing bass guitar or other instruments with a high proportion of very low frequencies, the release time control should not be set clockwise to more than about 11 o'clock. On other material, short release times will not cause harmonic distortion, but they may cause the compressor action to become noticeable due to obvious level changes of the program material and of the background noise.

The attack time is variable from 20 microseconds to 800 microseconds. Short attack times tend to reduce the percussive qualities of a signal, while a long attack time can increase signal punch by, in effect, emphasizing the first portion of the signal (by lowering the gain only on the second portion).

The Teletronix LA-2A (Fig. 5-36A) has only three front-panel controls: the meter selector switch, which selects either gain reduction in dB or signal output with 0 VU corresponding to either +4 or +10 dBm; the peak reduction (gain reduction) control, which determines how much gain reduction will take place; and the overall gain or output level control, which adjusts the output level of the compressed signal. A back-panel switch selects either a compression mode with a 3:1 ratio or a limit mode which provides 10 dB of compression at a 3:1 ratio, followed by 20:1 limiting. This



(A) The Teletronix LA-2A.

(B) The Teletronix LA-3A.

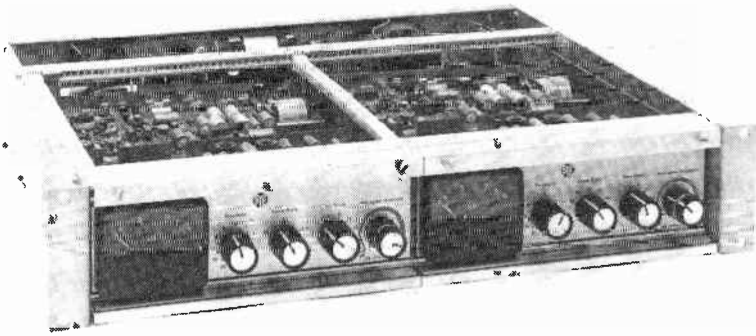
Courtesy United Recording Electronics Industries

Fig. 5-36. Leveling amplifiers.

enables the engineer to limit peaks while compressing a signal only slightly. However, if too much peak reduction is set (more than 10 dB indicated on the meter), limiting may occur all the time, resulting in the program having almost no dynamic range and sounding very dull.

The attack time of this unit is very fast (10 microseconds), and the release time is dependent on the program material so that repeating peaks cause reduction once rather than several times. The gain returns to 50% of normal in 0.06 second and recovers the other 50% in between 0.5 and 5 seconds, depending on the program content.

The Teletronix LA-3A is a transistorized version of the LA-2A and has replaced the LA-2A which is no longer manufactured. The unit has the same operating controls as the LA-2A except that the meter switch has only one output position (+4 dBm equals 0 VU). The LA-3A (Fig. 5-36B) has a rear-panel gain switch which adjusts the gain to either 30 or 50 dB. A compress/limit switch on the rear panel allows the unit to be used as either a compressor only or as a 50:1 limiter. Attack time is between 250 microseconds and 0.5 milliseconds and is determined by the program material. The release time varies from 500 milliseconds to 5.0 seconds depending on the duration of the peak causing the gain reduction. Both the LA-2A and the LA-3A have rear-panel controls which can increase the gain reduction at high frequencies relative to that below 1 kHz, as well as adjustments to equalize the gain reduction of two units interconnected for stereo. Two LA-3As can be mounted side by side in a standard 19-inch rack due to the reduced size of the unit.

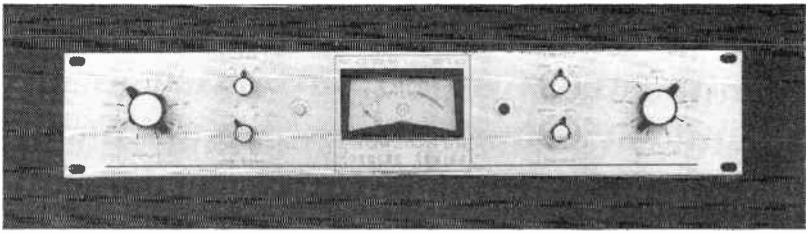


Courtesy Philips Broadcast Equipment Corp.

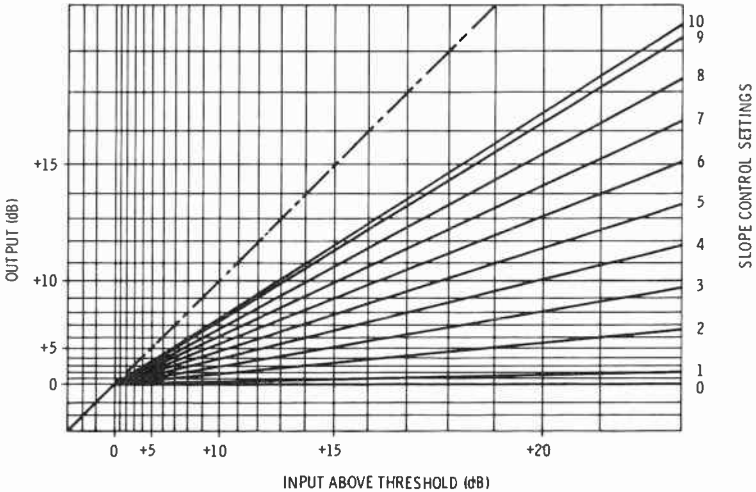
Fig. 5-37. The Pye compression amplifier features a noise gate with threshold adjustable from −20 to −50 dB relative to the compression threshold.

The Pye compressor (Fig. 5-37) has compression ratios of 2:1, 3:1, 5:1, and a limiting ratio of more than 20:1. The attack time is fixed at less than 0.5 millisecond for compression and 1 millisecond for limiting, while the release times are variable in steps from 100 milliseconds to 3.2 seconds. The amount of gain reduction is set by varying the threshold of compression. By placing the threshold below the input signal level, gain reduction is achieved and the amount is indicated directly on the meter. The Pye meter reads differently from those on the LA-2A and the 1176LN, in that it rests at the left-hand side and moves to the right to indicate gain reduction, while the others rest at 0 VU and move left to indicate gain reduction. The Pye unit has no output level control, and the meter cannot be switched to read the signal output.

The Spectra Sonics Model 610 (Fig. 5-38A) has compression/limiting ratios (slopes) continuously variable from 1.1:1 to 100:1. Its attack time varies automatically between 100 nanoseconds and



(A) Model 610.

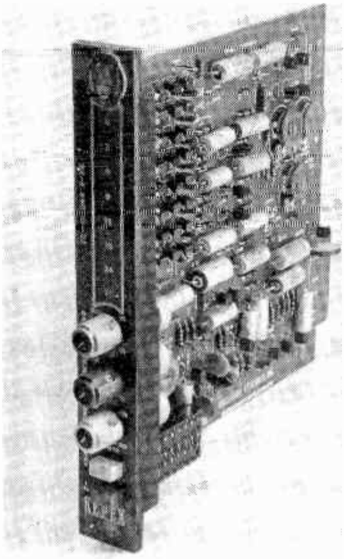


(B) Compression curves.

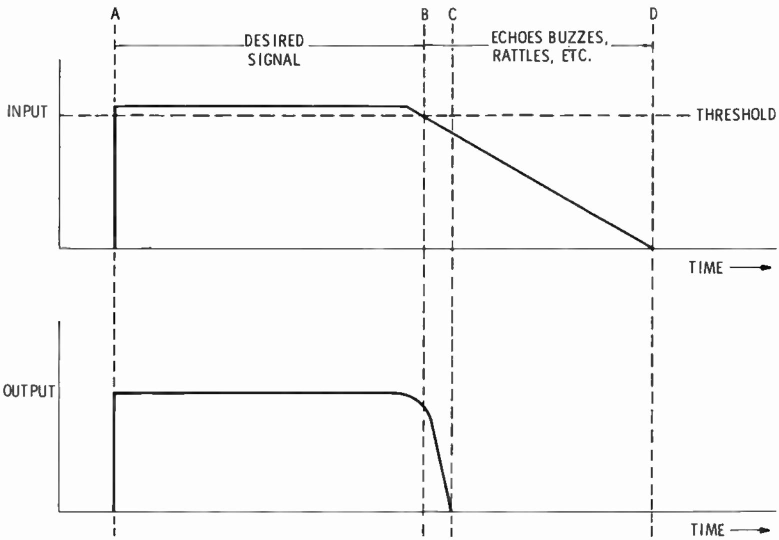
Courtesy Spectra Sonics

Fig. 5-38. The Spectra Sonics 610 Complimiter and compression curves.

2 microseconds for limiting and between 100 nanoseconds and 1.2 milliseconds for compression, depending on the program. The release time for the limiter section is very fast, being 90 nanoseconds for 90% recovery, while the time for 90% recovery in the compression section is continuously variable from 50 milliseconds to 10 seconds. The threshold level is the same for all settings of the slope control, eliminating the need to reset the input level control if the compression/limiting ratio is varied. A white lamp flashes to indicate when the peaks of the input signal pass the threshold of limiting, while a VU meter normally resting at 0 falls to the left to indicate the gain reduction accompanying compression. If peak limiting with no compression is desired, the input level control is advanced until the white threshold attack lamp flashes but no gain reduction is indicated by the meter. An output level control and a meter switch to calibrate the meter to +4 or +8 dBm are provided



(A) Kepex expander.



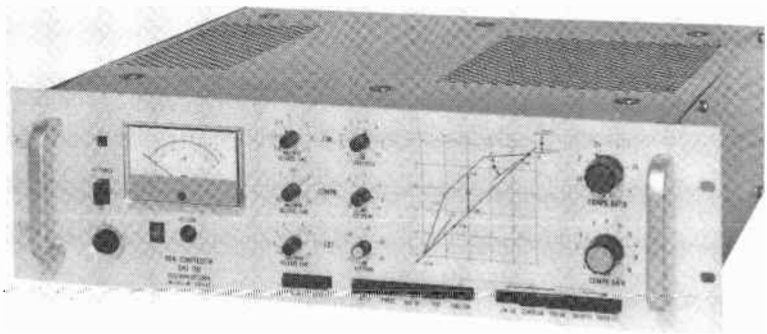
(B) Using the Kepex to remove echoes, buzzes, and rattles from a signal.

Courtesy Allison Research, Inc.

Fig. 5-39. The Allison Research Kepex and curves.

as well as a red overload lamp which indicates to the engineer that signal levels within the unit are approaching the clipping level and should be reduced by adjustment of either the input or the output level controls, or both.

One example of an expander is the Kepex (Fig. 5-39A). It acts somewhat like a compressor in reverse. With no input signal, gain reduction is at the maximum preset amount (variable from 0 to 60 dB). When the input signal level exceeds the threshold, gain is increased to unity. The Kepex has a fixed attack time of less than 20 microseconds and a variable release time from 50 milliseconds to 6 seconds. The threshold level and amount of expansion are both variable, and the expansion ratio varies from 1:2 to 1:4, depending on the setting of the expansion control. Kepex can tighten up signals by removing echos and rings from them by reducing the gain after the major portion of the signal is over (Fig. 5-39B). It has the additional capability to *key* or initiate the expansion of one signal on the basis of a second signal for special effects. Peak values of gain reduction are indicated by lights behind the front-panel scale. Threshold-type expanders are also called *noise gates* or simply *gates* because they can be set to permit a signal to pass and then shut down the channel when no signal is present to keep out noise.



Courtesy Gotham Audio Corp.

Fig. 5-40. The EMT 156 PDM.

Another gain-control device is the EMT 156 shown in Fig. 5-40. This is a combination compressor/limiter with separate thresholds and release times for each function. The release times can be controlled either manually or automatically. In the automatic mode, the release times are dependent on the program material. The compressor release times are determined by the peak-to-average ratio of waveform value. High ratios such as those produced by speech require short release times, while lower ratios require longer times. Limiter release time is determined by the amplitude and duration of the peaks as well as by the number of peaks and is adjusted to prevent the level of the program from noticeably decreasing after each peak due to excessively long release times.

The EMT 156 has no threshold in the compression mode. The compression *rotation point* is adjustable from −6 to −1.5 dB with respect to an internal reference level of 0 VU. The rotation point for the compressor is the input level which produces unity gain. If the input signal exceeds the rotation point, the compressor gain falls below unity, while if the input signal falls below the rotation point, the compressor gain rises above unity. The *compressor gain* control sets the limit on how high the gain can rise when the input signal falls below the rotation point, and it is continuously variable from 0 to +18 dB. The slope of the curve can be varied from 1.5:1 to 4:1 with an attack time preset to 2.5 milliseconds, but internally variable between 1 and 4 milliseconds. The limiter threshold is variable from −2 to +7.5 dB, and the limiter does not permit the output of the unit to exceed the threshold at any time. This is achieved by beginning the limiting 2 dB below the threshold. While the EMT 156 is not an expander like the Kepex, it has an expansion function which can reduce the compression gain to prevent the increase of background noise levels when only very low-level signals are present. The expander rotation point is variable from −55 to −35 dB, and its slope is switch selectable to either 1:1.5 or 1:2.5. As the input signal rises above the expander rotation point, the gain gradually increases up to the value set by the compressor gain control. Below the rotation point, the gain falls below unity. Individual compressor, limiter, and expander on/off switches are provided to enable the use of any combination of the three functions, and a switch is also included to bypass the unit entirely.

The EMT 156 is a two-channel unit interlocked for stereo. The control settings affect both channels equally, and the gain is controlled by the channel with the highest peak amplitude. The unit provides a meter which can be set to read the instantaneous gain in use by the compressor or limiter. The meter can also be switched to indicate the number of attacks per second in the limiter section. The engineer can thus determine how frequently limiting is occurring and can adjust the gain and threshold controls to achieve the desired amount of limiting.

Gain reduction can be achieved in several ways. The LA-2A and LA-3A use an *electroluminescent attenuator* which is a phosphorescent panel that glows in proportion to the input signal fed to it. This glow illuminates two photocells (one a cadmium sulfide cell and the

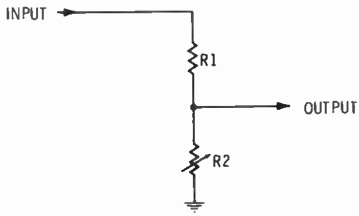


Fig. 5-41. A gain-reduction circuit using an FET or an LDR (light-dependant resistor), such as R2.

other a cadmium selenide cell) which are part of a volume control circuit. The value of the attenuator changes with the illumination received by the phtotocells, and thus turns the volume of the signal up or down. The Kepex and the 1176LN use a *field-effect transistor* (*FET*) as a *voltage-variable resistor*. The FET changes resistance in proportion to a voltage (obtained from the input signal) applied to it, and this resistance change is used to control the signal in the same

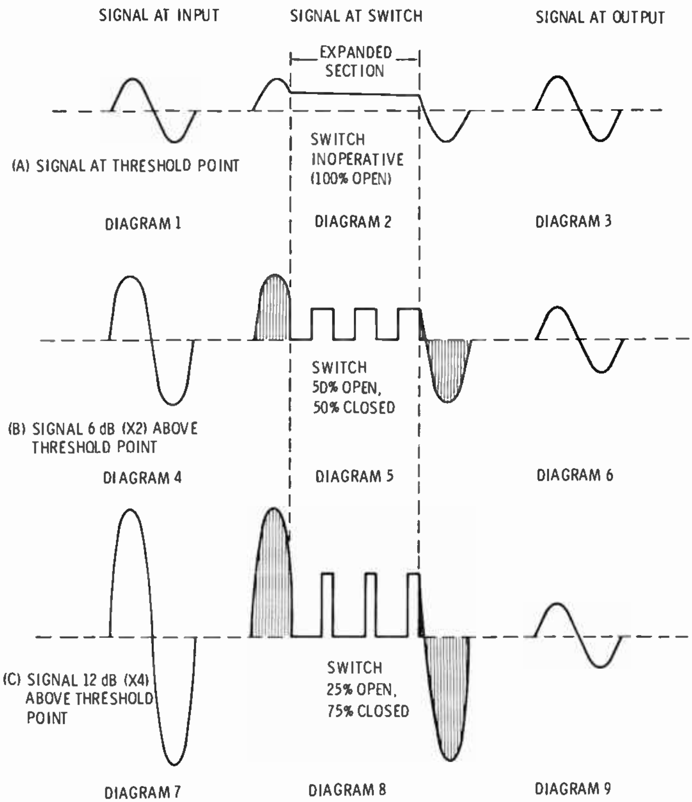


Fig. 5-42. Operation of the PDM system.

manner as the photocells (Fig. 5-41). The Pye and EMT compressors use a different method of gain control called *pulse-duration modulation* (*pdm*) which feeds the signal through an electronic switch that is opened and closed at a rate of 200 kHz (Fig. 5-42). By varying the time the switch is left open or closed during each cycle, energy is removed from the signal without affecting the program, and the result is an output signal of lower level than the input signal.

SPECIAL EFFECTS

In addition to EQ, compression, and expansion (especially the expansion of one signal triggered by another signal, as is possible with Kepex), there are many other devices or processes which can be used to make the original signal sound like something other than what it really is, adding interest to a song. The signal processed in such a manner is called an effect. The five most common effects are created through the use of: reverberation, time delay, variable-speed tape recorders, phasing, and backward signals.

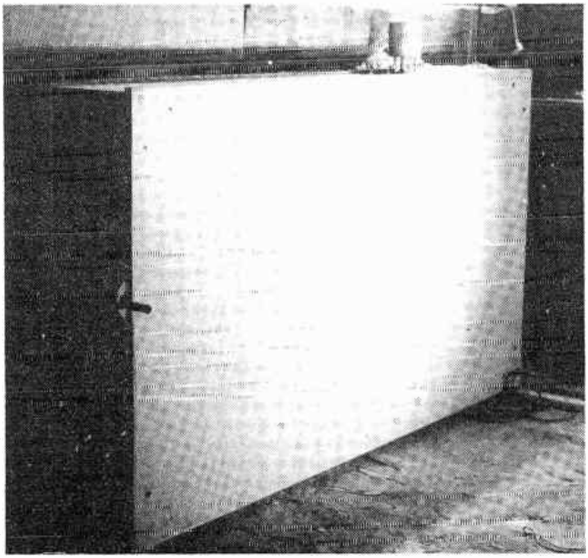
REVERBERATION DEVICES

There are three main types of reverberation devices: (1) echo plates, such as those made by EMT; (2) spring reverb units; and (3) acoustic echo chambers.

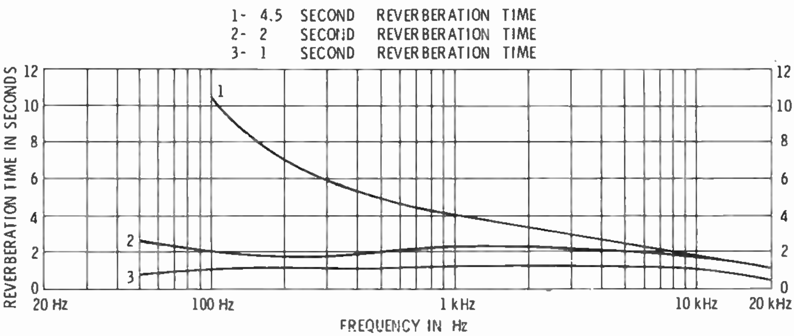
Echo Plates

The EMT 140 TS echo plate is a rectangular steel sheet $\frac{1}{64}$ inch thick and about 3 feet high by 6 feet wide, under tension and suspended in an enclosed frame approximately 8 feet by 4 feet by 1 foot (Fig. 5-43A). Reverberation is created by inducing wave motion in the plate. These waves reach the edges of the plate where they are reflected back across the plate to the other edges. This process continues until the heat created through the bending of the plate dissipates the wave energy. The waves are induced in the plate by a moving-coil driver which converts the electrical input signals into mechanical motion. The reflected waves are picked up by contact microphones which sense the varying pressure of the steel plate against them as the waves pass the points at which the microphones are mounted (Fig. 5-43B shows reverb time versus frequency for various decay-time settings) and the waves are converted back into electrical signals (Fig. 5-44).

The EMT is available in both a mono and a stereo version. The mono version uses a single driver and a single pickup mike, while the stereo version uses a single driver and two pickup mikes. Thus,



(A) The EMT 140 TS echo plate.



(B) Reverb time vs frequency.

Fig. 5-43. Echo plates.

in the stereo version all signals to be reverberated are either combined before being sent to the EMT and fed into one of the two inputs of the EMT (Fig. 5-45A), or the signals are divided between the two inputs of the EMT and are combined in the EMT amplifier (Fig. 5-45B). In both cases, the driver receives a mono signal, and it is the presence of two pickup mikes that produces the stereo effect. The stereo output is achieved by spacing the pickup mikes at different distances from the driver and on opposite sides of it. As a result of this spacing, not only does the wave traveling direct

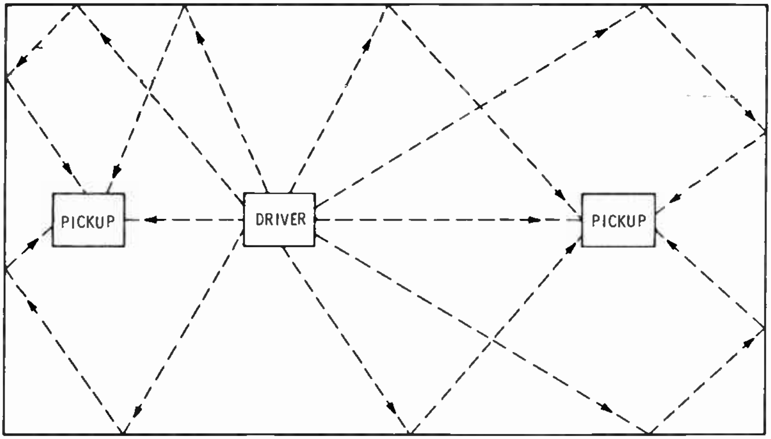


Fig. 5-44. The waves induced in the plate by the driver are reflected at the edges and converted into electrical signals by the pickups.

from driver to pickup take longer to reach one pickup than the other, but the reflected waves picked up by each driver at any time will have arrived there via different path lengths and will therefore be more or less out of phase with each other.

If we feed a dry signal equally to both speakers of a stereo system and add a reverberated signal which is 0° out of phase with the dry signal to the right speaker and a reverb signal which is 90° out of phase with the dry one to the left speaker, the reverb appears to

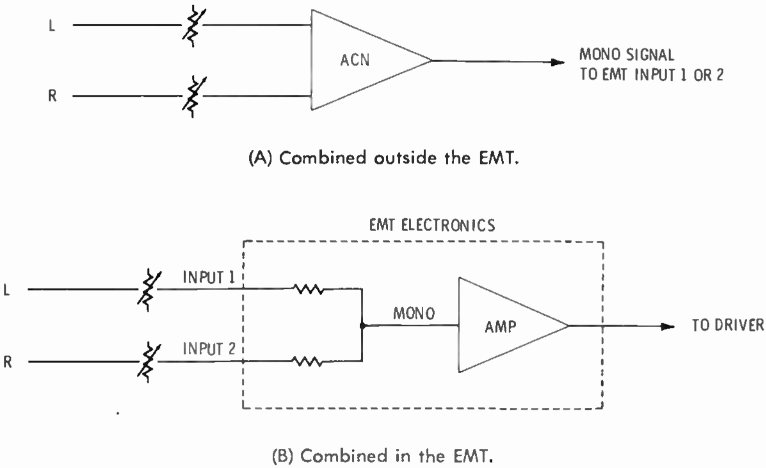


Fig. 5-45. Two methods of obtaining a mono signal to feed the EMT plate driver.

come from the right (Fig. 5-46A). If the left speaker reverb were 0° out of phase with the dry signal and the right channel reverb were 90° out of phase, the reverb would appear to come from the left (Fig. 5-46B). If the left and right reverb signals were in phase with each other but 90° out of phase with the dry signal, the reverb would appear to come from between the two speakers (Fig. 5-46C). If the left reverb were +90° out of phase with the dry signal and the right reverb were −90° out of phase with the dry signal, the echo appears to come from both the left and right side and not from the middle (Fig. 5-46D). In use, the phase relationships between the two pickup mikes and the dry signal are constantly changing when program material is sent to the EMT, since more than one wave reflection reaches a pickup at any time .Therefore, the reverb signal appears to come from all locations in and between the left and right speakers (or outputs of the EMT).

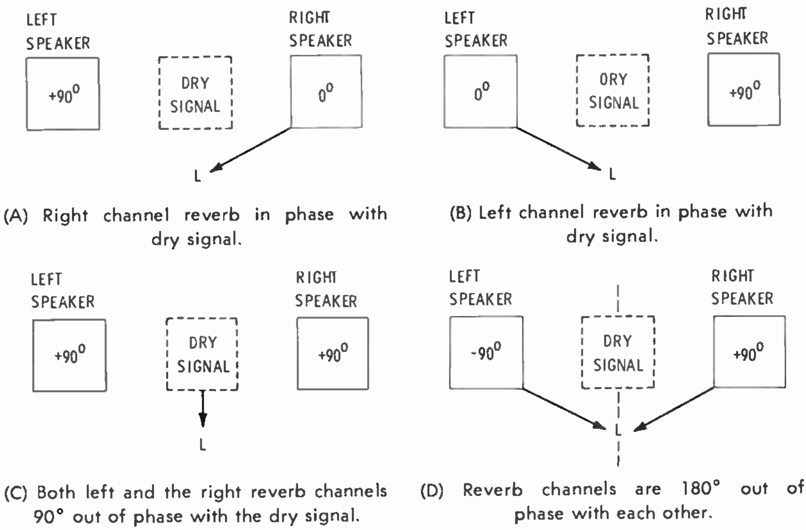


Fig. 5-46. The random phase relationship of the signals at the EMT plate outputs make the reverb appear to come from all directions.

The decay time of the EMT 140 TS is variable from 1 to approximately 4 seconds. It is changed by moving a plate covered with acoustic absorbing material closer to the steel plate to shorten decay time and farther from the plate to lengthen it. The motion of the damper plate is controlled either by a handwheel on the unit or remotely by a motor mounted on the unit which is operated by two push buttons on the console. The decay time is indicated by a meter just above the push buttons. One push button lengthens the decay time while the other shortens it. The meter is calibrated from 0 to 10, with each unit equal to ½ of a second (Fig. 5-47). The absorptive plate decreases the decay time by damping the vibration of the air molecules set in motion by the steel plate, which in turn damps the vibration of the plate.

Since operation of the handwheel or motor introduces some noise into the reverb output, the decay time should not be varied during the program. In addition, any movement of the steel plate produces an output signal from the unit, so it must be kept in a room where it is not likely to be bumped into or vibrated by sound waves. Usually several units are placed in the same room because they do not interact with each other. The units are fairly fragile and should not be transported or moved after they have been set up, or the plate may be damaged.

The input or output of the EMT plate can be EQed for special reverberation effects. A 700-Hz, high-pass filter connected at the output of each EMT channel is often used to increase the brightness of the echo and to reduce the low-frequency reverberation time which otherwise would increase more rapidly than the high-frequency time as overall decay time is increased. Although increased reverb time at low frequencies is typical of large halls, the persistence of too much low-frequency sound can make pop music sound boomy or overly full and unclear, as well as causing the VU meter to read considerably higher than the apparent loudness of a program. Reducing the low-frequency output of the plate enables the engineer to increase the overall volume without overloading the tape.

Recently, the EMT 240 reverb unit has been introduced, using a 12-inch-square piece of gold foil instead of the steel plate (Fig.

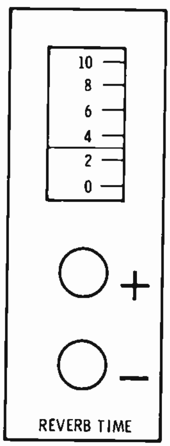
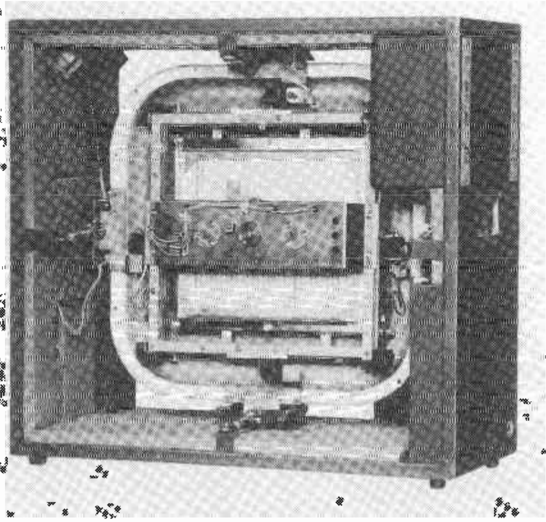


Fig. 5-47. The EMT 140R remote decay time controller.



Courtesy Gotham Audio Corp.

Fig. 5-48. The EMT 240 reverb foil with protective covers removed.

5-48). This unit has many advantages over the Model 140 TS: it is only 1/5 the size, approximately 2 feet by 2 feet by one foot; it is less sensitive to physical shock and requires no retensioning or recalibration after being moved, making it practical for location recording (although it does weigh 132 pounds); it is less sensitive to ambient noise, requiring an environmental noise level of less than 80-dB spl as compared to the Model 140 TS requirement of less than 50-dB spl; its first reflection is delayed three times longer than that of the Model 140 TS, giving a more realistic character to the reverberation; and the reflections are twice as densely spaced so that frequency coloration of the reverb is reduced. The decay time is variable from 1 to 4 seconds by adjusting the proximity of an absorptive plate to the foil and can be remotely controlled as with the Model 140 TS.

Spring Reverb Units

A second type of reverberation device is the spring reverb unit. One design utilizes springs which are connected to rotating magnetic rods on each end. The rods on one end are subjected to a magnetic field created by a coil through which the signal to be reverberated is flowing. The magnetic field causes the rods to rotate and create a wave in the form of a twisting motion in the springs. The propagation velocity of the wave in the springs is much lower than the speed of sound in air and varies with the thickness of the

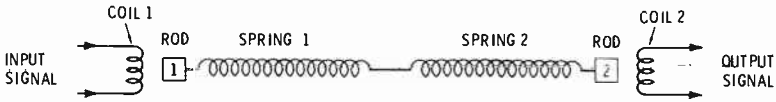


Fig. 5-49. A spring reverb unit.

spring wire, the number of turns to the inch, and the length of the spring. The spring length in these units is usually about one foot per set.

In practice, several sets of springs are used, and each set consists of at least two springs connected in series. The series springs are chosen to have different wire sizes and different numbers of turns to the inch. When the wave reaches the far side of the springs, it causes another set of magnetic rods to rotate and induce a current flow in a coil which converts the rotating mechanical motion into electrical current which is then amplified and becomes the reverb signal (Fig. 5-49). The signal does not just pass through the springs once, however. At each junction (rod to spring, spring to spring, and spring to rod), part of the signal is reflected back through the spring it just flowed through, so that it takes longer for certain parts of the input signal energy to reach the output of the springs. The springs dissipate energy in the form of heat, so that the waves which have followed longer paths are lower in level than the other waves by the time they reach the output. Since the springs are different, they create different delay times, so the net effect is that of many reflections coming shortly after the input signal, closely spaced but decreasing in both the time between reflections and in level as time passes (Fig. 5-50).

The decay time of this type of spring unit can be decreased by bringing some acoustic absorbing material such as fiberglass near the springs to absorb the energy of the air molecules around the springs and damp out their rotating motion. Several examples of this type of reverb unit are the RCL Electronic, the Fisher Space

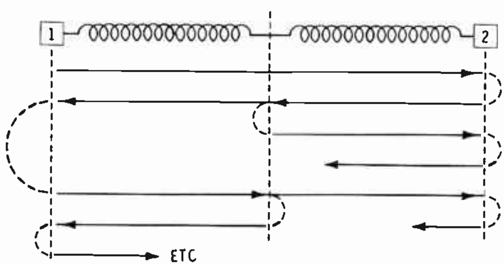


Fig. 5-50. The signal path in a spring reverb unit. A reflection occurs at each boundary: spring to spring and spring to rod.

Expander, and the Hammond spring reverb unit used in most electric guitar amplifiers.

A second type of spring reverb uses springs hanging vertically and connected to a moving coil driver. The springs are hung so that the bottom of one of them sits in a tube filled with oil, which damps its oscillations. Another of the springs is connected to a transducer that converts the mechanical motion of the spring into electrical energy. Instead of twisting the springs, this unit alternately stretches and compresses them (Fig. 5-51). The time delays are created by

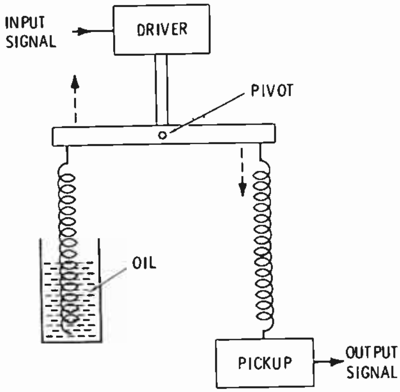


Fig. 5-51. A simplified diagram of a spring reverb unit using oil to damp the vibrations.

the propagation velocity of the waves in the springs and by the multiple reflections that occur both when the waves hit the oil and when a wave reaches the junction of two springs. The height of the oil in the tube determines the decay time of the unit. As the height of the oil rises, damping increases and the decay time decreases. The decay time in this type of unit usually remains fixed because draining and refilling the tube with oil is a messy job.

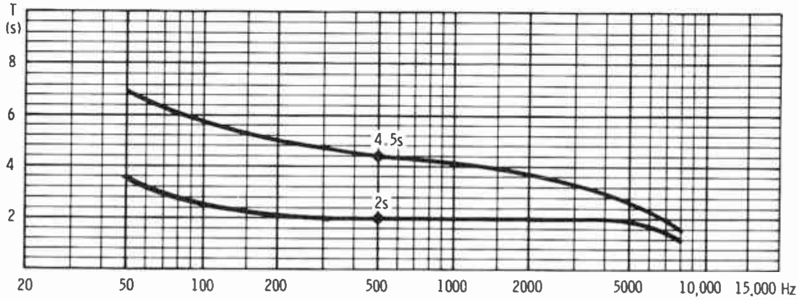
Most spring reverb units suffer from the problem of sounding tinny, unnatural, and in general very reminiscent of springs. Different designs have attempted to eliminate the problems through carefully controlled spring lengths and equalization to achieve smooth frequency response. Regardless of what is done, most of them still have not changed in type of sound. These units do have the advantage of small size and low cost, making them attractive for small studios or as a secondary echo chamber to be used only on instruments where their sound quality will not be objectionable.

AKG has overcome the problems of spring-type echo chambers in their Model BX 20E through the use of specially designed spring and damping elements (Fig. 5-52). The units consists of two identical reverberation channels which can be used either independently as two mono chambers (separation between them is better than

Courtesy North American Philips Corp.

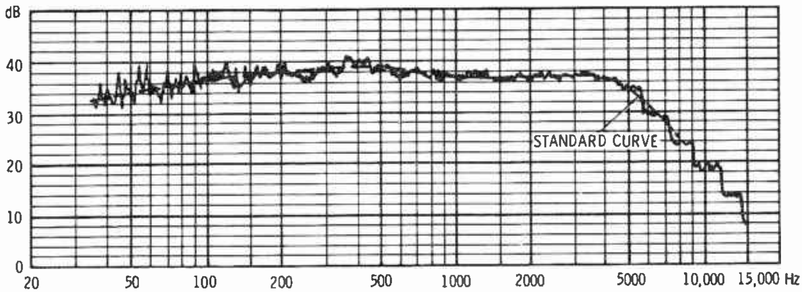
Fig. 5-52. The AKG BX20E reverberation unit (interior view).

60 dB) or used with their inputs in parallel to create stereo reverb. The decay time is variable from 2 to 4.5 seconds (frequency-response curves are shown in Figs. 5-53 and 5-54) and is controlled by electronic damping which can be adjusted remotely and separately by two dc voltages (Fig. 5-55). Since the decay time is varied electronically, it can be changed during a program without causing any noise. The unit is insensitive to physical shock, enabling it to be moved for location recording without excessive precautions to pre-



Courtesy North American Philips Corp.

Fig. 5-53. Frequency response of decay time for BX20E.

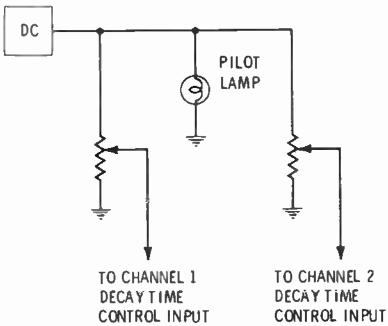


Courtesy North American Philips Corp.

Fig. 5-54. BX20E frequency response measured with ⅓-octave noise.

vent damage to the springs. It is well isolated from ambient noise and, if desired, may be left in the control room; acoustical feedback will not occur until the sound level of its output through the monitor speakers is greater than 100-dB spl near the unit. The realism of the BX 20E reverb is enhanced by its delay of the onset of the reverb by 20 milliseconds for high frequencies, with the delay increasing to 50 milliseconds for low frequencies. The unit is approximately 1.5 feet by 1.5 feet by 4 feet in size and weighs 110 pounds.

Fig. 5-55. The remote decay time-control circuit for the BX20E.



Acoustic Echo Chamber

A third type of reverb device is the acoustic echo chamber. This consists of a room with highly reflective surfaces, in which a speaker and a microphone are placed. The speaker is fed the signal to be reverberated and the mike picks up a combination of direct sound from the speaker and reflections off the walls, ceiling, and floor. By using a directional mike and by placing the mike and speaker back to back pointing in opposite directions, the pickup of direct sound can be minimized. Movable partitions can be set up in these rooms to vary the decay time (Fig. 5-56). Acoustic echo chambers have the most pleasing and natural quality of all artificial reverb units if properly designed. Their disadvantages are that they use a lot of space (typically 18 by 15 by 12 feet), require as much isolation from external sounds as a studio, and need high-quality speakers and mikes. This makes the price of an acoustic chamber very high. Smaller rooms can be used, but the bass response of the chamber would suffer. Parallel walls are avoided in the construction of the chamber to prevent the generation of standing waves. Acoustic chambers can be built using large oil tanks buried in the ground, but these tend to have a tinny quality and need equalization to produce acceptable sound.

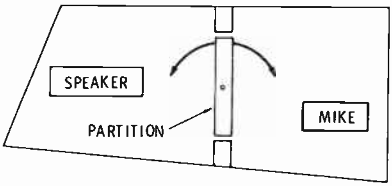


Fig. 5-56. An acoustic echo chamber. Decay time is a function of the partition position.

Distant miking can also be used in the studio to pick up the reflected sound of instruments to be mixed in with their direct sound. The studio reverb time is usually short, and the room sound picked up by the distant mikes is very mellow, giving an instrument a very round sound.

With most reverberation devices, a certain input level will optimize performance with respect to signal-to-noise ratio and distortion. The input to the device should be monitored with a VU meter which is calibrated to read zero for optimum input level. The overall level of the mix of the signals to be reverberated should be set for 0 VU on the meter and the output of the device mixed in with the dry signal so that the ratio of direct to reverberant sound can be controlled. The decay time will determine the liveness of the room in which the performance appears to be taking place, while the ratio of direct to reverberant sound determines whether the signal seems to be coming from the front or the back of the room.

DISCRETE REPEAT

The next category of effect devices consists of those which produce discrete repeats of the input signal at certain regularly spaced intervals. There are three ways of achieving this effect: using a tape recorder, using a digital delay line, and using an acoustic delay line.

With Tape Recorder

In order to produce a discrete repeat, a tape recorder must have separate record and playback heads. The signal is recorded on the

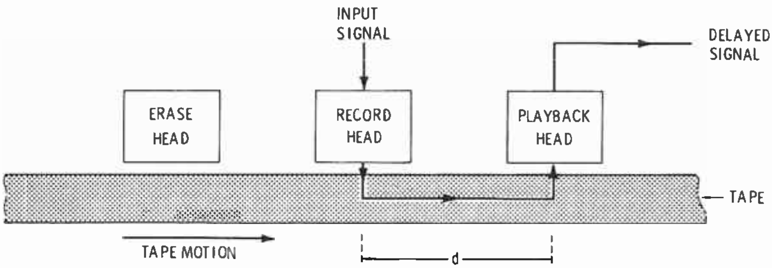


Fig. 5-57. Delaying a signal with a tape recorder.

tape and played back after a short delay. The delay is equal to the distance between the record and playback head divided by the tape speed. The playback head signal is a single repeat of what is fed to the tape recorder input (Fig. 5-57). Multiple repeats can be obtained by returning a part of the playback head output to the input of the recorder so that the first repeat is recorded again, delayed, and played back again (Fig. 5-58). Variable resistor R controls the number of repeats. The number of repeats that occur depends on the amount of the playback signal fed back to the recorder input. When many repeats are used, the signal gradually deteriorates into noise because each successive record and playback cycle adds a little distortion and tape noise. By returning too much playback signal to the input, a feedback type of oscillation is generated. The repeats differ from those of reverberation, regardless of how closely they are spaced, because the time intervals between them are all the same; while with an acoustic chamber, plate, or spring device, there are many different intervals between reflections.

The time between repeats can be varied by either changing the spacing between the record and playback heads or by varying the tape speed. In practice, varying the tape speed is used because the heads are not movable on most recorders. Some recorders, however,

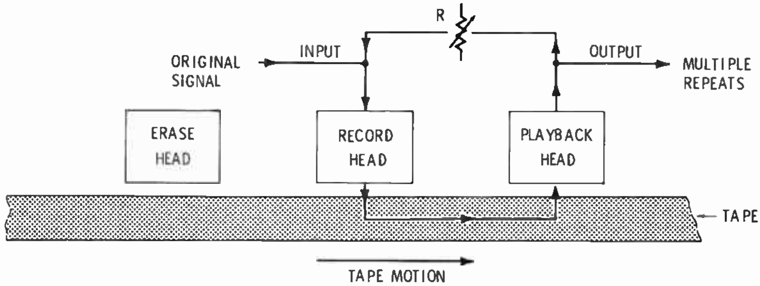


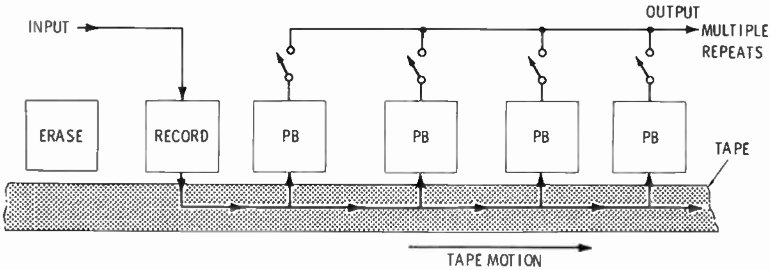
Fig. 5-58. Producing multiple repeats of a signal with a single playback head tape recorder.

are built especially for producing this type of *tape delay* and have movable or multiple playback heads so that different intervals between repeats can be obtained (Fig. 5-59).

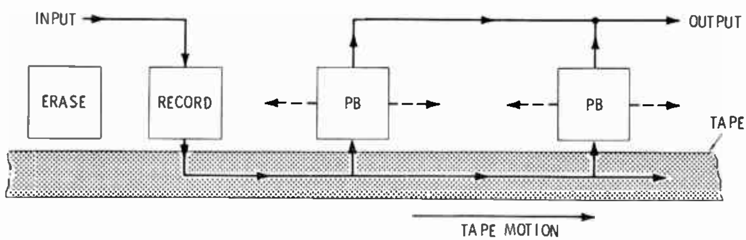
Digital Delay

The digital type of delay unit produces exactly the same effect as a tape recorder used for delaying the signal, but it makes the process considerably easier. The tape recorder requires the use of magnetic tape either on reels or spliced into a continuous loop. If on reels, the tape must often be rewound or turned over because it runs out frequently. This can be especially annoying if very high tape speeds are used to obtain short intervals between repeats. A continuous loop eliminates the problem of tape run out, but the short length of tape used in the loop passes over the heads so often that the oxide coating wears off quickly. This results in the need for frequent loop changes as well as frequent cleaning of the heads to remove the particles of oxide which accumulate on them. This problem also becomes more pronounced as the tape speed is increased. The digital unit eliminates these problems by eliminating the use of magnetic tape. In addition, the digital process has no moving parts to require lubrication and frees the recorder previously used for tape delay for other uses.

The digital delay unit encodes the analog audio signal into digital form and feeds it into a *shift register.* A high-frequency oscillator

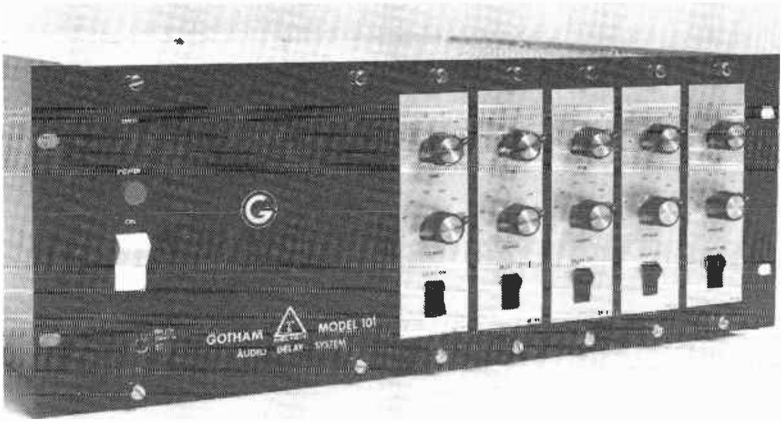


(A) By connecting different combination of playback heads.



(B) By changing the playback head locations.

Fig. 5-59. Producing multiple repeats and variable delays with a tape recorder.



Courtesy Gotham Audio Corp.

Fig. 5-60. The Gotham Delta-T Model 101 Audio Delay System.

feeds *clock pulses* to the shift register, causing the digital signal to move one step through the register for each pulse. The pulse frequency and the length of the shift register (the number of discrete positions between its start and its end) determine the delay time. At the output of the shift register, the signal is converted back into analog form. Multiple outputs with independently variable delay times are possible using a single shift register by tapping the register at many different points, and making the points available through a selector switch. A separate digital-to-analog converter and output amplifier is needed for each output.

The Delta-T 101 Audio Delay System available from Gotham Audio provides as many as five discrete repeats with delay times variable from 5 milliseconds up to 320 milliseconds in 5-millisecond increments for each repeat (Fig. 5-60). The Eventide Clock Works

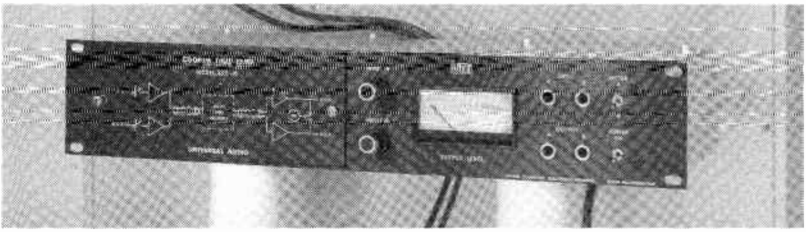
Courtesy Eventide Clock Works, Inc.

Fig. 5-61. The Eventide Clock Works DDL 1745 digital delay line.

Model DDL 1745 Digital Delay Line operates similarly except that its delay lines are available with two-millisecond step increments, and it is expandable to a total of 800 milliseconds of delay with an unlimited number of outputs (Fig. 5-61).

Acoustical Delay Line

The third method of producing time-delay signals is the acoustical delay line. Sound is fed through a long coiled tube or pipe and picked up at the far end by a microphone. The delay time is controlled by the length of the tube. Although this method is not new, it has always suffered from severe peaks and dips in frequency response.

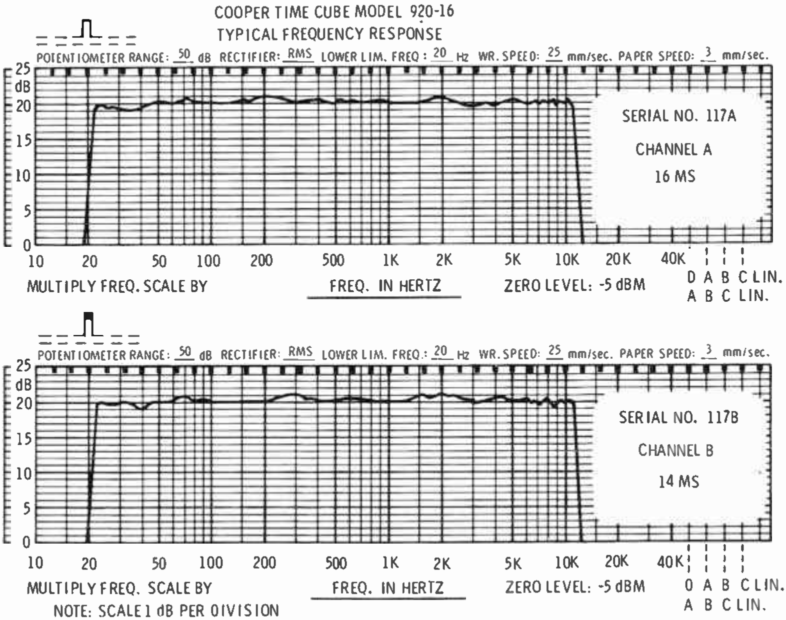


Courtesy United Recording Electronics Industries

Fig. 5-62. Cooper Time Cube electronics unit with the delay lines in the box behind it.

The Cooper Time Cube shown in Fig. 5-62 produced by UREI has overcome this problem and provides two delay channels with 40-dB isolation between them. The delay available is fixed at 16 milliseconds for one channel and 14 milliseconds for the other, with a 30-millisecond delay possible by connecting them in series. The Time Cube consists of a 2 feet by 2 feet by 9 inches wooden case which contains the delay lines and transducers, and a rack-mounted electronics unit which contains the driving and pickup amplifiers, equalization, and power supply. The delay line is relatively insensitive to ambient noise (an 80-dB spl signal at the outside of the wooden case produces no more than a −60-dBm output) and can be connected at a distance from the electronics by standard mike cables. Each electronics unit is factory equalized and calibrated to its delay lines. Fig. 5-63 shows typical frequency curves for the Cooper Time Cube.

The use of delayed signals is the same regardless of how they are obtained. The perception of the ear of more than one instrument playing at a time depends on the lack of exact synchronization between the instruments. No matter how good the musicians are, the fact that each instrument is a different distance from the listener ensures a lack of synchronization at the listener's ear due to the different times necessary for the waves to reach the listener. By repeating a single signal with a slight time delay, the apparent number of instruments playing is increased. This process is called *electronic doubling* or *automatic double tracking* (ADT). Often *doubling* and *tripling* is done by the artists themselves through multiple overdubs, so that one violin can sound like an ensemble or so that two vocalists sound like a chorus. Doubling can also be used to strengthen weak vocals because slight irregularities in one performance are hidden by the other performance. If the delay time is



Courtesy United Recording Electronics Industries

Fig. 5-63. Cooper Time Cube frequency-response curves.

long enough that the repeat is heard discretely (more than about 35 milliseconds), the repeat is often called *slap echo* or *slap back* and has the effect of causing the rhythm to bounce or double.

Another important use of delayed signals is in conjunction with reverb units. The ear judges the size of a room by the time interval between hearing the direct sound and hearing the first reflection from the surrounding surfaces. The longer this interval, the larger the room appears to be. Feeding a signal to a delay unit and then feeding the delayed signal to a reverb unit gives the engineer control of the first reflection and controls the apparent size of the room. This is called *delaying the echo send signal.*

VARIABLE TAPE SPEED

Although constant speed is required of tape machines for standard recording and playback, continuously variable tape speed can provide several interesting effects. When used for tape delay echo, a variable-speed machine can produce varying delay times. The pitch or key of a song can be raised or lowered to accommodate an instrument or voice which cannot perform in the original key of the song due to limited range or tuning to other than concert pitch but can also play its part if the key is changed during the overdub session. After the overdubs are done, the tape can be played back at normal speed and the overdubbed instruments will appear as though they were played or sung in the key of the song. Finally, a song which is too slow or too fast can have its tempo changed.

There are limits on these methods of changing tempo and key, however. As the tempo is increased, the pitch also changes, and instruments can sound unnatural if their pitch is changed too much. This results from the timbre of the instrument remaining the same while the pitch of the fundamental changes. When instruments change pitch in performance, the timbre changes too. A large pitch change without a timbre change sounds artificial.

METHODS OF VARYING TAPE SPEED

There are three methods of varying tape speed. One uses a standard tape recorder with a synchronous capstan motor (the speed of this motor depends on the frequency of the ac power driving it), a second uses a *servo motor* which rotates the capstan at a speed proportional to a dc voltage applied to it, and the third device uses a combination of rotating heads and a variable-speed capstan motor.

Synchronous Motor

The first method uses a standard synchronous capstan motor such as found in the Ampex AG440 and applies 110 volts ac to it at varying frequencies. The synchronous recorder runs at 15 ips only when the power-line frequency is 60 Hz. If the frequency is raised to 65 Hz, the tape speed becomes greater than 15 ips, and if the power-line frequency is lowered to 55 Hz, the tape speed becomes lower than 15 ips. The power frequency is made variable through the use of a *variable-frequency oscillator* (*vfo*) and a power amplifier, such as a MacIntosh 275 or Bogen M100, that can put out 110 volts ac at any audio frequency. The combination of the vfo and the power amplifier is called a *variable-speed oscillator* (*vso*) (Fig. 5-64). The standard power line (to the motor only, not to the electronics of the recorder) is replaced by the vso through the use of a switch or by the removal of a dummy plug on the tape transport and plugging the vso in through this socket. The speed of the tape is then varied by varying the oscillator frequency.

The voltage output of the power amplifier must be monitored with a meter so that neither too much nor too little voltage is applied to the motor. The voltage tolerance is from about 100 to 120 volts. Since the motor is designed to operate at 60 Hz, there is a limit to how much the applied frequency can be varied. Normal limits are between approximately 25 and 100 Hz. The motor may tend to stall

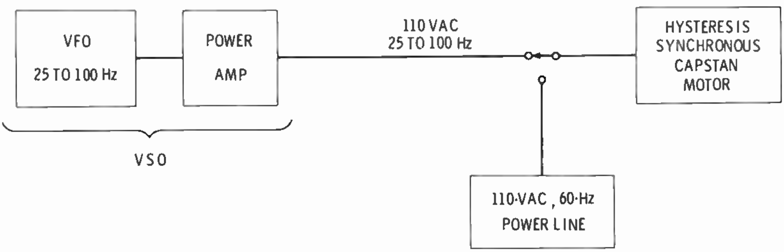


Fig. 5-64. Varying the speed of a hystersis synchronous motor.

or overheat if used for long periods of time outside this range. The range of speeds attainable can be increased by using the high- and low-speed controls on the recorder in conjunction with the vso.

Servo Motor

The servo motor capstan system uses a dc motor which changes speed with the applied dc voltage. Normally, a tachometer (tach) pickup is mounted next to a notched wheel on the capstan assembly and counts the number of notches that pass it per second. This frequency is compared with the frequency (determined by system design) which results when the capstan rotates at the correct speed and causes an increase or decrease in the dc voltage applied to the motor as necessary to produce the proper frequency output from the pickup. Variable speed can be achieved by varying the frequency that the speed-control circuit wants to see at the output of the tach pickup or by disconnecting the speed-control circuit and directly varying the dc voltage applied to the motor. The first of these methods provides more stability but requires more circuitry. The second method is simpler to achieve, but requires a well-regulated dc power supply to prevent speed fluctuations. A servo motor capstan can be made to move tape at speeds from 0 to 60 ips or higher, and therefore it provides much more flexibility than the vso system.

The newer Ampex MM1000s are supplied with a servo motor capstan and have built-in plus and minus half-tone speed change push buttons as well as continuously variable speed. Many other recorders such as those made by Revox, MCI, 3M, and Studer have this variable-speed capability; they also use servo motor capstans.

Rotating Playback Head

The third method of achieving the effect of variable tape speed uses a device that may be attached to any standard ¼-inch tape recorder. The unit has the ability to vary tempo without varying pitch or to vary pitch without varying tempo. These effects are achieved through the use of a rotating playback head which consists of four parallel playback gaps mounted on a cylinder and angled 90° apart. The tape is wrapped around this head in such a manner that one gap is always in contact with the tape, ensuring continuous reproduction regardless of the head's rotational position. Since the pitch of the reproduced signal depends on the speed of the tape with respect to the playback gap, rotating the head opposite to the direction of tape travel increases the relative speed without affecting the playing time of the signal, so the pitch rises while the tempo remains the same. If the head is rotated in the direction of tape travel, the tape-to-gap speed decreases and the pitch is lowered. If the playback head is stationary, the pitch is normal. To effect a tempo change without a pitch change, the reel-to-reel tape speed is varied via the capstan motor to achieve the desired tempo, and then the playback head rotational speed is adjusted to bring the pitch back to normal.

The actual lengthening of a program comes about by the head repeating certain sections of the tape, while the shortening of a program is achieved by the playback head omitting certain sections of the tape. For a 20% increase in speed, every fifth section of the tape is omitted, while for a 50% reduction, every second section is repeated. To prevent the omission or repetition of segments from causing noticeable distortion, the variations in speed adjustment must be limited to a maximum length of 30 milliseconds or less per section. This length is shorter than any musical or spoken sound and, therefore, can be repeated or omitted without being noticed. Thus, the distortion-free speed adjustment range is limited to between 90 and 120% of the original speed. The unit is called the Eltro Tempo Regulator.

Another effect which can be achieved using a tape recorder is that of playing an instrument backward. This can be achieved bv recording the signal normally (Fig. 5-65) and playing it back with the tape path around the capstan and capstan idler reversed from normal (Fig. 5-66), or by recording the signal normally and flipping the tape

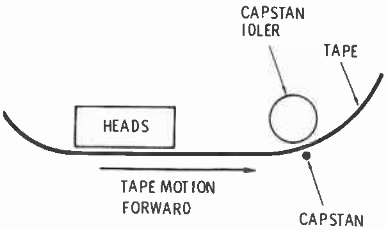


Fig. 5-65. Normal tape path.

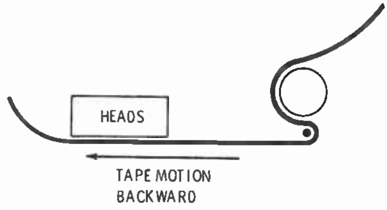


Fig. 5-66. Tape path to play a tape backward.

upside down for playback. The second method is the one normally used in multitrack work.

To record a backward guitar track, for example, the master tape is put on the machine upside down, and the guitar player listens to the master tape played backward in sel sync while playing his part normally (i.e., forward). The tape is then returned to the right side up position, and the master tape plays normally while the guitar reproduces backward.

The main concern in this procedure is that if the guitar track is to play backward on track 2 of a 16-track tape, it must be recorded on track 15 when the tape is upside down because 2 is the second track from the top of the head, and 15 is the second track from the bottom. When the tape is upside down, the track order is inverted relative

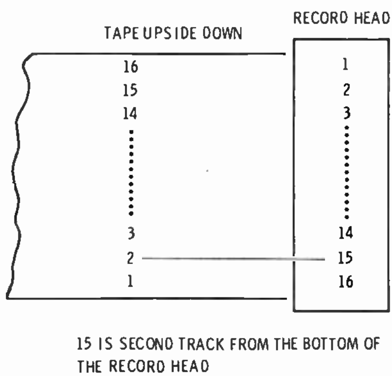


Fig. 5-67. To achieve a backward instrument effect on track 2, flip the tape upside down and feed the instrument to channel 15 of the tape machine. This corresponds to track 2 on the upside-down tape.

to the recorder heads (Fig. 5-67). The tape cannot be recorded backward using the method of reversing the tape-threading path around the capstan to move the tape from the take-up reel to the supply reel, because the erase head would erase the signal as soon as it left the record head. Disconnecting the erase head would only result in a distorted bias signal which would be too low in level to bias the tape properly unless some sort of dummy erase head were connected in its place.

Backward effects can also be created by splicing in a section of tape upside down so that it is played backward. Again, the tracks will be switched around so that on a two-track stereo tape a sound that played forward through the left channel will play backward through the right channel.

THE PHASING EFFECT

Phasing is a term used to describe the sound which results from any device that causes multiple sharp dips or notches in the frequency response of a system in such a manner that the dip frequencies vary (Fig. 5-68).

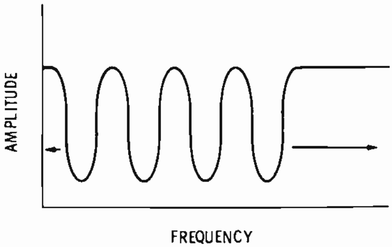


Fig. 5-68. The phasing effect creates a comb filter with variable dip frequencies.

The effect was originally discovered in an attempt to make use of the slight difference in running speed of tape recorders to produce a very short time delay to electronically double voices and/or instruments. The signal to be doubled was played back in sel sync during a mix, while the rest of the signals were played back from the repro head. The signal from the sync head was fed to the input of two different tape recorders running at the same speed as the master tape. The signals were recorded on the two separate machines, and the signals played back from their playback heads were combined and then mixed in with the other tracks of the master tape. Since the distance between the sync head and the playback head of the master tape recorder was the same as that between the record and playback heads of the two other recorders, and because the tape speeds were the same, the signals traveling through the two extra tape machines arrived at the console in sync with the signals from the master recorder playback head (Fig. 5-69).

The doubling effect should have resulted from the delay between the two signals caused by small variations away from constant speed in the two recorders. Instead of the straight doubling effect which was expected, a jet-like phasing effect resulted. This was caused by phase cancellations of the closely spaced signals to be doubled, hence the name *phasing*. These cancellations were the result of the phase inaccuracies which occur to some extent between all record and playback heads and amplifiers. As the instantaneous tape speed changed, the frequency of cancellation changed, giving the jet sound its varying pitch.

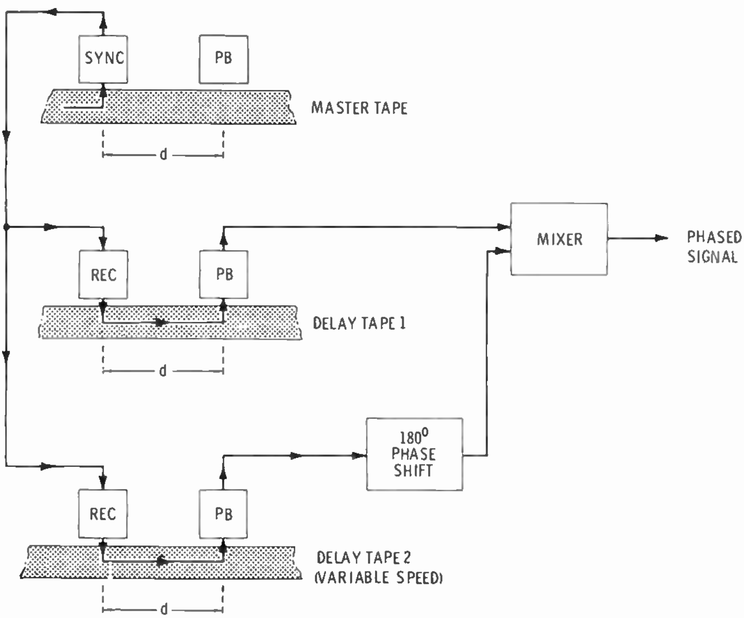


Fig. 5-69. The signal path to create the phasing effect using a master tape machine and two extra recorders.

Once engineers understood the principle involved in this cancellation, they discovered that the cancellation effect could be enhanced by reversing the phase of one of the signals fed to the two extra tape recorders (i.e., putting it 180° out of phase with the other signal) and that the frequency of cancellation could be changed by varying the speed of one of the two extra machines either above or below its normal speed. This was originally done by the engineer resting his thumb on the flange of the reel to slow down the recorder slightly. This gave rise to the term *flanging*, which is another name for the phasing effect. By using a vso, the speed can be varied both above and below normal. The phasing effect is very interesting because it makes a signal sound different from any naturally occurring sound.

The original method of producing it requires either two tape recorders in addition to the machine used to play back the tape and the machine used to record the mix, or one additional recorder and a blank track on the master tape (which can be used as the second extra recorder) (Fig. 5-70). For best results a means of reversing the phase of the signal is required. This is easily done with a balanced console and patch bay, but unbalanced patch bays require the use of a transformer connected to achieve 180° phase reversal. The vso must also be connected for best control of the effect, and the entire setup may take more than an hour before the effect can be heard and before the producer can decide whether he wants to use it or not.

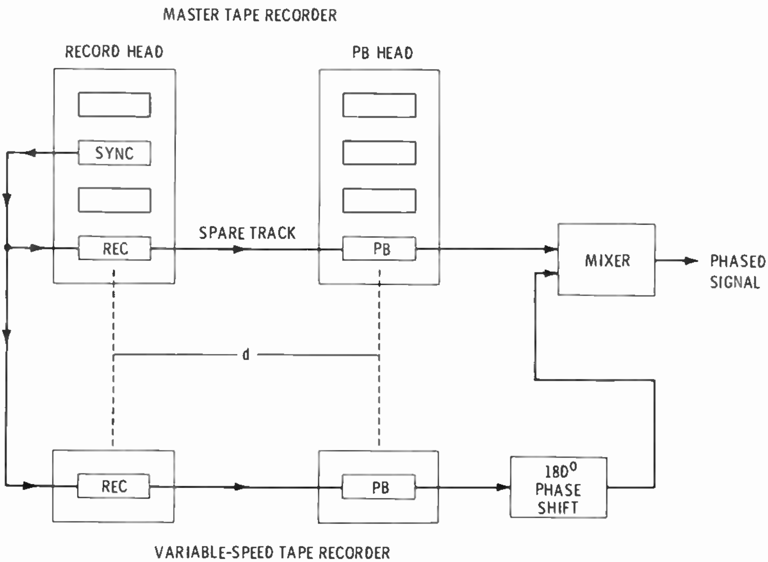


Fig. 5-70. The signal path to create the phasing effect using a master tape machine with a spare track and one extra recorder.

The phasing effect is the result of a moving *comb filter* which has an almost infinite number of dips in its response. The sweeping of these dips creates the phasing effect. If the dips are stationary, the signal merely sounds like it is extremely equalized.

Countryman Associates discovered that the ear perceives the phasing effect even when all but three of these dips are removed; they designed their Model 967 phase shifter to produce the effect (Fig. 5-71). The signal to be phased is fed into the input of the device, where it is split into two parts. One of them is fed directly to a mixer, while the other one is fed first to a variable phase-shift network and then to the mixer where it is recombined with and partially cancels the original signal (Fig. 5-72). A control is provided to vary the amount of phase shift produced and thus vary the fre-

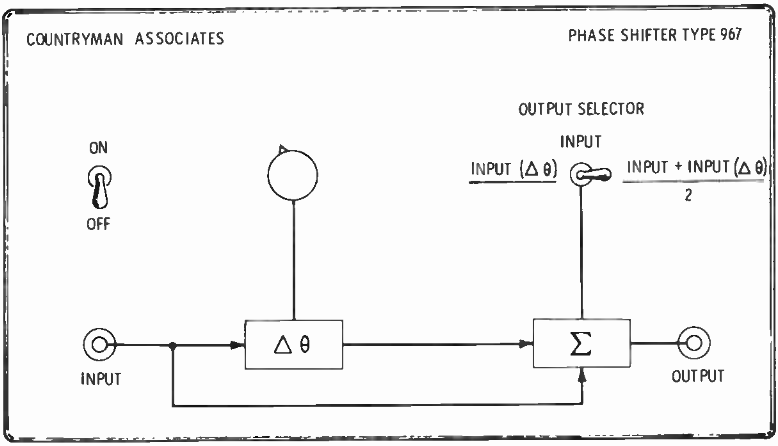


Fig. 5-71. The Countryman Associates Model 967 phase shifter.

quencies of cancellation. The phase-shifted signal itself does not sound phased. It is the combination of the shifted signal with the unchanged signal which produces the effect. The depth of the phasing effect is controlled by the amount of cancellation (i.e., depth of the notches), which is maximum when the level of the direct and phase-shifted sounds are equal. This mix is preset within the unit, but a front-panel switch enables the engineer to obtain the phase-shifted signal alone at the output of the device so that through the use of the multiple jacks on the patch bay and extra faders, the balance of direct to shifted signal can be changed.

The phase-shifted signal can be used to give a stereo effect to a mono sound by putting the direct sound in one speaker and the signal shifted by a constant amount (i.e., leave the knob on the unit in one position) in the other speaker. Since phase shifting is a form of short time delay, a stereo effect is produced. If, however, the signal is later combined into mono, the phase-shifted signal will cancel certain frequencies of the direct signal and the result may not sound very good. Therefore, when attempting this type of stereo effect, the

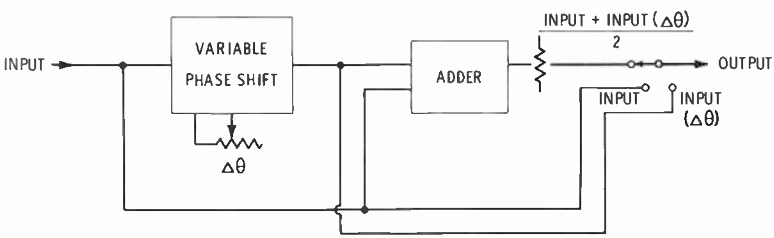
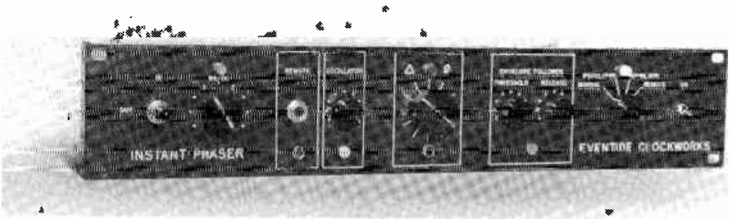


Fig. 5-72. Block diagram of the Countryman phase shifter.

engineer should listen to what happens when the signal is played back in mono. (Remember, a majority of the fm audience listens in mono not stereo and all a-m stations are mono; many people decide whether or not to buy albums on the basis of what they hear on the radio.)

The Model 967 has two switches on the front panel. One is an on/off switch for its battery and the other is a three position switch which controls the output of the device. In the Input position, the device feeds the input signal to the output jack and becomes a unity gain amplifier. In the Input ($\Delta\theta$) (Delta Theta) position, the phase-shifted signal only is fed to the output jack. In the Input + Input($\Delta\theta$)/2 position, the output is fed a combination of the input signal and the phase-shifted signal, which is adjusted so that the output is at the same level as the input signal. This third position produces the phasing effect when the front-panel knob is turned.

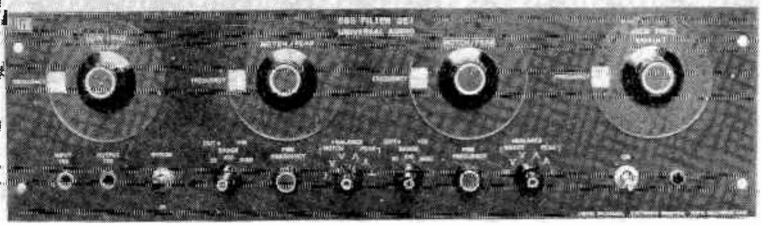
A more elaborate phase shifter is available from Eventide Clock Works. In addition to the control of the notch frequencies, their *Instant Phaser* (Fig. 5-73) allows the depth of the phasing to be varied by changing the mixture of the direct and phase-shifted signals. The comb filter can be swept in four ways: manually by a knob, automatically by a variable-frequency oscillator, as a function of the amplitude of the input signal, or remotely by a foot pedal. Two units can also be interlocked to provide stereo phasing.



Courtesy Eventide Clock Works, Inc.

Fig. 5-73. The Eventide Clock Works Instant Phaser.

The phase shifter makes the phasing effect extremely simple to connect (only two patch cords are needed) and easy to operate. The input of the device can be connected to an unused echo send buss and the output sent to an echo return fader so that more than one signal can be phased, or the phase shifter may be connected directly between a mike preamp output or recorder playback output and an input fader if phasing is only desired on one signal. There are several other units such as the UREI Little Dipper (Fig. 5-74) which can produce an effect similar to phasing. These units make use of a sharp single-dip filter with a continuously variable center frequency, which can be swept back and forth through the audio spectrum.



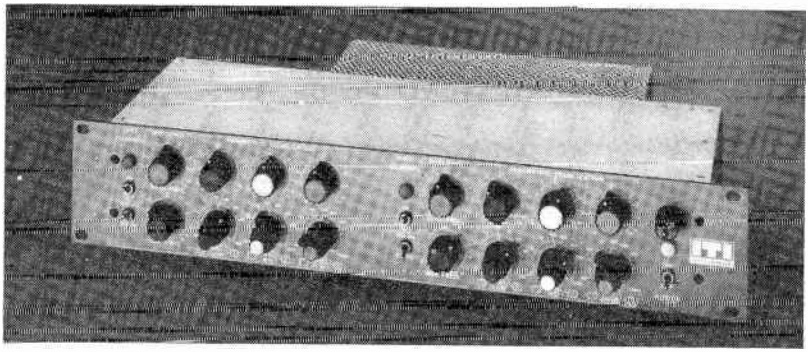
Courtesy United Recording Electronics Industries

Fig. 5-74. The Universal Audio Model 565 Little Dipper variable high- and low-pass, peaking, and notch filter.

OTHER EFFECTS

Many other effects can be used to alter the sound of instruments. The Teletronix LA-2A compressor can be used to get guitar amp-type distortion on a clean track because it uses tubes and can produce the type of overload distortion that tube-type guitar amps can. Distortion is achieved by increasing the gain control until the compressor output is overloaded. This same method can be used to overload transistorized compressors and other amps in the control room, but their distortion is fuzzy and grating due to the predominance of odd-order harmonics as compared to the even-order distortion products of tubes.

Wah-wah pedals can be used to add emphasis to certain notes in a musical passage. The wah-wah is a variable-bandpass filter, with the pedal controlling the center frequency of a peak boost circuit. The amount of boost stays constant, but as the pedal is moved forward, the center frequency rises; as it is moved backward, the center frequency falls. The same effect can be achieved with certain types of equalizers that have continuously variable center-frequency controls, such as the International Telecomm, Inc. Models ME-230 and MEP-130 (Figs. 5-75 and 5-76). The ME-230 provides peaking and dipping curves with EQ frequencies continuously variable in three overlapping ranges, 10 to 800 Hz, 100 to 8000 Hz, and 400 to 25,600 Hz. The amount of boost or cut is continuously variable over a ±12-dB range, and curve shape is variable from 4 to 14 dB per octave. A 10-kHz high-frequency and 50- or 100-Hz low-frequency shelving EQ with continuously variable range of ±12 dB is also provided. The Model MEP-130 single-channel, console-mounted unit has the same adjustments and specifications for the peaking and dipping EQ as the ME-230. In addition, the high- and low-frequency ranges can be switched to provide continuously variable frequency-shelving curves.

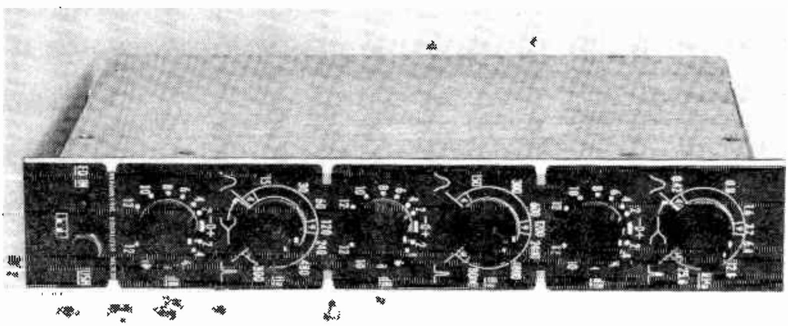


Courtesy International Telecom, Inc.

Fig. 5-75. The Model ME-230 two-channel, rack-mounted parametric equalizer.

Several companies produce frequency multipliers and dividers made especially for horns to add octaves or fifths above or below the fundamental frequencies. These devices can be used on other instruments, such as voice, but only on single notes. Fuzz boxes, wah-wah pedals, and frequency multipliers tend to be very noisy. In addition, they are designed to be fed from high-impedance, low-level sources, such as guitars or microphones, to high-impedance, low-input level guitar amps, so care must be taken not to overload them when connecting to the patch bay or severe distortion will result.

Signals can be passed through the filters and envelope generators (electronic volume controls) of an electronic music synthesizer, but even good synthesizers are not as quiet as other pieces of equipment in the studio, although they are considerably quieter than fuzz boxes and wah-wahs. The filter frequencies, signal volume, and left to right to front to back positioning can be changed gradually or rapidly and continuously by low-frequency oscillators or by other



Courtesy International Telecom, Inc.

Fig. 5-76. The Model MEP-130 single-channel, console-mounted unit.

audio signals. Synthesizers themselves can be used to create new instruments with timbres and amplitude variations unlike any existing instrument or, if desired, very much like real instruments.

The *Leslie speaker* can be used to produce tremolo through the use of the *Doppler* effect. The Doppler effect is responsible for the change in pitch of a car horn as the car passes by a listener. As the car approaches, the car speed is added to the velocity of propagation of the sound wave, and more cycles reach the listener's ear per second than if the car were standing still, so the pitch appears higher than it really is. When the car moves away from the listener, the car speed is subtracted from the velocity of propagation of the sound wave, and fewer cycles reach the listener's ear per second, so the pitch appears to be lower than it really is. When the car is next to the listener, the wave propagation velocity is that of sound in air because the car is neither approaching nor leaving the listener and the true pitch is heard.

The Leslie speaker uses rotating baffles to produce pitch variations by causing the direction of sound radiation to rotate 360° continuously. Thus, every time the axis of radiation passes the ear or mike, the pitch rises and then falls. Since the direction of radiation is varying, the loudness of the signal also increases and decreases.

The Leslie speaker has two speakers in it, one for frequencies above 500 Hz and one for frequencies below 500 Hz. Since the two speakers are located respectively at the top and bottom of the speaker cabinet, best results are achieved by using one mike for the highs and another for the lows. The mikes should be pointed into the louvres (vents) rather than into the open back of the cabinet because the vents act to change the phase of the signals and produce a stronger effect. The baffles have two speeds, a fast tremolo and a slow one. The slow speed produces an effect very reminiscent of phasing.

Although the Leslie speaker was designed for use with an electric organ, it can be used with any electric instrument or any sound picked up by a mike. Signals can also be played back into a Leslie speaker from a tape recorder and be added to a mix or rerecorded on another track to give the Leslie effect to any signal. The effect is most pronounced for middle and high frequencies and is only noticeable on low-frequency tones that are sustained. Combining the Leslie with reverberation greatly enhances the effect; as a result, several studios have put Leslie speakers inside their acoustic echo chambers so that this effect may be selected at the push of a button. Just as signals can be played back through a Leslie speaker and recorded, signals can be played back through cheap car-radio speakers and rerecorded to give the sound of a record being played or sung through a radio.

The *Sono-Vox* sound-effects machine enables recorded sound effects to be formed into words by the human mouth. A device called an *articulator* is held against the throat of the person who will control the effect, and it induces the sound effect into his throat at the same point that the human vocal cords produce sound. He then mouths words into a mike, but makes no sound of his own. Through the use of this device, whistles, for example, can be made to pronounce words.

Electronic rhythm boxes can be used instead of real drummers or to keep time if a drummer is to be overdubbed. Similarly, metronomes can be recorded on the tape so that musicians have an exact time base to work with when no drummer is present. These devices can be connected into the patch bay if they have electrical outputs. If they are mechanical, they can be miked.

Crossfading can be used between songs on an album to eliminate the space between songs. Final mixes are used, and the beginning of one song and the end of another song are recorded on a multitrack tape (at least four tracks for stereo mixes or eight tracks for quad mixes) so that the second song begins at the desired point relative to the first. This tape is then mixed down to two tracks (four for quad) so that the songs overlap smoothly. This crossfaded piece of tape is then spliced into the end of the first song and the beginning of the second song.

Care must be taken that the level of the songs on the mix match those of the original tapes so that the splice will not be noticeable. There is an increase of tape noise during the spliced-in portion, but if the tape is quiet to begin with, it will not be noticeable. If the hiss is noticeable at the splice, both songs can be recorded entirely on multitrack tape, beginning the second song where the crossfade is desired so that the increased noise level will be constant.

Through the use of the *stereo interconnect* outputs on compressors, the volume of one instrument can be controlled by another. For example, a voice track can be set to cause gain reduction both in its own compressor and in a compressor through which background music is passing. The voice signal can be tapped before entering its compressor and mixed in with the output of the music compressor. The music compressor is set so that the music does not trigger gain reduction. When the voice is present, compression is triggered in the music compressor and the music fades down to allow the voice to be heard. When the voice stops, the music fades back up. The signal output of the voice compressor is not used in this application.

Other sound effects such as birds, whistles, trains, etc., can be mixed into songs by overdubbing them at the appropriate spots on the multitrack tape. Bottles, ash trays, hand claps, foot stomps, coins, or any other object imaginable can be used for rhythm instruments and even for melodic lines if they can be tuned (for example, bottles with different amounts of water in them). With proper EQ and placement in the mix, many odd noise makers can become musical instruments which complement the music but which listeners cannot identify as a specific instrument.

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CHAPTER 6

Consoles

The function of the console is to provide control of volume, tone, blend, and spatial positioning of signals applied to its inputs by microphones, electronic instruments, and tape recorders, and to route these signals to a monitor system and a tape machine so that they can be heard and recorded. Before the introduction of multitrack tape machines, all of the sounds which were to be part of a recording were mixed together during the live performance. If the recorded blend was not satisfactory, the selection had to be performed again and again until the desired balance was obtained. The availability of multitrack tape machines has made recording a three-stage process.

RECORDING

During the first stage, called *recording*, either some or all of the instruments to be used in a song may be recorded. The instruments are divided into groups of one or more, as determined by the desires of the producer and the engineer, to exercise individual control over their tone, level, and spatial characteristics in the final product. Each of these groups is recorded on a separate track of the master tape. This is done by connecting the microphones to the *input section* of the console and assigning each mike to a console output through a series of switches or push buttons called the *output assignment matrix*. If several instruments are to be recorded on one track, their microphones are assigned to the same console output channel and the balance between them is adjusted by using the level control provided for each microphone input. Although tonal and spatial changes can be made at this point, the bulk of these are usually made in a later stage. Each of the console main outputs is connected to a different track on the tape machine. For simplicity, console output one is connected to track one, output two to track two, and so forth. Each track is recorded at a level as high as possible without overloading the tape and without regard to its musical relationship to the instruments on the other tracks. This is done to achieve the best signal-to-noise ratio on each track of the tape, so that the realism of the final product will not be impaired by the audibility of tape hiss.

Monitoring

Since the producer and engineer must be able to hear the instruments as they are recorded and when the tape is played back in order to judge the quality of the performance, the console and tape machine outputs are connected to a *monitor mixer*. In the *program* mode, the console outputs feed this mixer to allow the different groups of instruments to be blended in the monitor speakers while being recorded separately on the master tape. In order to hear an approximation of the final product, a separate monitor level control is provided for each console output, as well as a switching control to assign each signal to a spatial location in the stereo or quad monitor speakers. The monitor mixer often has echo facilities so that monitoring can be even closer to the final product.

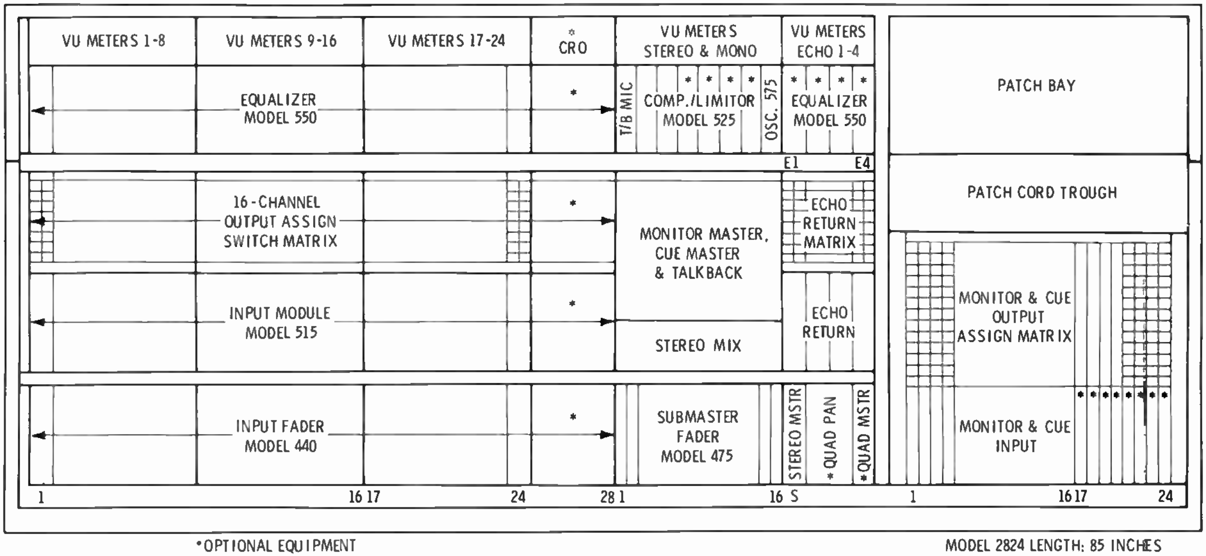
In the *tape playback* mode, the tape machine outputs are connected to the mixer level controls. The switching is set up so that the mixer control, fed by console output one in the program mode, is fed by tape track one in the tape playback mode at the same level. This permits the balance set during recording to be heard again during playback without readjusting the level controls.

Overdubbing

If instruments not present during the original performance are desired in the recording, or if one or more musicians made minor mistakes during an otherwise good performance, a second recording stage called *overdubbing* is begun. The musicians wear headphones and play along with the previously recorded tracks. The new performances are recorded in synchronization with the original on empty tracks or on tracks containing information no longer desired. When overdubbing is done, the first recorded tracks are called the *basic* or *rhythm tracks*. The monitor mixer is put in the *sync* mode which allows either program or tape playback to be selected individually for each mixer input.

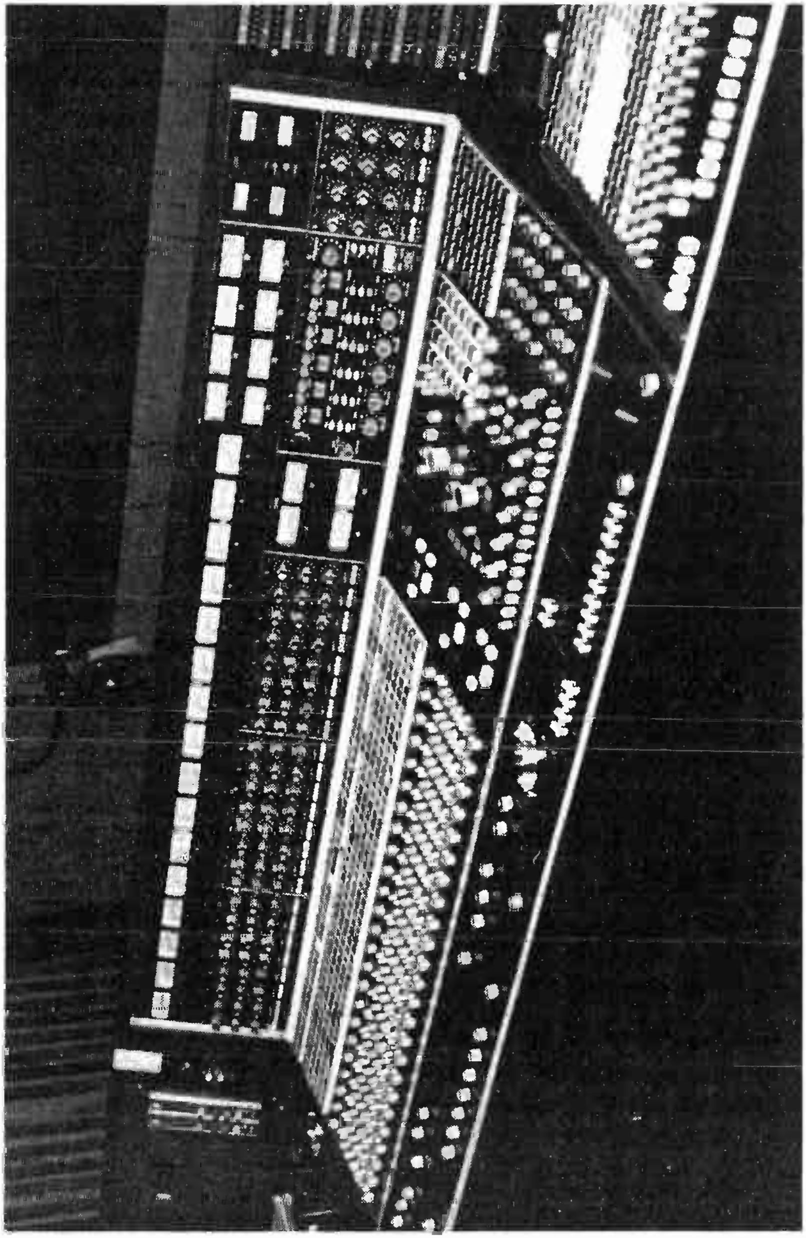
Mixdown

After all the desired musical parts have been performed and recorded satisfactorily, the *mixdown* or *remix* stage takes place. The microphones are disconnected from the console inputs, and the outputs of the multitrack tape machine are connected in their place.



(A) Line drawing of the component layout.

Fig. 6-1. The API



(B) The 2824 console installed in a studio.

Courtesy Automated
Processes, Inc.

Model 2824 console.

This is usually done by changing the mike/line switches on the appropriate console inputs to the *line* position. The master tape is played over and over while the level, tone, and spatial characteristics of each track are adjusted. At the same time, the individual signals are blended into one or more composite signals which are sent to the console outputs. The outputs are now connected to a one-, two-, or four-track recorder, depending on whether the product is to be mono, stereo, or quad. The blend is monitored in the program mode of the monitor mixer, and when it is satisfactory, it is recorded. When approved, this recording is called the *final mix* and is used to cut the records. The mix tape can be played through the monitor system by switching it to the tape playback mode, which now connects the monitor mixer to the outputs of the mixdown recorder.

PROFESSIONAL CONSOLES

The consoles used in modern professional multitrack studios have similar controls and capabilities. They differ mainly in appearance, the location of the controls, and ease with which certain results can be achieved (i.e., at the flick of a switch, or patched in at the patch bay). The consoles are usually designed using plug-in modules which are mounted in a custom-designed enclosure. The plug-in nature of the modules makes them easily removable for servicing and makes modules of the same type interchangeable. For example, if one equalizer breaks down, that EQ module can be replaced by either a spare or a unit borrowed from an input channel not in use at that time. An example of a well-equipped console with features paralleling or improving on those in most other consoles is the Automated Processes, Inc. (API) Model 2824 (Fig. 6-1).

Patch Bays

The components of the console interconnect in the *patch bay* (also called the *patch panel* or *patch rack*) shown in Fig. 6-2. This is a panel which contains a jack corresponding to the input and output of every discrete component or group of wires in the control room. One jack is said to be *normalled* to another if the components connected to the two jacks are connected directly to each other when there is a plug in neither jack (Fig. 6-3), but are not connected if a plug is inserted in either jack. When a plug is inserted into one of the two normalled jacks, it is said that the normal between those jacks has been *broken* (Fig.6-4).

The purpose of breaking normalled connections by the insertion of a plug is to enable the engineer to connect different or additional pieces of equipment between two normally connected components using *patch cords*. For example, a limiter can be temporarily patched

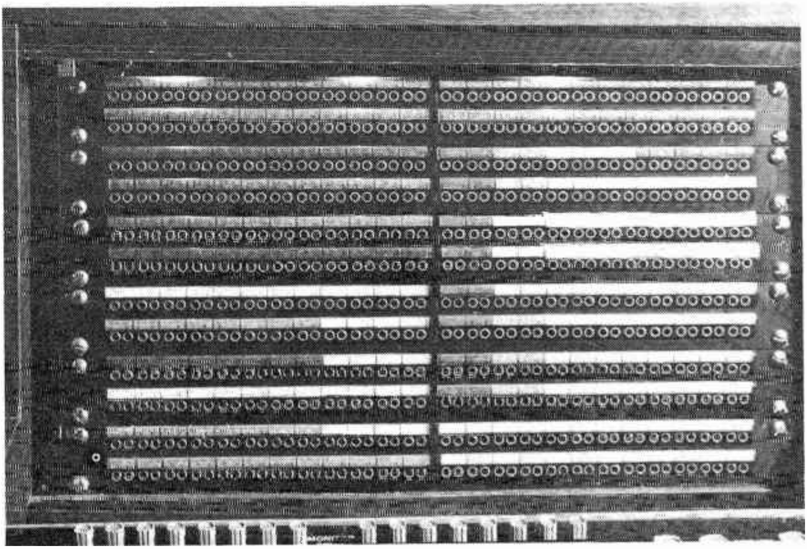


Fig. 6-2. A patch bay.

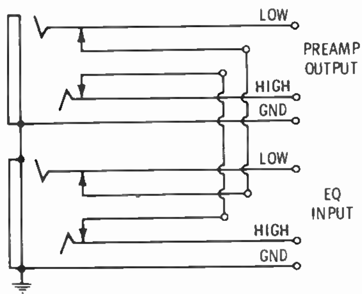
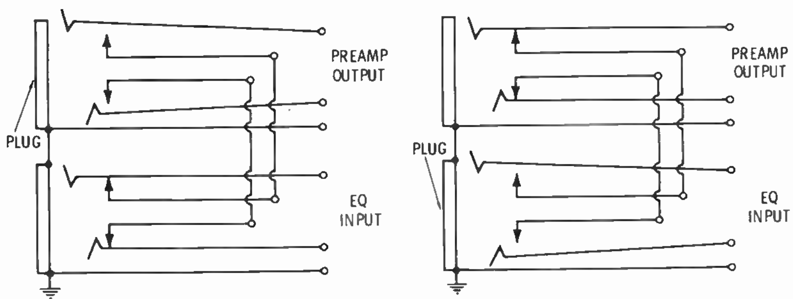


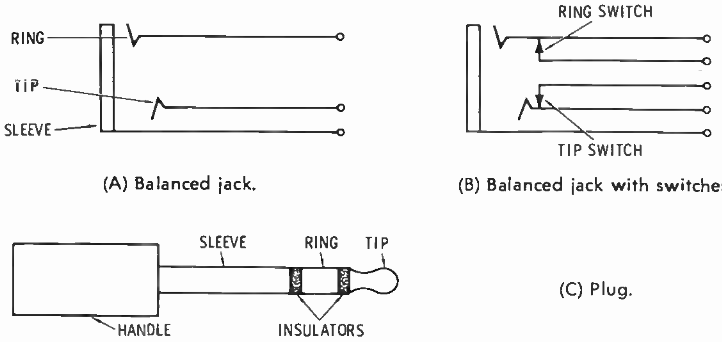
Fig. 6-3. Normalled jacks: the preamp output is normalled to the EQ input.



(A) Plug in preamp output jack.

(B) Plug in EQ input jack.

Fig. 6-4. A plug inserted into the preamp output jack or the EQ input jack breaks the normal connection.



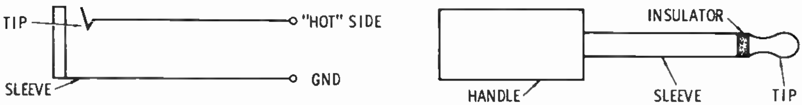
(A) Balanced jack.

(B) Balanced jack with switches.

(C) Plug.

Fig. 6-5. Types of balanced jacks, and plug used with them.

between a mike preamp output and an equalizer input. The same limiter could later be patched between a tape machine output and a console line input. Other uses of the patch bay are to bypass defective components or to change a signal path to achieve a certain effect. Patch bay jacks are usually balanced tip-ring-sleeve types having two conductors plus ground, for use with balanced circuits (Fig. 6-5). Unbalanced jacks and plugs have only tip and sleeve connectors (Fig. 6-6).



(A) Jack.

(B) Plug.

Fig. 6-6. Unbalanced jack and plug.

Preamps

The output level of microphones is very low and requires specially designed amplifiers called *mike preamps* to raise this level to that needed by the console without degrading the *signal-to-noise ratio*. The electrical signals produced by a mike are actually electrons in motion. Since electrons are moving from atom to atom in any conductor at temperatures above the absolute zero of −273 °Celsius (formerly centigrade), there is a limit to the quietness of an electrical system with a certain frequency response. When any signal is amplified, noise caused by random electron motion in the signal source is also amplified. Transistors and tubes also generate noise which is added to the signal. Since amplifiers cannot distinguish between desired signals and noise, both are amplified equally. Once the signal reaches a certain level, however, the noise added to it by high-quality components is insignificant. The signal-to-noise ratio is primarily determined by the mike preamp which raises the signal to this higher level. Balanced mike lines and mike preamp inputs reduce the noise content of the signal in comparison to unbalanced lines and inputs.

All amplifiers have a maximum undistorted output level. If a signal fed into a mike preamp is high enough that the preamp gain raises it above this level, the signal becomes distorted. Fig. 6-7 shows the

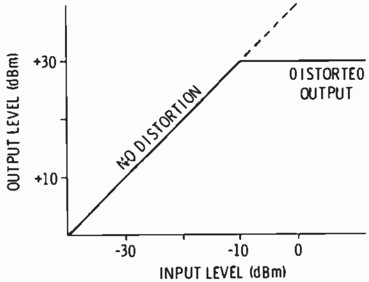


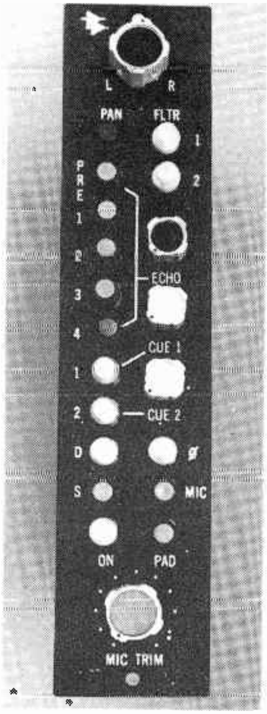
Fig. 6-7. The input-output curve for a mike preamp.

input-output curve for a mike preamp with 40-dB gain and a +30-dBm maximum output before clipping. The preamp output follows the 45° line when the input signal is below −10 dBm. When the input signal rises above this level, the gain of the amp tries to raise the output signal above +30 dBm, so the signal is clipped, producing severe distortion. To avoid this, API provides a means of reducing both the gain of the preamp and the level of the incoming signal. Fig. 6-8 shows API's Model 515J input module which contains a mike preamp. The MIC TRIM control varies the preamp gain over a continuous 36-dB range from +4 to +40 dB, and the PAD switch inserts a 20-dB attenuator ahead of the preamp. The pad decreases the signal-to-noise ratio, so it is used only when the signal level applied to the mike input is so high that even the 4-dB minimum gain of the preamp results in distortion. To obtain the lowest noise levels, the gain is set to amplify the input signal only as much as is needed by the console.

The output of the mike preamp is fed to a phase reversal switch. This switch is used to effect a 180° change of phase in the input signal to compensate for a microphone or mike cable that is out of phase. This prevents the cancellation which occurs if out-of-phase mikes are mixed. It can also be used to deliberately put two signals out of phase.

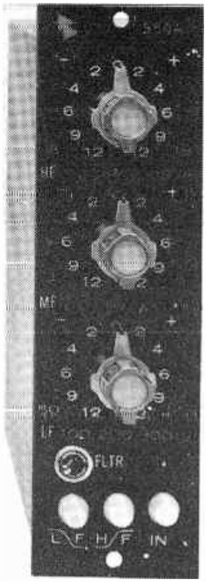
Equalizers

The output of this switch is normalled to the Model 550A equalizer (Fig. 6-9). This unit provides simultaneous control of three



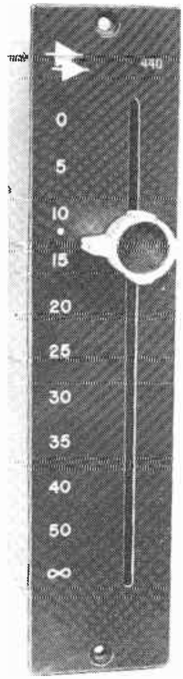
Courtesy Automated Processes, Inc.

Fig. 6-8. The API Model 515J input module.



Courtesy Automated Processes, Inc.

Fig. 6-9. The API Model 550A equalizer.



Courtesy Automated Processes, Inc.

Fig. 6-10. The API Model 440 fader.

overlapping frequency ranges: The high frequencies are 5, 7, 10, 12.5, and 15 kHz; the midrange frequencies are 0.4, 0.8, 1.5, 3, and 5 kHz; and the low frequencies are 50, 100, 200, 300, and 400 Hz. The frequency to be affected is selected by the inner of the two concentric knobs provided for each range. The outer knob controls the amount of boost or cut, which is variable in steps to ±2, 4, 6, 9, or 12 dB. All three ranges produce peaking and dipping curves, and the high and low ranges can be individually changed to produce shelving curves by depressing the HF and LF buttons, respectively. An audio-frequency bandpass filter with 3-dB down points at 50 Hz and 15 kHz is inserted into the equalizer circuit by the FLTR toggle switch. With the filter, any signals above or below the audio range (such as high-frequency oscillations or rumble transmitted through the studio building) are attenuated with minimal effect on the sound of the program material. The IN push button enables the engineer to put the selected EQ in the circuit or to remove it. No switching noise occurs, so EQ can be added or removed during a recording. An LED illuminates when the EQ is in the circuit.

The EQ output is normalled through the input channel on/off button to the linear motion channel volume control, which is called a *fader* or *pot* (potentiometer). The fader is an API Model 440. Its scale indicates the number of dB attenuation introduced into the circuit (Fig. 6-10). When the knob is at the top position, attenuation is 0 dB (neglecting the constant insertion loss of the fader which is inherent in its design). Attenuation increases to a maximum of 90 dB when the knob is in the ∞ position. The on/off button allows a signal to be inserted into or removed from a program on command without disturbing the setting of the fader.

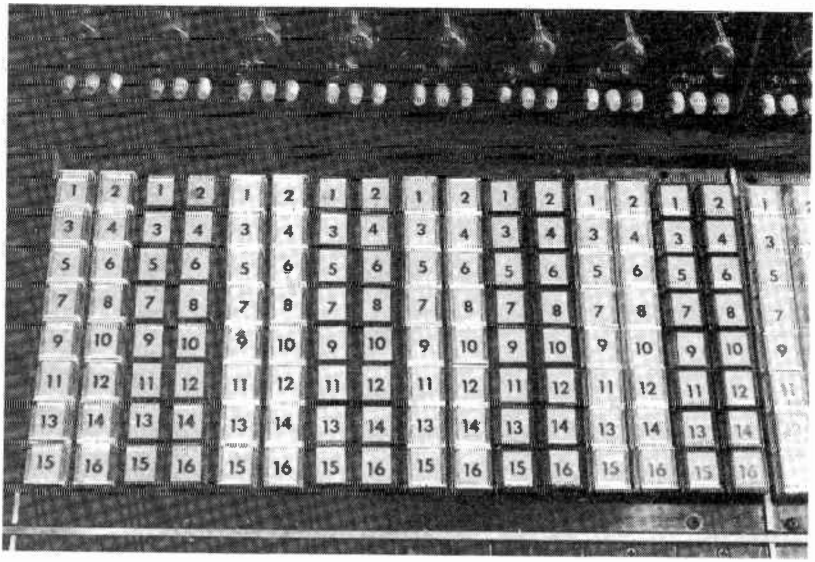
The fader output is normalled to a switchable high-pass filter controlled by the FLTR 1 and 2 buttons on the input module. These provide a bass rolloff at 12 dB per octave with turnover frequencies of 70 Hz or 100 Hz for buttons 1 and 2, respectively, or 200 Hz for both buttons together. These rolloffs are included to compensate for the increased bass output of directional microphones when used in close miking if no rolloff is available on the mike itself. They can also be used in conjunction with the equalizer to provide additional control over low-frequency signals.

THE ACTIVE COMBINING AMPLIFIER

The output of the high-pass filter is connected to the output switching matrix (Fig. 6-11) either directly or through the *pan pot,* depending on the position of the PAN push button. The switching matrix (also called the track, channel, or submaster selector switches) consists of 16 push buttons which determine to which output of the console (and therefore which track of the tape) a signal is sent. Each button connects the signal of that input to a mixing amplifier. The mixing amplifier combines it with the signals from all other inputs for which the same channel-selector button is depressed. This mixing amplifier is called an *active combining amplifier* (*ACA*), *active combining network* (*ACN*), or *summing amplifier*. An ACA is designed to have many inputs, each isolated from all the other inputs.

In Fig. 6-12, due to the isolation between the inputs of an ACA, signal A does not appear in the output of ACA_2, and signal C does not appear in the output of ACA_1, even though signal B is fed equally to both ACAs. The channel-selector push buttons illuminate when they are in the ON position to enable the engineer to easily see where each input signal is assigned.

The pan pot has one input and two outputs; the relative levels of the two outputs vary with the pan-pot setting. When the pot is cen-



Courtesy Automated Processes, Inc.

Fig. 6-11. The switching matrix on the 2824 console.

tered, the outputs are at equal levels. As it is turned to one side or the other, one output increases in level while the other decreases until one output is fully on and the other is fully off, when the control is turned completely to one side. The outputs of the pan pot are labeled for the direction of knob rotation which increases the level of that output. The output which increases when the knob is turned to the right is called the right-channel output, and similarly for the left channel.

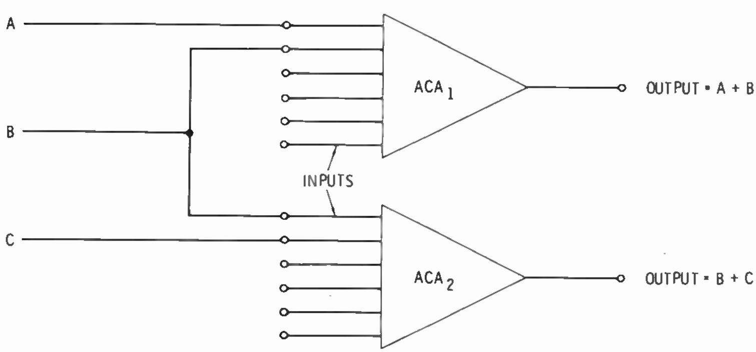


Fig. 6-12. Isolation between inputs of an ACA.

The most common use of pan pots is in mixing where they are used to position instruments to the left, right, front, back, or somewhere between the stereo or quad speakers. Pan pots can also be used to make the signal source appear to move from one location to another. When the pan button is on, the left output is connected to the odd-numbered push buttons on the output switching matrix, while the right output is connected to the even-numbered buttons. This allows the input signal to be panned between any odd/even combination of output channels desired, determined by the channel-selector buttons depressed.

Quad pan pots having one input and four outputs which vary according to the position of a *joystick* control are available at the patch bay (Fig. 6-13). The position of the joystick indicates where

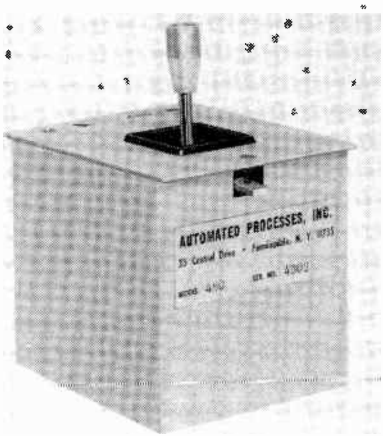
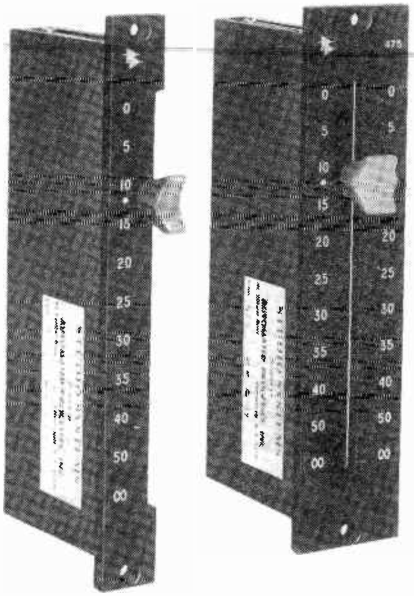


Fig. 6-13. The API Model 480 quad pan pot.

Courtesy Automated Processes, Inc.

the signal source would appear to be in a quad mix. Moving the joystick in a circle would make the signal appear to move in a circle around the listener. The signal to be panned is patched into the quad panner input, and the four outputs are patched into the inputs of the ACAs for the output channels.

The output of each ACA is normalled to a Model 475 *submaster fader*. It has the same characteristics as the Model 440 input fader but is only one-half the width (Fig. 6-14). This fader controls the amount of signal fed to the output or line amplifier for that channel. In turn, the channel feeds both the monitor system and the tape recorder electronics. The submasters are unnecessary for much of the work done at the studio, especially if only one mike is assigned to any of the console outputs as is common in pop music sessions. In this case, the submaster for that channel merely duplicates the func-



Courtesy Automated Processes, Inc.

Fig. 6-14. The API Model 475 fader.

tion of the input fader. One may be set at a fixed level, while the other is used to adjust for the proper record level as determined by watching the VU meter. Since the two controls are electrically in series, having either of them other than fully on (i.e., 0-dB attenuation) will decrease the gain range controllable by the other fader. For example, if the submaster is set for 15-dB attenuation, the highest gain available from the console by adjusting the input fader will be 15 dB below maximum.

The submasters are convenient for controlling an output channel which is fed by more than one mike. For example, the engineer might mike a set of drums or a horn section with three mikes, combining them so that they can be recorded on one track. He sets the input faders to get the desired musical balance between the three mikes and then adjusts the console output for the proper recording level as indicated on the VU meter. Rather than moving the three input faders, he can adjust the submaster for that channel either up or down to get the desired output level. It is good practice to set all submasters at some fixed amount of attenuation, such as 15 dB, before setting input levels. Doing this leaves some flexibility should more gain be needed later. Also, if the submasters are inadvertently moved between takes, they can be reset to their previous positions without new level checks.

MULTITRACK RECORDING

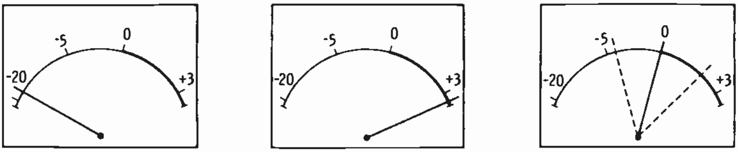
In multitrack recording, most tape tracks are fed by a single microphone, so only a few mixing channels are ever used at once. Providing more than 16 mixing channels, each requiring its own ACA, submaster, channel-selector push buttons, and wiring, increases the console cost unnecessarily. To record 24 tracks, the output of each input module high-pass filter is also connected to a *direct-output* switch (D on the input module). This switch permits feeding the signal in that module to the correspondingly numbered output channel. For example, module 1 signal is fed to output 1, module 2 to output 2, and so forth, through module 24 output being fed to console output 24.

The direct switch on modules 1 through 16 disconnects the corresponding submaster fader from its line amplifier and connects the module high-pass filter output in its place. On modules 17 through 24, the direct switch just makes or breaks the connection between the filter output and the proper line-amp input. The direct outputs of modules 25 through 28 are available at the patch bay. If a mixing channel is needed for any of the tracks 17 through 24, one of the ACAs for tracks 1 through 16 is used. Its output, or the output of its submaster fader, is patched to the line amplifier input for the desired output channel.

The output of each of the 24 line amps is monitored by a VU meter and an LED peak indicator which displays waveform excursions that are too short in duration to deflect the meter. Together, they indicate the level of the signal to be recorded on tape. If the signal is too low, tape noise will be a problem when the recording is played back. If the level is too high, the tape and possibly the console or tape-recorder amplifiers may cause distortion of the signal. Proper record level is achieved when the highest reading on the VU meter is near the 0 level and the peak indicator only flashes occasionally, although levels slightly above or below this will not cause difficulties (Fig. 6-15).

The two CUE volume controls and on/off switches on the input module permit the signal at the output of the equalizer to be sent to either or both of two cue-summing busses (a *buss* is any wire carrying line-level signals, i.e., any signal higher than mike output level). The signals fed to each buss are combined in an ACA and used to provide a mix for the musicians' headphones. By using cue 1 for the left earphone and cue 2 for the right one, the engineer can provide the musicians with a stereo headphones mix.

The *echo send control* determines the amount of signal sent to the input of external echo (reverberation) units through any or all of the four echo send on/off buttons. Since the number of separate echo devices in a studio is limited by their high cost, the echo send on/off switches connect to four separate mixing busses so that more than one signal can be sent to each echo unit. Each buss connects to an ACA which forms a composite echo send signal and drives a booster amplifier through an *echo send submaster*. The output level of each booster amp is monitored by a VU meter. Each booster is normalled to a different echo device. The submaster and VU meter are used to adjust the level of the echo send mix to read near 0 VU for best signal-to-noise ratio and lowest distortion in the echo device.



(A) Too low. (B) Too high. (C) Readings within the dotted lines are acceptable recording levels.

Fig. 6-15. VU meter readings.

The pre/post button determines whether the signal applied to the echo send pot is obtained before or after the input fader. If before the fader (PRE position), varying the input fader setting has no effect on the echo send signal level. Raising the input fader level increases the ratio of direct to echoed sound and makes the signal appear closer to the listener. Lowering the input fader reduces the amount of direct sound and makes the signal appear farther away. If the echo signal is obtained after the input fader (*post* position), changing the fader setting varies the echo send signal by the same amount. Thus, the direct to echoed sound ratio remains constant, the signal remains at the same apparent distance and changes only in volume.

Depressing the *solo* button (s) cuts off the normal signal feed to the monitor speakers and replaces it with the signal present in the input fader of that module. If the pan button is on, the left-right positioning of the soloed signal in the speakers is determined by the pan-pot setting; otherwise, the signal is centered between the left and right speakers.

Soloing a signal enables the engineer to hear exactly what is being recorded by eliminating the masking effects caused by the presence of other signals. When soloed, distortion or other undesired sounds in an input channel can be detected easily.

Solo buttons can be used at any time without interrupting the signals sent to the headphones or recorded on tape. By pressing additional solo buttons, as many signals as desired can be soloed simultaneously. The volumes of the individual soloed signals are controlled by their input faders, while the overall solo volume is controlled by the solo master fader.

The mike/line switch determines whether the output of a microphone or a track of tape is fed to that input channel. In the line position, the signal from the tape passes through a 30-dB pad and then into the preamp which is set for a fixed gain of 24 dB (the mic trim control is disconnected in the line position). From this point on, the tape signal follows the same path as a mike signal. The line position is used during mixdown sessions and during record and overdub sessions where the producer wants to hear EQ on an existing track.

The Model 2824 console is supplied with up to 28 identical input channels. Line inputs 1 through 16 are usually connected to a 16-track tape machine outputs, with 17 through 20 connected to a 4-track machine, 21 and 22 to a 2-track machine, 23 to a mono machine, and 24 through 28 left as spares. These connections, however, are left to the discretion of the studio.

The outputs of the studio echo devices are normalled to four echo control modules known as *echo return* modules (Fig. 6-16). The module provides the *echo program send* submaster previously mentioned to control the overall signal sent to the echo device, as well as a rotary CHAMBER RETURN level control. The most frequent use of the echo return signal is to mix it with the dry (nonechoed) signal. To add echo to a voice, the voice signal could be assigned to track one of the tape and to the echo device. The echo return module for the echo unit in use would then also be assigned to track one of the tape via the echo return channel selectors. The tape would record a mixture of dry and reverberated sound. The relative positions of the input fader and the echo return pot would determine the amount of echo heard on the voice by varying the ratio of direct to reverberated sound. The module has provision for an external delay unit to be used in conjunction with the echo chamber. This unit delays the echo send signal in order to provide control of the time of first reflection of the sound. The external delay unit can function by itself for slap-echo and repeat echo effects.

The TAPE control sets the level of the delayed signal sent to either the chamber input or returned to the echo module output as determined by the adjacent send/return button. The DELAY-REVERB control enables the output of the delay unit to be fed back to its input

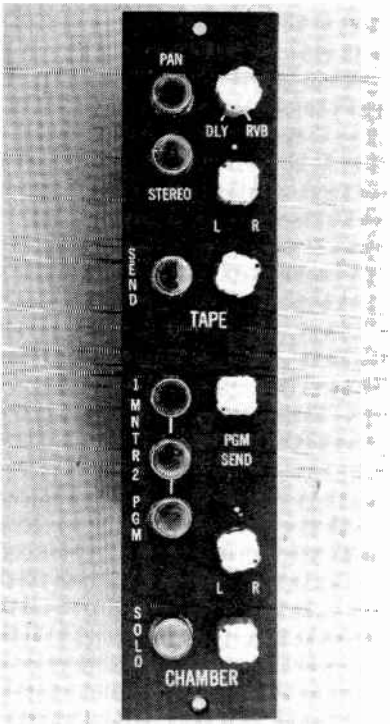


Fig. 6-16. The API echo return module.

Courtesy Automated Processes, Inc.

at any desired level. As the position of this control is varied in the clockwise direction, the output of the delay unit varies from a single delayed repeat of the input signal to greater and greater numbers of repeats, until a feedback oscillation occurs (Fig. 6-17).

The echo return module has separate pan pots for the chamber and the delay echo devices. The PAN button connects these pan pots to any odd/even combination of output channels selected on the

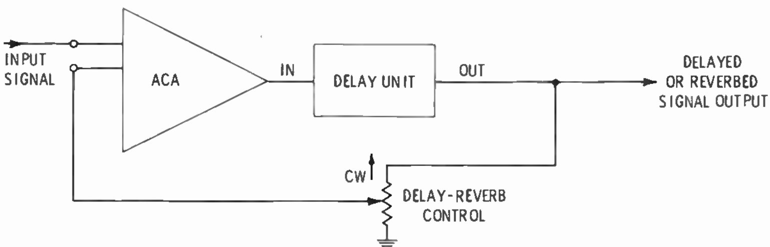


Fig. 6-17. Schematic diagram of a delay feedback system.

echo return output assignment matrix. The STEREO button feeds the outputs of the pan pots to the stereo mixdown busses. The program, monitor 1, and monitor 2 buttons determine the source of the echo send signal, and they determine to which busses the echo return is connected.

In PROGRAM, the echo send signal is derived from the input modules; the echoed signal is returned to the program through the echo return module and channel selectors.

In MONITOR 1 and 2, the send signal is derived from the echo send controls on the multitrack monitor module. The return is connected to the monitor system through the controls and switches on the monitor control panel to be discussed later in this chapter.

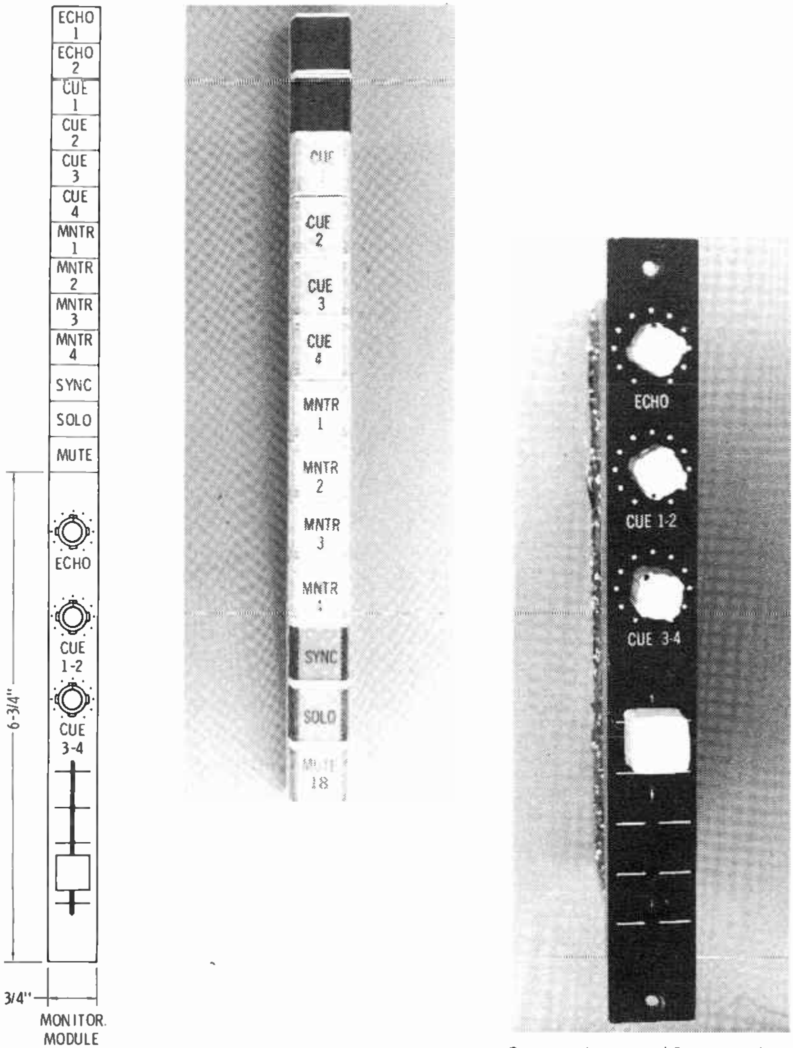
The SEND and RETURN buttons on the echo VU Panel determine whether the echo VU meter reads the level of the signal being sent to the echo chamber or the level of the signal at the output of the chamber.

The multitrack monitor mixer consists of 24 monitor modules (some are shown in Fig. 6-18). The modules derive their input signals from the console or tape machine outputs as determined by whether the *program* or *tape* mode is selected on the *monitor master control* panel (Fig. 6-19).

The MASTER SYNC button is used with the program button to permit tape playback on those tracks preselected by the individual sync buttons on the monitor modules in combination with program output from those channels not preset into sync. Using the master sync button together with the tape button plays back only those tape channels not preset into sync. This enables newly overdubbed tracks to be heard without the basic tracks.

The monitor modes are used as follows:

1. To record basic tracks, the program mode is selected and the console outputs (which are the signals being sent to the master tape) are heard.
2. To listen back to the recorded performance, the tape mode is selected and the monitor is fed the tape playback signals.
3. To overdub one or more instruments, the individual sync buttons on the monitor modules are pressed corresponding to the tracks for which tape playback is desired (i.e., the basic tracks) and program and master sync are selected on the monitor master control panel. Tape playback is heard on the channels set to sync, while microphones (console outputs) are heard on the others.
4. To check the performance of the newly recorded tracks only (without the basics), tape and master sync are selected and the modules set to sync are muted.

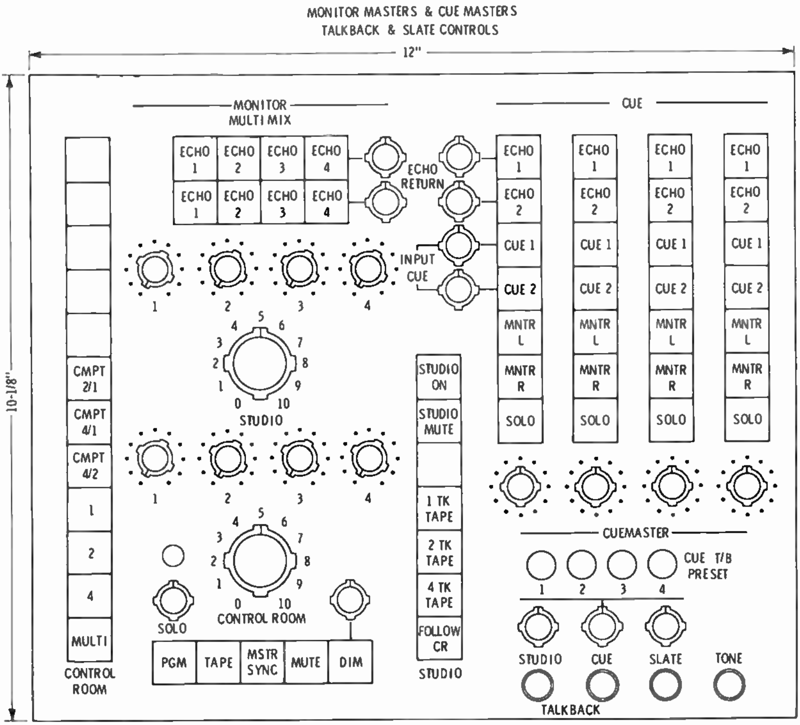


Courtesy Automated Processes, Inc.

Fig. 6-18. The multitrack monitor mixer used on the API Model 2824 console.

5. To play back all the tape channels together, the tape mode alone is selected. The console meters follow the monitor switching so that the engineer can see both the level of signals being recorded as well as those already on tape.

Each monitor module consists of a miniature level control fader, two cue level controls, an echo send control, and thirteen push but-

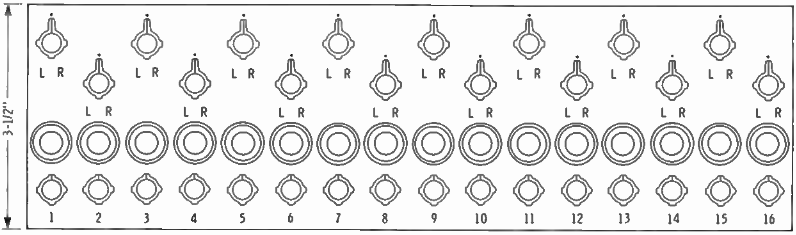


Courtesy Automated Processes, Inc.

Fig. 6-19. The API monitor master control panel.

tons. The ECHO 1 and ECHO 2 push buttons select either or both of the two available echo devices assigned to the monitor system. The echo control determines the level of the echo send signal. The CUE buttons enable the signal in that module to be sent to any or all of the four headphones mix busses, with the level for busses 1 and 2 set by one control and the level for busses 3 and 4 set by the other. The MONITOR buttons enable the signal in a module to be assigned to any or all of the four monitor speaker mix busses at a level determined by the monitor fader. The SYNC button is used in overdubbing, as described in the last paragraph. The SOLO button mutes all sources not soloed and feeds signals in the soloed module(s) to the monitor speakers selected by monitor buttons 1 through 4. The MUTE button shuts off all the outputs of the module.

The *simultaneous stereo mixdown* panel (Fig. 6-20) contains sixteen rotary level controls, on/off switches, and pan pots which can be used to create a mix to feed the monitor speakers, the headphones, or a two-track recorder for recording a rough mix while overdubbing or recording basic tracks. The level controls are fed from the same source (program or tape) as the corresponding monitor module. The stereo output of this panel is fed through a stereo master fader and is displayed on the console stereo VU meters. The two outputs are also mixed together in an ACA to create a mono signal which is displayed on the console mono VU meter. The stereo and mono signals are available at jacks in the patch bay. The stereo master fader is also accessible at the patch bay for use in mixdown sessions.



Courtesy Automated Processes, Inc.

Fig. 6-20. The API simultaneous stereo mixdown panel.

In addition to the program, tape, and master sync functions previously mentioned, the monitor master control panel has several other functions. The control room source-selector buttons determine whether the console is connected to a 1-, 2-, 4-, or multitrack (8-, 16-, or 24-track) tape recorder.

The *CMPT 4/2*, *4/1*, and *2/1* buttons test the compatibility of quad programs in stereo or mono and of stereo programs in mono. The 4/2 compatibility is checked by combining the front and rear channels for each side of the mix at equal levels and displaying them on the left and right monitor speakers. The compatibility of 4/1 and 2/1 are checked by mixing the channels at equal levels and displaying them in mono.

By switching the compatibility tests on and off, the producer and engineer can hear how the balance changes when the mix is played in different formats and can adjust the mix to compensate for any unacceptable changes. Several spare selector buttons are provided for playing additional sources, such as a cassette machine or turntable, through the control-room speakers.

The studio source-selector buttons enable the monitor speakers in the studio to play back either the same signal feeding the control-room speakers, or a 4-, 2-, or 1-track tape. For example, the studio speakers can play back a quad mix of one song while a 16-track tape of a different song is being played in stereo over the control-room speakers. The studio speakers are muted when the monitor mode is set to program in order to prevent feedback and leakage into the microphones. The STUDIO ON button must be pressed for the studio speakers to operate in the tape playback mode. The STUDIO MUTE button is used to shut them off.

The monitor echo return selector buttons and echo return level controls permit echo to be heard in any or all of the four monitor speakers without recording it. The producer and engineer leave themselves more mixdown flexibility when the tracks are recorded dry; for it is impossible to completely remove the echo mixed with a signal once it is recorded.

The cue echo return selector buttons and level controls enable the musicians to hear the echoed signals through any or all of the headphones mix busses. The input cue selector buttons and level controls allow the cue mix from the input modules to be mixed in with that from the monitor modules. The monitor cue selectors enable the multitrack monitor mix to be fed to the headphones.

The MONITOR LEFT buttons select the combined output of the left-front and left-rear monitor busses. The MONITOR RIGHT buttons do the same for the right-front and right-rear busses. The cue solo buttons enable the cue mixes to be soloed through the monitor speakers so that the engineer and producer can hear the same mix the musicians hear without putting on headphones.

Each cue buss has its own master level control to set the overall headphones level. The studio and control-room speakers have individual speaker level trim controls as well as master level controls. The trim controls enable the outputs of the four speakers to be balanced so that applying the same drive signal to each speaker individually produces equal sound-pressure levels behind the console. The trimmers compensate for any differences in efficiency between the four speakers. They also counteract the effect on the engineer and/or producer of being closer to some speakers than to others.

The solo control sets the monitor level of all solo functions. Depending on the monitor and program mix, the settings of the submasters, and the number of sources soloed at the same time, the difference in level between soloing and normal monitoring can be considerable. For example, if input faders are set high and submasters are set low, the volume of a soloed input channel is much greater than the normal monitor level, with painful results. The solo control compensates for this without affecting normal monitoring levels. The solo status light illuminates whenever any of the console solo buttons are activated, thereby warning the engineer that the normal monitor feed has been overridden.

The studio TALKBACK button and level control permit the engineer and producer to speak to people in the studio. The control-room talkback mike is mounted on the console, and its amplified output is routed through the studio playback speakers. The mike has a gain trim control to adjust its sensitivity to that needed for good communication. The cue talkback button, level control, and buss selector presets permit talkback into the headphones only.

Any or all four of the headphones systems can be set to hear the talkback mike. This feature is useful in giving musicians cues during a recording, for it eliminates the danger of the talkback leaking into the studio mikes or distracting musicians who do not need the cues. It also provides a means of getting the musicians' attention if they cannot hear the studio speakers due to the volume of the amplified instruments, or if they are wearing headphones which isolate them from outside sounds.

The control-room speaker level is automatically *dimmed* (attenuated) when either talkback button is pressed to prevent acoustic feedback through the studio mike to control-room speakers to talkback mike to studio speakers (or headphones) path. Dimming the speakers also increases talkback intelligibility by cutting down on the level of any instruments which are playing while communication is taking place, decreasing the noise the talkback mike might pick up. Since the control-room speakers are not cut off completely, two-way conversation is possible (using a live mike in the studio) while the talkback button is pressed.

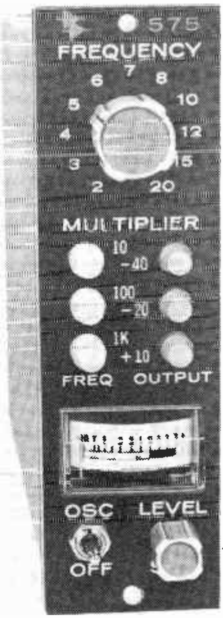
The slate button and level control feed the talkback mike to the console outputs for recording on tape to identify different selections or *takes* of the same selection. Pressing this button disconnects all other program sources from the console outputs. The TONE button also disconnects the program sources and feeds the output of the built-in audio oscillator to the console outputs. In addition, it also reduces the monitor system level so that the tone is not heard at earsplitting volume. If the beginning of each take is marked with a low-frequency tone, the engineer can easily return to that spot by listening to the tape and watching the tape machine VU meters as the tape is being wound at high speed. The high speed raises the pitch of the recorded program to a shrill whine with much of its energy above the frequency response of the human ear and the VU meter.

A low-frequency tone (20 or 30 Hz) is raised to the center of the audio spectrum where it stands out clearly both to the ear and on the VU meter. Since both human hearing and most monitor speakers are less sensitive at very low frequencies, the playback of the tone at normal speed will not be very audible. A higher pitched tone would be very disturbing and painful if played at high levels over the speakers or through the headphones. Care must be taken in setting the level of the tone if it is to be recorded on more than one track, for the signal output of all the tracks is added in proportion to the monitor fader settings. The results can be very loud if a tone is recorded on sixteen tracks at 0 VU.

The MUTE button cuts off all output from the monitor speakers, while the DIM button merely reduces the level by an amount determined by the setting of the dim control. These controls are used to permit conversation in the control room without having to reset monitor levels.

The console oscillator is an API Model 575 (Fig. 6-21). Thirty-one different frequencies between 20 Hz and 20 kHz are available through the use of its rotary selector switch and frequency multiplier push buttons. Its output level is varied with the level set control and the output range buttons; the level can be read on the calibrated meter. The on/off switch can be wired to shut off the oscillator or to let the oscillator run and merely turn its output on and off so that it is instantly available for use.

The console is supplied with two API Model 525 compressor/limiters which are patchable to any point in the console (Fig. 6-22). The attack time is fixed at 15 microseconds, while the release time is frequency dependent with signals below 300 Hz causing longer release times than those above this frequency. The high-frequency



Courtesy Automated Processes, Inc.

Fig. 6-21. The API Model 575 oscillator.

Courtesy Automated Processes, Inc.

Fig. 6-22. The API Model 525 compressor/limiter.

release time is adjustable via two push-button switches to 0.1, 0.5, 2.0, and 2.5 seconds. The combination of preset release times for the high frequencies and variable release times in the presence of low frequencies helps to prevent pumping and breathing from being introduced to the program. A 2:1 compression or 20:1 limiting is selected by depressing the C or L buttons, respectively. An OFF button is provided which disables the gain-reduction function, making the 525 a unity-gain amplifier. This button can be operated during a recording to insert compression or limiting on only part of a program, without introducing any switching noise. The input control sets the unit threshold level, while the output control sets the output level. The CEILING control adjusts the gain reduction in 2-dB steps up to 20 dB of compression or limiting while maintaining a constant peak output level, so that changing the amount of compression or limiting does not make readjustment of the output level necessary, as it does with most compressors and limiters.

The D-S button inserts a filter into the control circuit to cause more gain reduction for high-frequency signals than for low frequencies in order to reduce sibilance on voices. The amount of gain reduction is indicated on the unit meter. The control circuits of the two units can be connected for use with stereo program material by a switch just below the console talkback mike. Switching to the STEREO position prevents changes in channel balance which would otherwise result from different amounts of gain reduction in the two channels. The stereo mode causes equal compression or limiting in both channels regardless of which one triggered the gain reduction. The console is prewired for four additional compressor/limiters. The MULTI position of the limiter mode switch interconnects all of the units for use with quad or other multichannel programs. In the MONO position, the units are independent of each other.

The console is also prewired for two quad pan pots, for a quad master fader to fade quad mixes with one control and for EQ on each of the four echo send or return busses. The console amplifiers are powered by a regulated supply which is protected against overload, short circuits, and high power-line voltages. The relays and lamps are powered by a separate regulated supply to eliminate any possibility of switching noise getting into the console output.

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CHAPTER 7

Noise-Reduction Devices

TAPE LIMITATIONS

The dynamic range of music approaches 120 dB while the dynamic range of magnetic recording is limited by its signal-to-noise ratio of about 60 to 65 dB. The limitations imposed by the tape recorder are due to distortion caused by tape saturation when the recorded level is too high and tape noise which intrudes when the recorded level is low (Fig. 7-1). The desire to reduce the noise level heard in playback causes engineers to either record at high levels and live with some distortion or change the dynamic range of the signal. Neither of these methods approaches the ideal storage medium, i.e., a device which can store a signal and play it back so that it is identical to the input signal. In practical audio work, it is sufficient to settle for an output signal which is indistinguishable from the input signal to the trained ear.

The types of noises to be eliminated can be classified as: hum, crosstalk between tape tracks, print-through, clicks caused by making splices with magnetized razor blades, pops due to tape recorder bias circuits turning on and off too quickly, buzzes, tape and amplifier hiss, and modulation noise. *Modulation noise* is high-frequency noise which causes fuzziness of waveform and is due to the

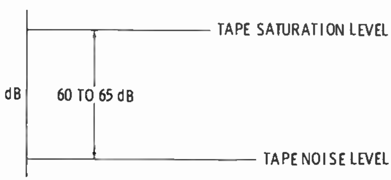


Fig. 7-1. The dynamic range of tape.

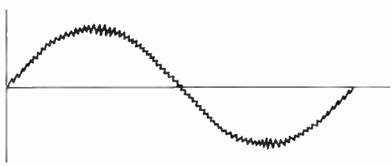


Fig. 7-2. Modulation noise on a sine wave.

irregularities in the coating of magnetic tape (Fig. 7-2). It is present only when a signal is present and increases in level as signal level increases. It is often called *behind the signal noise* or *asperity noise.*

There are several obvious methods to eliminate these noises: (1) use amplifiers that have very little hum, buzz, and hiss; (2) design the tape heads and electronics so that there is very little crosstalk between the channels; (3) use demagnetized razor blades in making splices; (4) include pop-suppression devices in the bias current circuits; (5) increase the width of the tape track and/or increase the tape speed so that higher flux levels can be recorded; (6) make tape that plays back with no hiss, modulation noise, or print-through.

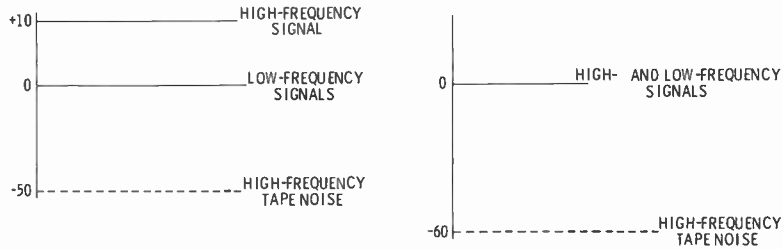
The first four of these methods can presently bring their respective noises down to the point where they are not the limiting factor in the signal-to-noise ratio. The fifth method of improving signal-to-noise ratio is impractical because doubling the track width or the tape speed gives only a 3-dB improvement, and a tenfold increase would be needed to better the signal-to-noise ratio by 10 dB. Making tape that does not contribute hiss, modulation noise, or print-through is where the noise problem rests. Low-noise tapes can be made by making the size of the oxide particles very small, but this is limited by the minimum size of a particle needed to form a magnetic domain with a particular oxide formula. Better signal-to-noise ratios can also be obtained by making tapes that have higher output levels; that is, they can store higher levels of magnetic flux before saturation, but the presence of this high flux level increases the amount of print-through.

While waiting for tape manufacturers to make the perfect tape, several methods have been developed to reduce the obtrusiveness of the noises introduced by the tape. The noise gate can eliminate noise that occurs before and after a signal but not noise that occurs while a signal is present. Several other methods are the Dolby System, the DBX system, the Burwen Noise Eliminator, digital audio recording, and the Burwen Dynamic Noise Filter.

THE DOLBY SYSTEM

Two different Dolby systems are available. The A-type system is used professionally and provides 10 dB of noise reduction below 5 kHz, increasing gradually to a maximum of 15 dB at 15 kHz. The A system will reduce any noise induced in the signal between the record processor output and the playback processor input. The B-type system is designed for the consumer and only reduces hiss. It has no effect on low-frequency noises such as hum, rumble, and pops. The noise reduction is 3 dB at 600 Hz and rises to 10 dB at 5 kHz, where it levels off.

The Dolby system makes use of the same concept as the NAB pre- and post-equalization used with standard tape recording. The function of the NAB EQ is to boost the high frequencies before they are recorded (Fig. 7-3A) so that when they are played back their level can be reduced. Since the high-frequency tape hiss was not boosted by the record EQ (because it is generated by the tape itself), the playback high-frequency cut decreases the level of the tape noise as it brings the level of the high frequencies back to normal and the high-frequency tape noise is reduced (10 dB in Fig. 7-3B). The pre-



(A) High frequencies are boosted by 10 dB. (B) High-frequency signal level reduced to original level by the post-EQ.

Fig. 7-3. Noise reduction by pre- and post-equalization, using 10 dB of high-frequency EQ.

and post-EQ of the tape recorders is set to one standard curve at a time (NAB or CCIR), and because the EQ is fixed, it must enable the recorder to handle the worst-case conditions present in recording. This means that the tape must not be overloaded by the pre-emphasis when recording any sound source, such as the extreme lows of organ pedals or the extreme highs of a cymbal crash. Because saturation must be avoided in these worst-case conditions, the EQ cannot provide the optimum boost for each type of program material. For example, an NAB recording of a piano or a viola does not make full use of the tape storage capacity in the high-frequency range because the higher harmonics of the piano and viola are much softer than their fundamentals, and most of their output is in the midrange. Thus, the high-frequency pre-EQ could be boosted a lot more than the NAB standard without overloading the tape. To play the tape properly, the post-EQ would have to cut the signal by the same amount the pre-EQ boosted it. This is, in effect, what the Dolby system does. It provides an automatic means of making full use of the tape storage capacity at all times, based on the level of the signal fed to the Dolby input.

Variable tape loading is achieved through compression in recording and expansion in playback. Compression is achieved by increasing the gain of low-level signals rather than by decreasing the gain

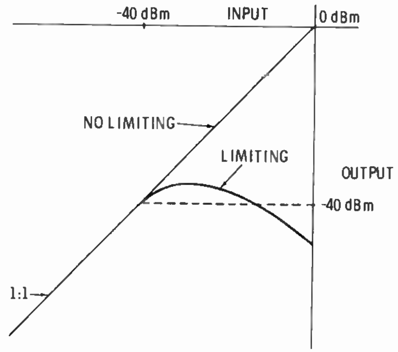


Fig. 7-4. Output of the limiter used in the Dolby system rises slightly above −40 dBm.

of the loud signals. The circuit uses a limiter which prevents the signal at its output from rising much above −40 dBm (Fig. 7-4). The output of this limiter is then added algebraically to the uncompressed input signal (Fig. 7-5). Since the output of the limiter only rises slightly above −40 dBm, the effect of the addition of this signal to the input signal depends on the input signal level. When the input signal is low, the output of the limiter is large in comparison and adding the two, results in a boosted signal. When the input signal is high, the limiter output is small in comparison and its contribution when the two signals are added is negligible.

Expansion follows the same procedure except that the output of the limiter is subtracted from the input signal, which reduces the gain at low levels (Fig. 7-6). What occurs, in effect, is that an extra component is added to the signal when it is recorded, and the same component is subtracted when the signal is played back.

If the signal is processed exactly as described above, full noise reduction would only occur at low signal levels, while at high levels the noise would have its normal value. In addition, because the level

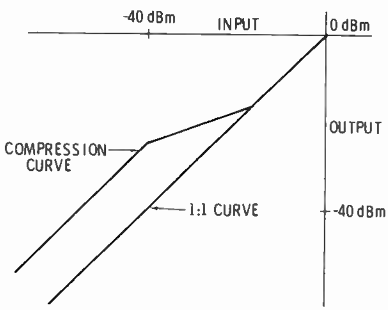


Fig. 7-5. The Dolby compression curve.

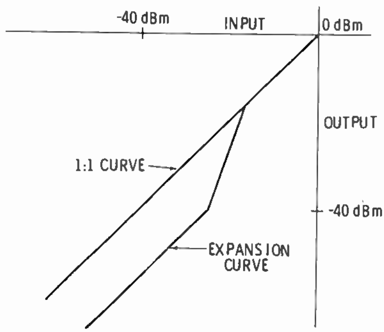


Fig. 7-6. The Dolby expansion curve.

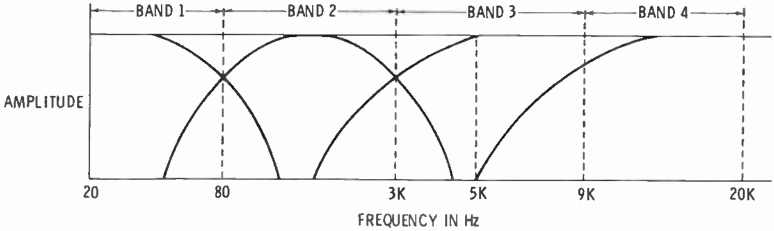


Fig. 7-7. The four filter bands of the A-type Dolby system.

of the noise would be changing, swishing or breathing sounds would be heard. The Dolby system takes advantage of the masking effect of the ear to avoid these unacceptable results. Since the ear cannot hear noise in the same frequency range as the signal, only noise frequencies outside of the signal bandwidth are responsible for the swishing and the noise heard during loud passages.

The A-Type System

In order to reduce these noise frequencies, the audio spectrum is divided up into four bands in the A-type system. Each band of frequencies has its own limiter so that the presence of a loud signal in one band does not defeat the noise reduction in the other bands. The four bands are: (1) 80-Hz low pass (80 Hz and below); (2) 80-Hz to 3-kHz bandpass; (3) 3-kHz high pass (3 kHz and above); and (4) 9-kHz high pass (Fig. 7-7). The outputs of the four filters and limiters (Fig. 7-8) are combined in such a manner that low-level

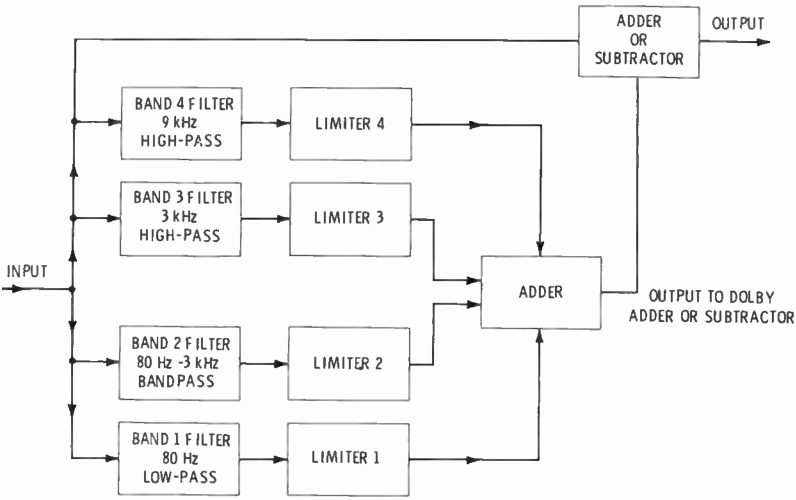


Fig. 7-8. Block diagram of the filter/limiter networks.

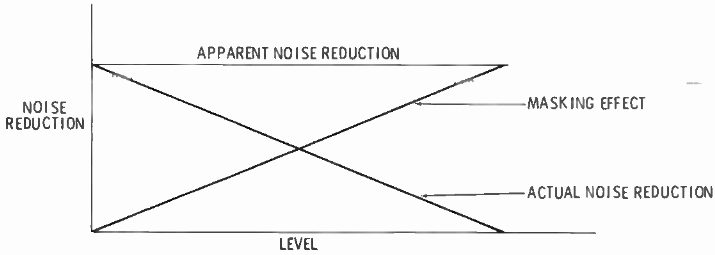
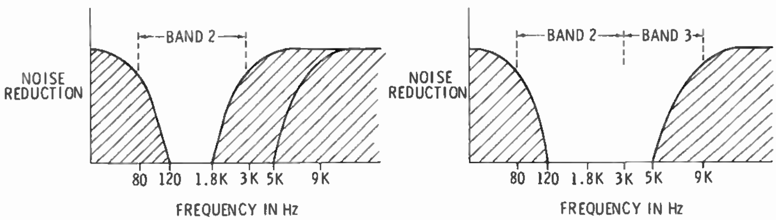


Fig. 7-9. The masking effect and the actual noise reduction combine to keep the apparent noise level constant.

signals (below −40 dBm) are boosted 10 dB from 20 Hz to 5 kHz with the boost rising gradually between 5 kHz and 15 kHz to a maximum of 15 dB. As the level of a signal in one band rises, its noise reduction decreases, but the effectiveness of the masking effect increases so the noise level appears to remain constant (Fig. 7-9). The bands are not sharply defined, so when the Band 2 noise reduction is disabled by the presence of loud signals between 80 Hz and 3 kHz, some noise reduction (in addition to masking) is provided up to about 120 Hz by Band 1 and also above 1.8 kHz by Band 3 (Fig. 7-10A). If Band 3 also has its noise reduction turned off by loud signals between 3 kHz and 9 kHz, Band 4 contributes noise reduction from 5 kHz up (Fig. 7-10B). Bands 1 and 4 rarely have their noise reduction shut down completely except by very loud organ tones or cymbal crashes, respectively. Although the actual amount of noise reduction throughout the audio spectrum changes from one moment to the next, the noise level perceived by the ear remains constant. The B-type circuit works similarly but uses only a single high-frequency filter/limiter band.

The operation of the Dolby system depends on its use both when a tape is recorded and when it is played back. Each Dolby channel has a record and a play mode, and the processing can be inhibited



(A) Band 2 noise reduction is shut off by a high-level signal between 80 Hz and 3 kHz.

(B) High-level signals between 80 Hz and 9 kHz shut off the noise reduction in both bands 2 and 3.

Fig. 7-10. Demonstrating noise reduction in the A-type system.

through the use of the *noise reduction in/out* switch. When this switch is in the out position, the Dolby becomes a unity gain amplifier. The modes are switched by controlling the use of the combined signal output of the filter/limiters. This output is added to the input signal in the record mode, subtracted from the input signal in the play mode, and is not used when the noise-reduction switch is in the out position.

Each track of a tape is assigned a separate Dolby unit. The same unit that processes the signal for recording is used for playback processing by changing its input and output connections whenever the processing mode is changed. In the record mode, the console output is connected to the Dolby input and the Dolby output is connected to the input of the appropriate track on the tape machine (Fig. 7-11). In the play mode, the tape machine signal output is

Fig. 7-11. Connecting the Dolby system for record processing.

connected to the Dolby input and the Dolby output is connected to the appropriate console monitor and line inputs (Fig. 7-12). Depending on the model of Dolby used, the connections are changed either manually by unplugging one set of cables and plugging in another set, or automatically by relays or solid-state switches which are activated by the mode switches. The mode and signal connections for units of the latter type can be remotely controlled by the record relay voltages of the individual tracks of the tape machine. Each Dolby remains in the play mode until its tape channel is punched into record, at which point its processing mode and signal connections change to record.

Each Dolby is initially set up so that a 0-VU level signal on tape plays back at the *Dolby level* indicated by the NAB mark on the Dolby front-panel meter. This ensures that signals below the threshold during recording will also be the same amount below the threshold on playback. A ±3-dB tolerance exists before a difference

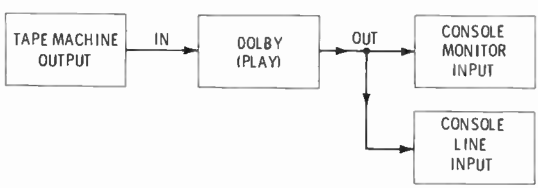


Fig. 7-12. Connecting the Dolby system for playback processing.

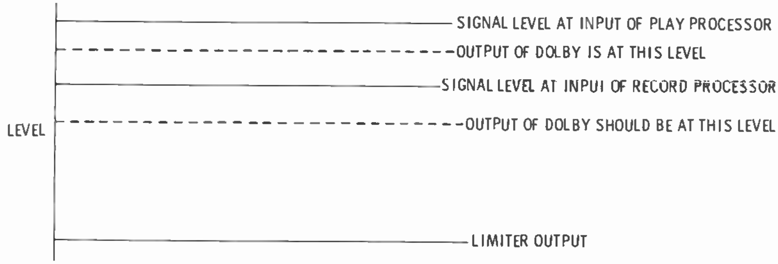


Fig. 7-13. The tape output is fed to the Dolby playback processor at too high a level.

in the signal played back would be very noticeable. If the signal is played back too loud through the Dolby, too little expansion takes place because the limiter is blocked by the high level and the signal sounds compressed and overly bright (Fig. 7-13). If the signal is played back too soft, the expander expands too much, and the signal sounds dull and has too great a dynamic range (Fig. 7-14). The signal record level does not matter as long as the playback level is the same.

In order to ensure that the tapes are played back at the same level at which they were recorded, a 400- or 700-Hz, 0-VU level tone is often put on the tape just before the program so that the repro level can be adjusted properly when the tape is played back. This is especially important if the tape is to be interchangeable with those made at other studios which might set their machine playback levels differently. Some Dolby units have built-in oscillators with which a reference tone at Dolby level can be recorded on tape. The oscillators have a pulsating characteristic which identifies the tape as being Dolby processed.

The Dolby system can be used in any audio storage or transmission chain if the signal is available for processing at both ends. It can be used for discs, telephone lines, radio, etc. The record processed signal is referred to as the *stretched*, *encoded*, or *unrestored* signal. The playback processed signal is called the *unstretched*, *decoded*, or *restored* signal. An unrestored signal is not distorted. Its waveform is the same as the original signal, only its amplitude is

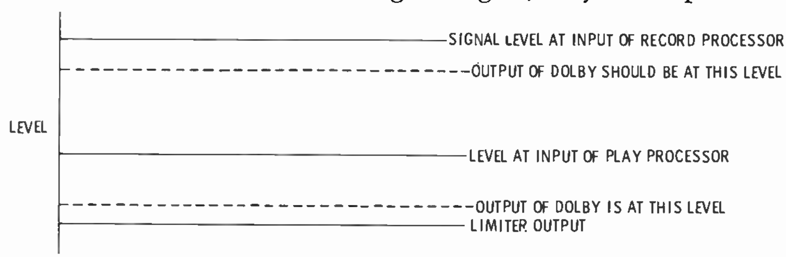


Fig. 7-14. The tape output is fed to the Dolby playback processor at too low a level.

changed. If it is stretched, the signal sounds very bright and compressed, but it is still intelligible.

When making a stretched copy of a stretched tape, no processors are necessary; the signal retains its noise-reduced quality and the copy can be made directly from machine to machine. The same amount of noise increase occurs in making tape copies of a stretched tape whether it is decoded and then re-encoded (Fig. 7-15A) or transferred direct from machine to machine without processing (Fig. 7-15B).

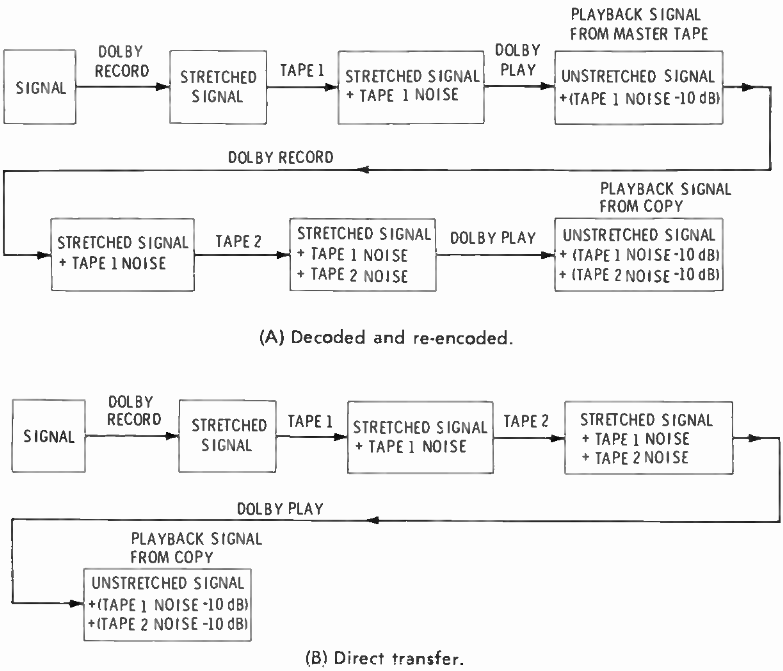
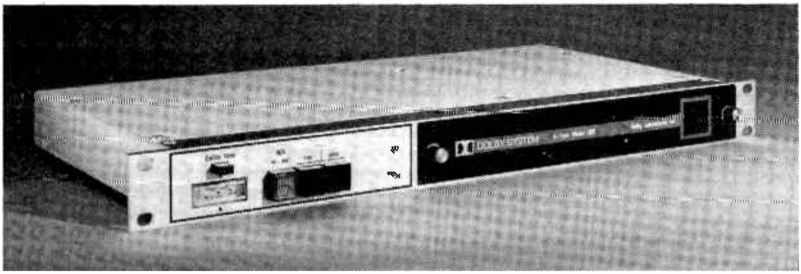


Fig. 7-15. The same amount of noise increase occurs in making tape copies of a stretched tape.

If it is desired to equalize a stretched signal, or mix a stretched signal with another stretched signal or with an unstretched signal, all of the signals must first be restored to the unstretched state. If the mix is to be stretched, it should then be reprocessed through a Dolby record processor. If stretched signals are compressed, equalized, or changed in level through mixing, the signals may be changed to the point that they cannot be unstretched properly. In practice, stretched signals can be mixed and equalized but only on a trial and error basis. That is, the desired effect might be achieved in one case, but not in another, depending on the change of level and EQ used.

Dolby A systems are available in several models: the A301, the A301S, the 360, the 361, and the M series. The Model 360 is shown in Fig. 7-16. The A301 is a two-channel, manually switchable record or playback processor. The two channels have 80 dB of isolation between each other, so they can be used for completely separate programs if desired. The A301S is the same unit as the A301 except that it only has electronics for one channel. The 360 is a single-channel, manually switchable record or playback processor, which is considerably smaller than the A301 but is fully compatible with it.



Courtesy Dolby Laboratories, Inc.

Fig. 7-16. The Dolby 360.

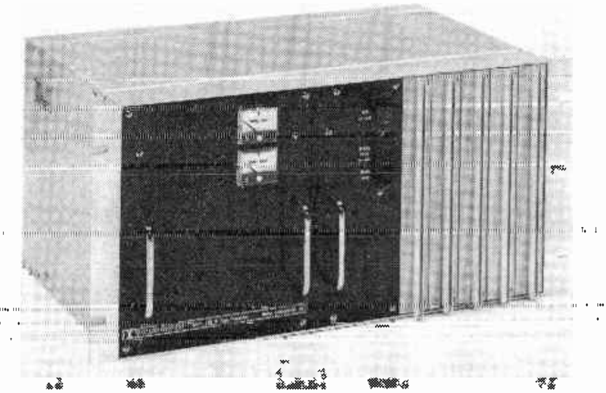
The 361 is the same as the 360 except that it has automatic record/play mode and connection changeover facilities which can be controlled by the tape machine. The M8, M16, and M24 units are designed for use with 8-, 16-, and 24-track recorders, respectively, and use the same plug-in printed-circuit cards as the 360 series, combined with solid-state, record/play connection switching and a power supply common to all channels to reduce the size and cost of multitrack noise reduction.

The B-Type System

Dolby makes only one B-type processor, the Model 320, which looks very much like the A301 (Fig. 7-17). It is designed for use in making master tapes for high-speed duplication of cassettes and reel-to-reel tapes. All other B-type processors are made by manufacturers under license from Dolby.

THE DBX SYSTEM

The DBX noise-reduction system (Fig. 7-18) is a compressor/expander system which provides twice as much noise reduction as the Dolby system. Noise is reduced to 20 or 30 dB below what it would otherwise be. This system can also be considered a means of loading the tape more effectively than the NAB curves. It is connected between the console and the tape machine in exactly the

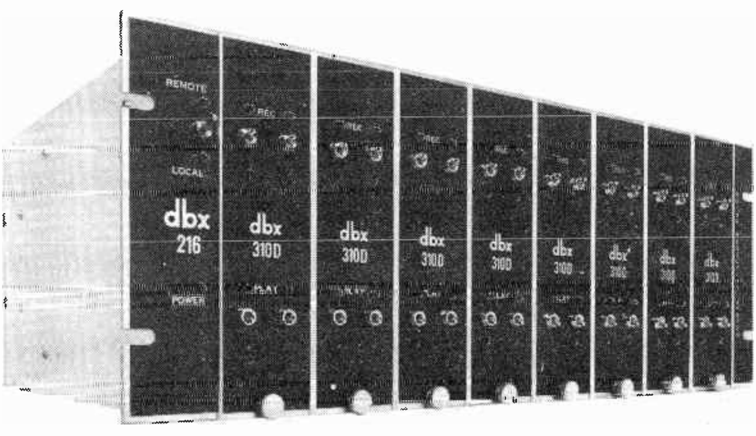


Courtesy Dolby Laboratories, Inc.

Fig. 7-17. The Dolby Model 320 B-type processor.

same way as the Dolby System. The compressor has a 2:1 ratio between −90 and +25 dBm with the unity gain point occurring at 0 VU. Unlike the Dolby system, all signals are compressed, not just low-level signals.

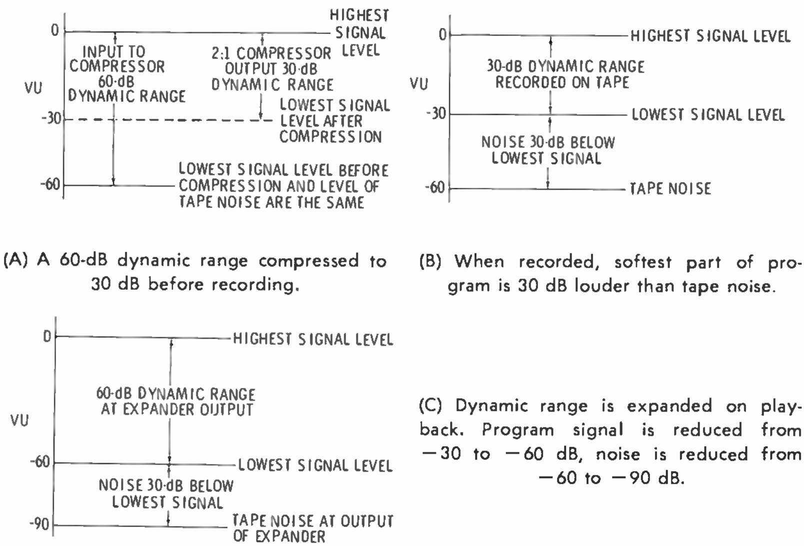
The noise reduction occurs as follows (Fig. 7-19): Assume that there is a 60-dB signal-to-noise ratio in the tape recorder and a 60-dB dynamic range program is recorded. During the softest passages of the program, the noise added by the tape is just as loud as the program, and it is therefore very audible. With the DBX system, the



Courtesy DBX, Inc.

Fig. 7-18. The DBX Model 216, a 16-channel simultaneous record/playback processor.

program passes through the DBX record section and is compressed 2:1 into a 30-dB dynamic range program (Fig. 7-19A). This compressed signal is then recorded on the tape where it is, in effect, mixed with the tape noise. The tape noise, however, is now 30 dB below the softest passage of the music (Fig. 7-19B). On playback, the expander reduces the signal level 30 dB on the softest passages and reduces the noise 30 dB at the same time, so the signal is now 60 dB below 0 VU and the noise is 30 dB below that, at 90 dB below 0 VU (Fig. 7-19C); so the noise reduction is 30 dB.



(A) A 60-dB dynamic range compressed to 30 dB before recording.

(B) When recorded, softest part of program is 30 dB louder than tape noise.

(C) Dynamic range is expanded on playback. Program signal is reduced from −30 to −60 dB, noise is reduced from −60 to −90 dB.

Fig. 7-19. Noise reduction in DBX system.

The difference in level between the noise due to the tape and the softest signal recorded on the tape determines the signal-to-noise ratio. In order to achieve more than 30 dB of noise reduction with the same program, the compression ratio would have to be increased. For example, a 3:1 ratio would compress the signal down to a 20-dB dynamic range so it would be 40 dB above the noise level, and 40 dB of noise reduction would result after expanding the signal. The 2:1 ratio was chosen over the 3:1 because of the effect of tape dropouts on the expander. Since the expander reduces its output signal twice as much as the signal played back from the tape, a tape dropout of 2 dB causes a 4-dB dropout of signal with the 2:1 ratio. A 3:1 ratio would cause a 6-dB dropout. A ratio of 1.5:1 would make the dropout problem less noticeable but would do so at the cost of noise reduction (only 20 dB possible). The 2:1

ratio was considered to be the best compromise between amount of noise reduction and sensitivity to dropouts.

The next difference in the DBX system is that it operates on one full frequency range band, from 20 Hz to 20 kHz. Dolby found that having the compressor operate over the full frequency range caused a loss of noise reduction throughout the frequency spectrum in the presence of loud signals, even though they might have a restricted frequency range. The noise not masked by the signal became apparent on loud passages. The DBX system overcomes this problem to a certain extent by using a filter to pre-emphasize high frequencies by 12 dB before the compressor and by using another filter to de-emphasize the high frequencies after the expander. So, even though the signal may be at 0 VU where no compression or expansion takes place, there is an effective 10 dB of noise reduction at the high frequencies. This 10 dB is only achieved when most of the signal energy is below 500 Hz. When the major portion of signal energy rises above 500 Hz, the effect of the filter is diminished and is replaced to a certain extent by the masking effect of the program. A special control circuit for the compressor and expander reduces the gain when the level of high-frequency signals becomes very high, to avoid the tape saturation that would be caused by the high-frequency boost.

An even more important reason for the use of the high-frequency pre-emphasis is the high-frequency modulation noise of tape, which increases in level as the signal level increases. At 0 VU the modulation noise would be very noticeable, and since the expander constantly changes gain, the modulation noise also changes noticeably. The pre-emphasis prevents this from becoming obtrusive, but the variation is noticeable at very high listening levels when only low-frequency signals are present.

Low-frequency pre-emphasis is achieved by making the level-sensing device less sensitive to low-frequency signal components, so that signals with dominant low-frequency energy are compressed less than other signals and are in effect emphasized. This pre-emphasis provides additional noise reduction of hum and other low-frequency noises behind high-level signals.

The main reason for this desensitization is to eliminate low-frequency modulation of the compressor. This could occur if frequencies in the 3- to 5-Hz range (due to subways, trucks, and air conditioning, which can cause buildings to vibrate) are picked up by transmission through the mike stands to condenser or ribbon mikes, which have good low-frequency output. If the level sensor was not desensitized to these frequencies, they would trigger the compressor; but since most professional tape recorders will not pass such low frequencies, they would not be present at the input of the expander to control its gain. Consequently, the background noise would rise and fall at a rate of 3 to 5 Hz. The desensitizing amounts to 3 dB at 50 Hz and 20 dB at 5 Hz.

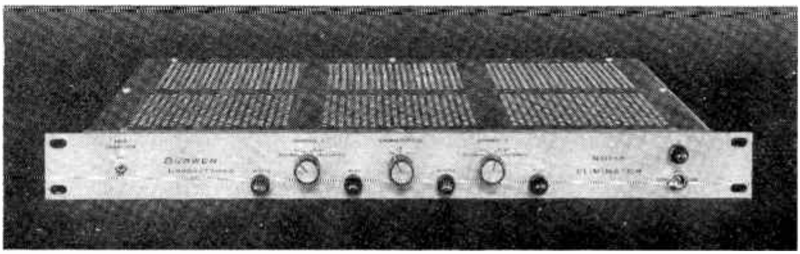
The DBX level sensor computes the true rms value of the input signal and uses this value to determine how much compression or expansion should be applied. The Dolby system uses a combination of peak and a type of averaging (different from rms) to approximate rms sensing. Peak and non-rms average values can be changed by phase shifts and frequency response errors that occur in amplifiers and tape machines. Rms values are also changed by these defects but by a much smaller amount, so that the DBX compressed signal does not lose its control information when passed through amplifiers and tape heads. The Dolby stretched signal level control information can be changed by these devices so that the unstretching may not exactly remove the stretching. As with the Dolby system, DBX tapes can be copied in the encoded state, and the copies played back through a decoder.

THE BURWEN NOISE ELIMINATOR

The Burwen Noise Eliminator Model 2000 (Fig. 7-20) operates on the same principle of full range compression as the DBX system and is connected in the same way as the DBX and Dolby units. It has somewhat more elaborate modes available and is much more expensive per channel than the other two systems.

Burwen's main differences are:

1. The use of a 3:1 compression ratio over only the top 82 dB of its input range, i.e., above −66 dBm and no compression below this level (1:1 ratio of input to output), while DBX compresses over a 115-dB range. This increases noise-reduction capability at the cost of increased dropout sensitivity.
2. Switchable noise-reduction characteristics are available for (A) 15 ips, (B) 7½ ips, and (C) 3¾ ips, 1⅞ ips, fm radio, records, and background music service. The A and B charac-



Courtesy Burwen Laboratories

Fig. 7-20. The Burwen Model 2000 Noise Eliminator.

teristics use the same compression, but B has less record pre-emphasis to prevent overloading 7½-ips tapes. The C characteristic uses the same pre-emphasis as the B, but the compression range and maximum gain are reduced to avoid excess amplification of background noise present in the input signal when the music stops. The C characteristic is designed for the consumer cassette and fm market, and the gains of two or four channels can be interlocked for stereo or quad.

3. The pre-emphasis of the Burwen system also occurs before the compressor to prevent tape saturation. It rises gradually to a maximum of 13 dB at 20 kHz with the A characteristic and 4 dB at 20 kHz with the B and C characteristics. The low-frequency pre-emphasis rises to 5.4 dB at 20 Hz with all three. In the DBX system, pre-emphasis rises to 12 dB between 500 and 2,000 Hz, where it levels off and remains at 12 dB. Thus, the reduction of noise at high levels (modulation noise) is better with the DBX system, which claims 10 dB as opposed to Burwen's 8 dB averaged over all the noise frequencies (the pre-emphasis is greater at some frequencies than others so certain noise frequencies are suppressed more than others) even though the Burwen reduces low-level noise more.
4. The Burwen uses a peak rather than an rms value following level sensor and is therefore sensitive to phase shift and poor frequency response in tape recorders and amplifiers. In order to compensate for these problems, Burwen has added variable low-frequency record EQ and high-frequency playback EQ to the unit to trim the frequency response of the tape recorder to be used so that it is flat. The Noise Eliminator uses a medium rather than a fast attack time to compensate for any high-frequency phase errors. This attack time allows high-level transient signals to saturate the tape for a short time (less than 1 millisecond) before compression takes place. According to Burwen, tests have shown that distortion of the first millisecond of a signal is not noticeable if the high-frequency content of the signal is reduced. This is done through the use of a high-frequency limiter following the compressor. The subsequent distortion products, being mainly high frequencies, are reduced in level by the playback de-emphasis so that they are not noticeable and are less than would occur if the transient was recorded at 0 VU without compression. Since the DBX system uses rms level sensing, it is relatively insensitive to phase shifts and can use a fast attack time with its compressor to prevent the distortion of transients. Therefore, the DBX playback de-emphasis is available entirely to reduce tape and modulation noise rather than to cover up the signal distortion.

Noise buildup when copying tapes is the same whether in an encoded or nonencoded state. The *generation loss* of signal-to-noise ratio cannot be prevented with analog noise-reduction systems. The tape can only start out with a better signal-to-noise ratio so that the noise buildup from successive generations is not too noticeable. The noise buildup from tape alone is approximately 3 dB more than the original for the first generation and 2 dB more for the second. The significance of the amount of noise added by successive generations decreases as the number of generations increases so that after ten generations, the total loss of signal-to-noise ratio is 10 dB.

DIGITAL AUDIO RECORDING

A fourth noise-reduction system which is being developed is called *digital audio recording.* The audio signal instantaneous amplitude is measured at evenly spaced intervals, and the result of each measurement is converted into a *binary word,* which is stored on magnetic tape (Fig. 7-21). The binary code uses combinations of ones and zeroes to form numbers which correspond to the signal level at

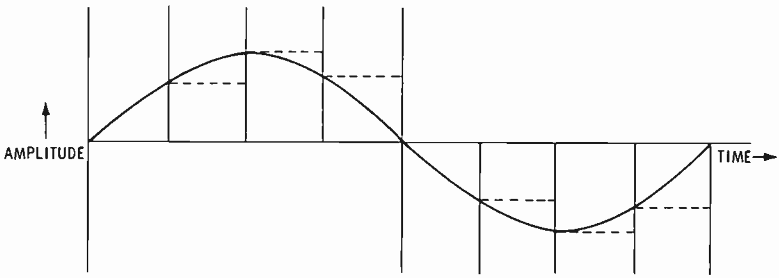
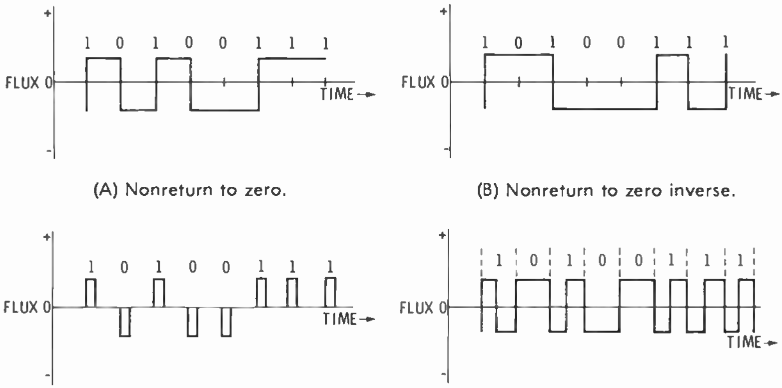


Fig. 7-21. The audio-signal amplitude is measured at each vertical line, and each measurement is converted into a binary word.

that point of time. As opposed to the decimal system, where moving one digit to the left in a number increases the value of the digit by ten times, in the binary system moving one digit to the left increases the digit value by a factor of two. For example, the decimal number $7593 = (1 \times 3) + (9 \times 10) + (5 \times 100) + (7 \times 1000)$, while the binary number 10011 equals $(1 \times 1) + (1 \times 2) + (0 \times 4) + (0 \times 8) + (1 \times 16) = 19$. Binary numbers are often used because they can easily be represented mechanically or electronically by a switch that is in one of two possible positions. *On* is usually considered "1" and *off* considered "0."

There are several formats for recording binary digits on magnetic tape (Fig. 7-22). The boundaries of each bit interval are determined by a timing mechanism called a *clock* which can be either an external oscillator or can be carried within the data format. The *nonreturn to zero* method (Fig. 7-22A) changes the recorded flux polarity whenever the data changes from a "1" to a "0" or vice versa. Successive "1s" or "0s" cause no polarity change. The *nonreturn to zero inverse* format (Fig. 7-22B) changes polarity each time a "1" is written. The *return to zero* (Fig. 7-22C) code records a pulse at each clock interval, leaving the tape blank between the intervals. The "1s" and "0s" are recorded as pulses of opposite polarity. The *biphase* mark (Fig. 7-22D) method produces a flux polarity change at the beginning of each bit. A "1" produces a second polarity change within the bit interval, while a "0" does not.



(A) Nonreturn to zero.

(B) Nonreturn to zero inverse.

(C) Return to zero.

(D) Biphase mark.

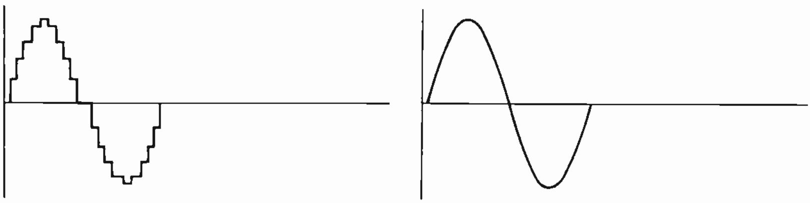
Fig. 7-22. Binary code formats.

The digital audio system records the flux to the tape saturation level to minimize dropouts, for it is the presence and polarity, not the waveform of the flux, that carries the information. Since flux is either present or not present, the distortion characteristics and dynamic range (or signal-to-noise ratio) of the recorder and tape used have no relation to the quality of the audio signal which can be stored on the tape in digital form. The only devices which affect the quality of the signal are the record processor, which converts the signal from analog (continuously varying amplitude) to digital form (varying only between two states), and the playback processor, which converts the digital signal back to analog. The greater the accuracy of these devices, the higher their cost.

The digital process is not completely noise free. Since the words stored on the tape are digital, the smallest signal which can be represented is determined by the number of digits in the binary word. If there are only four binary digits, only fifteen different levels can be represented, so there is necessarily a loss of resolution for any audio signal which falls between two digital levels and must be assigned the closer of the two levels. Increasing the number of digits in the binary word reduces the error which is inherent in the rounding off of the waveform amplitude to the nearest digital quantity. This error is called *quantization error.*

The effective number of bits can be increased by nonlinear quantization methods which divide the word into *ranging bits* and *adjusted magnitude linear bits.* [10]. With an 80-dB dynamic range system, two ranging bits can be used to assign the following bits to one of four 20-dB wide ranges, while the remaining bits represent the signal levels within this range. Without ranging bits, an 80-dB dynamic range system would require at least 14 bits, while with ranging bits only 10 bits are needed.

When the digital signal is fed to the playback processor, it is converted into the equivalent voltages and the processor holds each output level constant until the next word instructs it how to modify its output. The result is a waveform that looks like a staircase. If the audio signal is sampled at more than twice the highest audio frequency to be recorded and reproduced, and the playback processor output is filtered to remove this supersonic sampling frequency, the staircase effect (Fig. 7-23A) is smoothed out and with it the quantization error disappears (Fig. 7-23B).



(A) Before filtering. (B) After filtering.

Fig. 7-23. Sine-wave output of a digital playback processor.

Since all signals below the threshold of the lowest quantization level are ignored, a type of noise called *quantization noise* is induced in the signal when it is in this range. This noise has a very grainy quality and can be reduced by increasing the number of digits used in the word, or it can be hidden by adding hiss to the original signal to mask it.

This system has certain advantages over the Dolby, DBX, and Burwen noise-reduction devices:

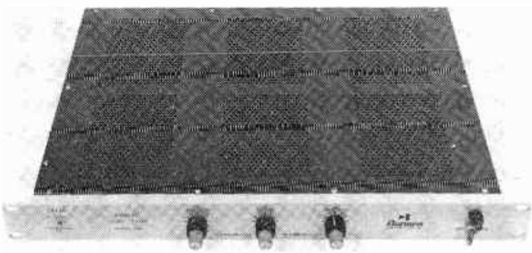
1. The dynamic range and signal-to-noise ratio are limited only by the cost of using longer binary words.

2. Since tape noise, output level, and distortion have no effect on signal quality (as long as a signal larger than the noise level can be recorded), inexpensive tape can be used, and tape machines would need no pre- or post-EQ, or bias. Erasure is not needed unless the return to zero code is used because recording to the saturation levels wipes out any previous signals.
3. Cross talk between channels and print-through are eliminated because they occur at a level significantly below that of the recorded flux and are ignored.
4. Dropouts are minimized because of saturation recording.

The disadvantages of the system are its cost at the present time (although integrated-circuit technology will reduce this in the future) and problems encountered in splicing digitally recorded tape. Splicing can be done electronically by duplicating the tape, placing it on two machines, and crossfading the two signals while recording on a third machine, as is done in splicing video tape. No generation loss occurs, but a means of getting the machines in sync at the crossfade point must be provided. The only prerequisite for the machine (aside from being able to record and play back somewhat) is that its speed constancy be at least as good as present analog audio recorders.

THE BURWEN DYNAMIC NOISE FILTER

The systems discussed up to this point all require processing in recording and playback to achieve noise reduction. Noise cannot be removed from a signal with these devices; it can only be prevented from entering. Burwen Laboratories has developed their Model 1000 Dynamic Noise Filter (Fig. 7-24) to remove noise already present in a signal by up to 20 dB at 30 Hz and 22 dB at 10 kHz. The device is a variable bandpass filter which attenuates high and low frequencies when there is no music present. When high or low frequencies appear in the signal, the bandpass rapidly changes to allow



Courtesy Burwen Laboratories

Fig. 7-24. The Burwen Model 1000 Dynamic Noise Filter.

either or both of them to pass without attenuation, relying on the masking effect to hide the noise. Rapid attack times prevent the filter from attenuating sharp transients. The filter can be used to reduce noise on existing records and tapes. Since it will remove noise present from all sources ahead of it, multitrack tapes that were not made using noise-reduction devices can be mixed through the dynamic noise filter, and discs can be mastered through it.

The high- and low-bandpass controller sensitivities are variable in order to optimize noise reduction for different types of program material. A high-frequency cutoff adjustment is available to prevent the upper high-frequency limit from increasing above a certain frequency when especially noisy program material is processed. The noise filter reduces tape noise by a total of about 10 dB (when the varying reductions at different frequencies are averaged).

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CHAPTER 8

Speakers and Monitoring

In the recording process, judgments and adjustments of the sound quality are based entirely on what is heard through the monitor system; thus it is extremely important that monitors be set up and used properly. Speakers are the weakest link in the audio chain because their response is the most difficult to make flat. In addition, the acoustics of a room can create large peaks and valleys in the frequency response at the listening location. The only place a speaker can truly be designed to have flat response is in an anechoic chamber, i.e., a room that absorbs all the speaker output and reflects none of it back. In a room such as this, there can be no constructive or destructive interference of the reflected waves as normally occurs and what is heard is the direct output of the speaker. Since people do not listen to music in anechoic chambers, the listening room must be taken into consideration when choosing a speaker.

SPEAKER/ROOM CONSIDERATIONS

Unless the rooms are of identical dimensions and furnishings, a speaker will sound differently (it will have a different frequency-response curve) in every room in which it is placed. This means the speakers must be tested in the room in which they are to be used. Direct comparisons of various speakers in the same room are valid only in demonstrating differences between the speakers; direct comparisons cannot determine which one will sound best in a specific room. This problem of sound variation in rooms makes it difficult to interchange control rooms in a recording studio. After a tape is recorded in a specific control room the producer and artists become accustomed to hearing the material sound a certain way.

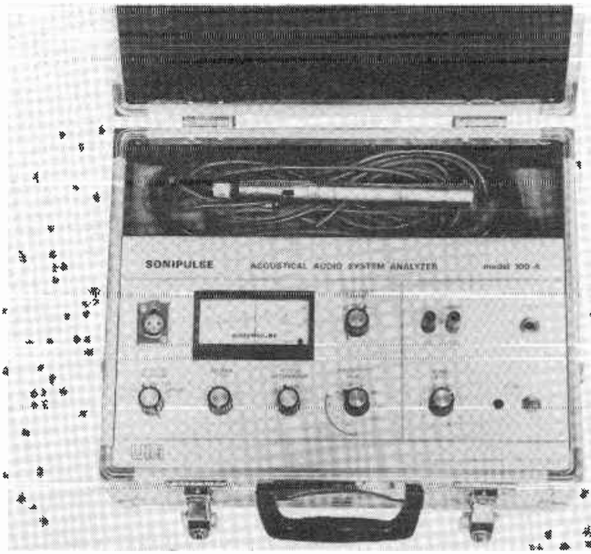
If the tape is then mixed down in another control room with the same speakers but a different speaker/room response curve, there can be a disturbing difference in the sound of the instruments.

To eliminate this, many studios *tune* their speakers to the room so that the frequency-response curve of the speaker/room is the same for each control room. The tuning is done using peaking and dipping filters usually of 1/3-octave bandwidth connected before the monitor power amp. Pink noise (which has a flat energy spectrum curve throughout the audio range) is fed into the speaker system. Then, the acoustic outputs are measured one at a time in 1/3-octave increments with filters and an spl meter which has a calibrated omnidirectional microphone. This allows any mike deviations from flat to be corrected from the results of plotting the speaker/room curve. An omnidirectional mike is used because the ear is omnidirectional and thus hears the reflected sound of the room as well as the direct sound from the speaker. Since the mike can only be in one spot at one time, the response curve obtained will be accurate only for a listener at the spot the mike was placed. The response curve of a tuned system will vary from one spot in the room to another, and the response at the engineer's and producer's seats must often be a compromise so that both hear similar responses.

Examples of equalizers used in tuning speaker systems are the Altec Acoustavoice and Acoustavoicette, the UREI Model 527-A Graphic Equalizer (1/3-octave increment bandwidths), the Soundcraftsman equalizer, and the Advent Frequency Balance Control (1-octave increments). Pink noise is used for testing rather than sine waves, because it is of a random nature and does not stimulate standing waves in a room as a sustained tone would. The presence of standing waves would introduce inaccuracies in the readings on the spl meter that would vary with the position of the microphone in the room.

Another method of obtaining a response curve is to use a full frequency spectrum pulse of known frequency content and measure the response at the output of the calibrated microphone. The pulsed nature of the signal prevents the buildup of standing waves. This method is used in the UREI Sonipulse Acoustical System Analyzer (Fig. 8-1) which has the advantages of being more compact and inexpensive than the pink noise method as well as providing more stable and thus more accurate meter readings than those made with pink noise (the random nature of pink noise makes meter readings fluctuate, and they must be averaged by eye).

Although these devices can flatten speaker response, the output of many speaker systems and amplifiers falls off at the low end and becomes distorted if too much low-frequency signal is applied. Thus, flattening the response of a speaker system may lower its undis-



Courtesy United Recording Electronics Industries

Fig. 8-1. The UREI Sonipulse Acoustical Audio System Analyzer.

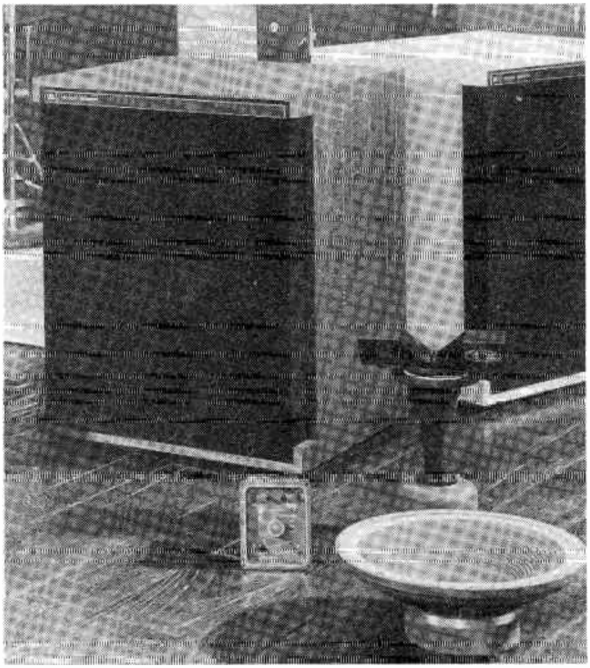
torted volume-producing capability, making it unusable if high monitor levels are desired.

CROSSOVER NETWORKS

Because individual speaker elements (called *drivers*) are cleaner and more efficient in some frequency ranges than in others (i.e., have more undistorted output for the same level input signal), different drivers are often used in conjunction with one another to obtain the desired output. Large-diameter drivers such as 15-inch units produce low-frequency information more efficiently than high-frequency information; medium-size speakers such as 4- and 5-inch units produce midrange better than highs or lows; and small speakers (½ or 1 inch) produce highs better than any other range.

These speakers are connected by *crossover networks*, which prevent any signals outside a certain frequency range from being applied to the speaker. The networks usually have one input and two outputs. Input signals above the crossover frequency are fed to one output, while signals below the crossover frequency are fed to the other output. The crossover network uses inductors and capacitors and is designed so that a signal at the crossover frequency will be sent equally to both outputs to provide a smooth transition from speaker to speaker. If a speaker system has only one crossover frequency, it it called a *two-way system* because it divides the signal into two bands. If the system has two crossover frequencies, it is called a *three-way system.* As many crossover frequencies as desired can be used, but most manufacturers use either two- or three-way systems.

The JBL 4320 monitor speaker (Fig. 8-2) used in many recording studios is a two-way system. It utilizes a 15-inch woofer for the bass and a horn-type driver for the midrange and high frequencies. The crossover frequency is 800 Hz, and a level control determines how much energy is sent to the high-frequency output of the crossover. This control enables the user to partially compensate for differences in room acoustics. Dead rooms need more high-frequency energy than live rooms to produce the same audible effect.

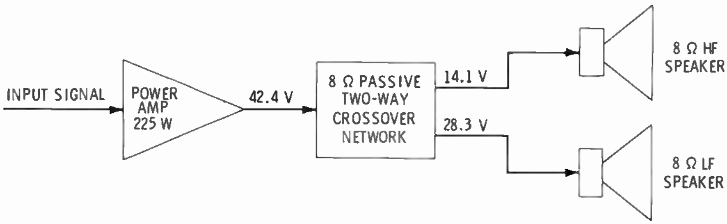


Courtesy James B. Lansing Sound, Inc.

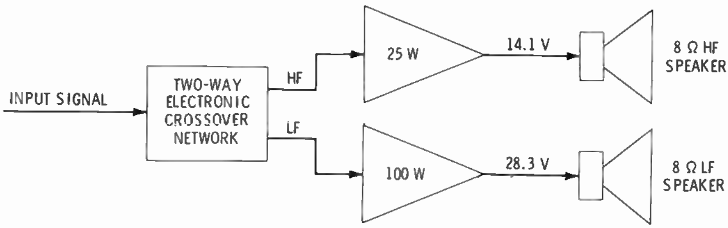
Fig. 8-2. The JBL 4320 studio monitor speaker, a two-way system.

Better crossover networks, called *electronic crossovers,* have been designed in the past few years. Instead of being connected between a single power amp and several drivers, they are used between a preamp and several power amps. Each driver is fed directly by its own power amp (a three-way system would need three power amps per channel). There are several advantages to this approach:

1. Since the signals are at low levels within the electronic cross-



(A) A passive crossover system.



(B) A biamplified system.

Fig. 8-3. Passive and biamplified crossover systems.

over, active filters without inductors can be used, thus removing a source of intermodulation distortion.

2. Power losses due to the resistance of inductors in the passive crossover network are eliminated.
3. Since each frequency range has its own power amp, the full power of the amplifier is available to it regardless of the power requirements of the other ranges.

For example, given a 100-watt amplifier feeding high- and low-frequency range drivers through a passive crossover network: If the low frequencies are using 100 watts of power, and a high-frequency signal comes along that requires an additional 25 watts of power from the amp, the amplifier cannot supply it, and both the low- and high-frequency signals become distorted.

To illustrate further (following Siniscal in [5]), picture a two-way system with 8-ohm drivers and an 8-ohm passive crossover network (Fig. 8-3A). The program material requires that the power amp be capable of supplying 100 watts of power to the low-frequency speaker and 25 watts of power to the high-frequency speaker at the same time. Producing 100 watts in 8 ohms requires 28.3 volts at the speaker terminals, while 25 watts requires 14.1 volts ($\text{voltage} = \sqrt{\text{power}/\text{resistance}}$). This means that the power amp must feed 42.4 volts to the crossover network (assuming a no-loss network), which is a power output of 225 watts. The system requirements would also be met through the use of a 100-watt amplifier to drive the low-frequency speaker and a separate 25-watt amplifier to drive the high-frequency speaker, with the input signal fed to the power amps through an electronic crossover network (Fig. 8-3B).

Systems using electronic crossovers and multiple-power amplifiers are called *bi- or triamplified systems*, depending on the number of power amps used per channel. Since the price of power amplifiers increases rapidly with their power output, it is often cheaper to use several low-power amps with an electronic crossover network than to use one high-power amp.

Passive and active crossover networks are similar in their frequency responses. The crossover point is always 3 dB down from the flat section of the response curve; slopes outside the filter passband are usually 6, 12, 18, or 24 dB per octave, with 12 being the most common (Fig. 8-4). Common crossover frequencies are 500 Hz, 800 Hz, 1200 Hz, 5000 Hz, and 7000 Hz.

SPEAKER PHASING

A pair of speakers can be in phase or out of phase with each other. If they are in phase (Fig. 8-5A), the same signal applied to both speakers will cause their cones to move in the same direction. If they are out of phase (Fig. 8-5B), one cone will move in and one will move out. Phase can be tested by applying a mono signal to the two speakers at the same level. If the signal appears to originate from between the speakers, they are in phase; but if the image is hard to locate and appears to move as the listener moves his head, they are out of phase. This effect is especially noticeable with low frequencies. An out-of-phase speaker condition is corrected by reversing the leads connecting to one of the speakers.

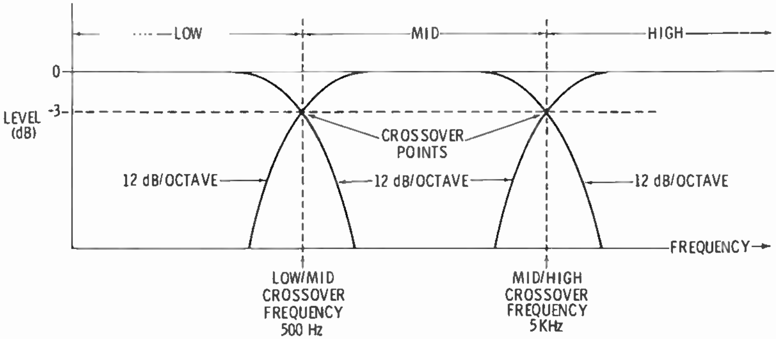
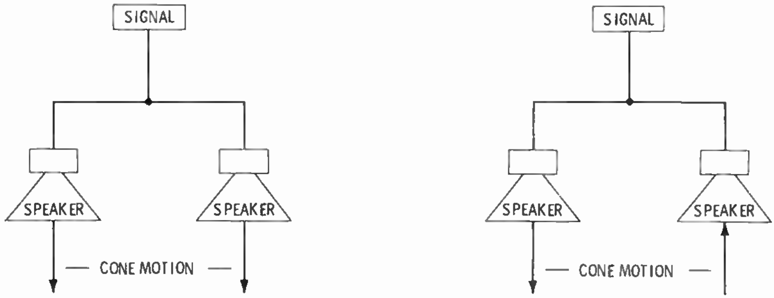


Fig. 8-4. Frequency response of a three-way crossover network with crossover frequencies of 500 Hz and 5 kHz.



(A) In phase. (B) Out of phase.

Fig. 8-5. Relative cone motion, speakers in phase and out of phase.

Speakers are phased according to a standard code: a 1½-volt battery is connected to the speaker so that the cone moves outward. The speaker lead to which the positive side of the battery is attached is then marked with a "+" or a red dot. This test should not be attempted on high-frequency tweeters because it may damage their diaphragms or voice coils.

Lamp cord is often used for speaker connections, and it has a polarizing code which make proper phasing easy. One side of the line has either a thread under the insulation or a rib or moulding on the outside of the insulation. If this coded wire is always connected to the + or red dot of the speaker and to the high side of the power amp output, speaker phase problems will be avoided.

Speaker wire should always be heavy duty; No. 18 is the proper size for less than 25-foot lengths, No. 16 for 25- to 50-foot lengths, and No. 14 for 50- to 100-foot lengths. The reasons for increasing the thickness of the conductors as cable length increases (No. 16 is thicker than No. 18) are: (1) all cable has resistance, and the resistance builds up as length increases. The more resistance there is in the cable, the more power is dissipated in the cable and is unavailable to drive the speaker; (2) the higher the cable resistance, the lower the effective damping factor of the amplifier. The amplifier damping factor is related to how well the amplifier can control the motion of the speaker cone. The lower the damping factor, the less control the amp has over the speaker and the less clarity and definition the speaker will produce. Thick conductors have lower resistance and minimize these problems.

MONITORING

In mixing, it is important that the engineer be seated exactly between the stereo speakers and that their volume be adjusted to be equal. If this is not done, signals desired in the center of the created stage may be off to one side or the other. If the engineer is closer to one speaker than to another, that speaker will seem too loud and the other speaker too soft. The engineer may be tempted to either pan the instruments toward the far speaker or boost that entire side of the mix to equalize the volumes. As a check against doing this, the engineer should always make sure that an audible volume difference between speakers is accompanied by a corresponding visual difference between VU meters monitoring the signal sent to the tape and try to keep the levels similar. While the meters should not read exactly the same at all times, the presence of a solo in one channel should read only a few dB higher than the other channel unless the other channel is being kept very low for a specific purpose. The maximum reading of the meters should be about the same. Center channel balance can be checked when setting balances by shutting off all tracks, except those desired in the center, and checking that the VU meter readings for the left and right channels are identical.

Mixing

Several other problems remain to be considered with respect to monitoring. Even if the monitor speaker in use is absolutely flat in the control room, few of the people who buy records have flat speaker/room curves and as a result they will not hear exactly the same mix that is heard in the control room. Purchasers will hear different frequency balances due to response variances between types of speakers and listening rooms.

There are several possible views on this matter:

1. Mix for those few people who have flat systems; make the rest of the purchasers accept whatever inaccuracies are introduced by their systems.
2. Mix on home-type hi-fi speakers of medium quality; compare the sound of the mix on several different brands of speakers, adjusting the mix until it sounds good on as many systems as possible.
3. Mix through inexpensive speakers such as those found in car radios and some portable record players.

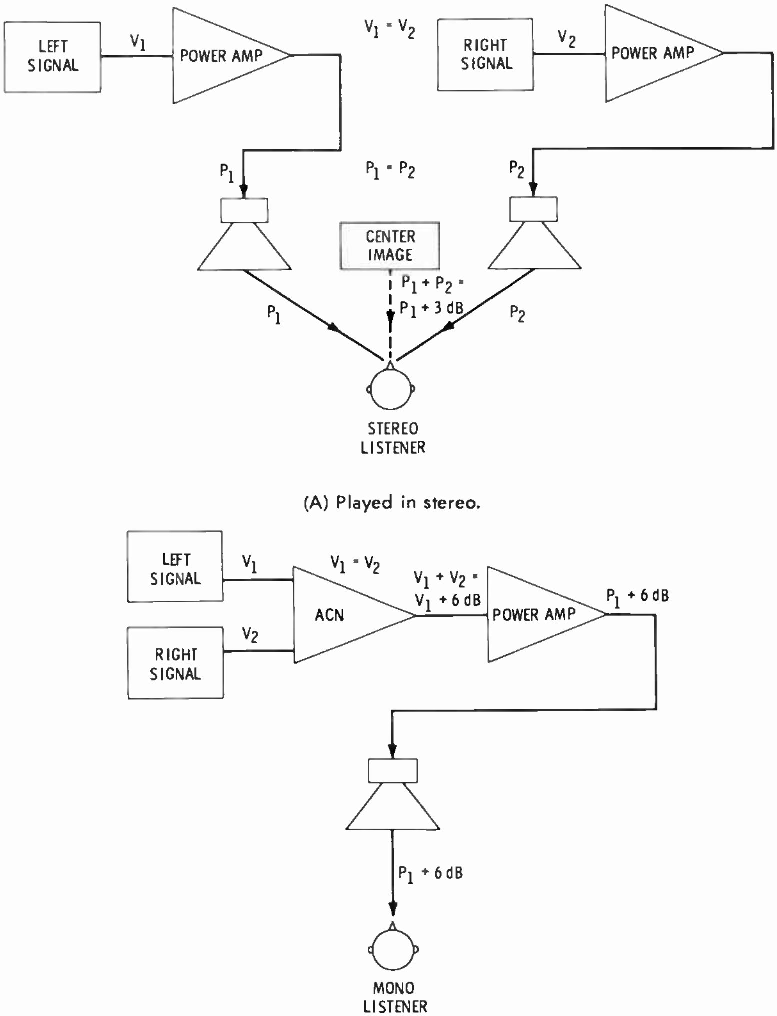
The problem with the first view is that owners of flat systems are not the majority of record buyers. Catering to this market, while neglecting the larger group of people who buy most of the records, will result in poor sales. In the second view, mixing for the consumer with medium-quality speakers will please the majority of record buyers, but not the artist or producer. The third view of mixing with inexpensive speakers applies to special markets, such as a-m radio or teen-age music, where the song will receive most of its play through low-quality equipment and its balance must be set so that the music sounds good on this equipment. Car radio speakers and cheap phonograph speakers are notable for their lack of bass and highs but abundance of peaky midrange response. Deficiencies in the frequency balance of the monitor speakers are inverted when the mix is played on flat speakers, i.e., if the mix sounds good on a speaker with a big peak in the midrange response, it will sound muddy on a flat system.

Another possible view is that the sound of the control-room speakers is irrelevant as long as the engineer knows how the speakers must sound in order for the song to sound good on other speakers. If there is a midrange peak in the control-room speakers, the engineer can purposely make the mix sound shrill and piercing because he knows that on flatter speakers the excess midrange will not be present and the mix will not sound shrill. While this approach works in practice, it is very difficult to convince producers and artists that the mix must sound bad at the studio in order to sound good at home.

A compromise solution is to tune the monitor speakers flat up to 5 or 8 kHz and then gradually roll off the response so that it is down about 10 dB at 16 kHz. This will make the engineer boost the level of the high frequencies in the mix. Since almost all consumer hi-fi equipment is deficient in high-frequency response, the boost is rolled off in the listener's home. The boosted high end has become a standard in the pop music industry over the past decades so that the high-frequency rolloff on the monitor speakers will make records sound the way people are used to hearing them. This approach, in addition to having several different types of untuned speakers for comparison, is used by many recording studios.

Monitor Volume

To make monitoring even more difficult, the Fletcher-Munson curves come into play, making the frequency balance of a mix vary depending on the monitor volume. If the balance is set while listening at loud levels and then the mix is played back softly, the bass and extreme highs disappear. If the balance is set while listening softly and the mix is played back loudly, there is too much bass and too many highs. Since it is impossible to know how loudly a listener will play the song, it is difficult to know how loudly to monitor while mixing. It appears that 85-dB spl is the best monitoring level from the standpoint of minimum apparent frequency balance. In other words, a change in playback loudness or softness is less at 85-dB spl than at any other mixing monitor level. The response that is considered flat at 120-dB spl changes when it is played back at lower levels. As playback volume decreases, the bass first increases and then begins to decrease, and output in the presence range begins to fall. So, a mix made at 120-dB spl will sound distant and lifeless at lower levels. If the mix is made at 100-dB spl, dips occur in the presence range as monitor volume is decreased but the most noticeable change is at the low end, where 50 Hz is 12 dB down at 70-dB spl. If mixed at 85-dB spl, however, playback between 90- and 60-dB spl causes response changes which are less than 5 dB at the extremes of the spectrum and practically no change within the spectrum. Thus this is a good compromise level for mixing, allowing for a cer-



(A) Played in stereo.

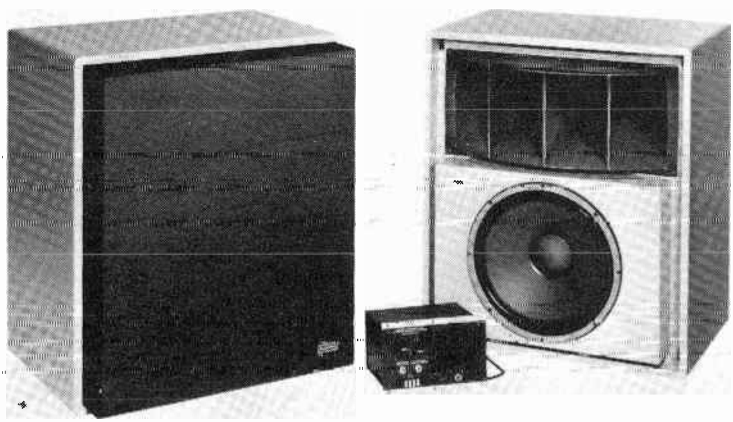
(B) Played in mono.

Fig. 8-6. Comparison of stereo and mono playback.

tain amount of variation in playback level by consumers without adversely affecting the sound. A meter can be set up in the control room to indicate when average spl is 85 dB so that all mixes can be monitored optimally. Conveniently, average home listening levels are in the 75- to 85-dB spl range.

Compatibility

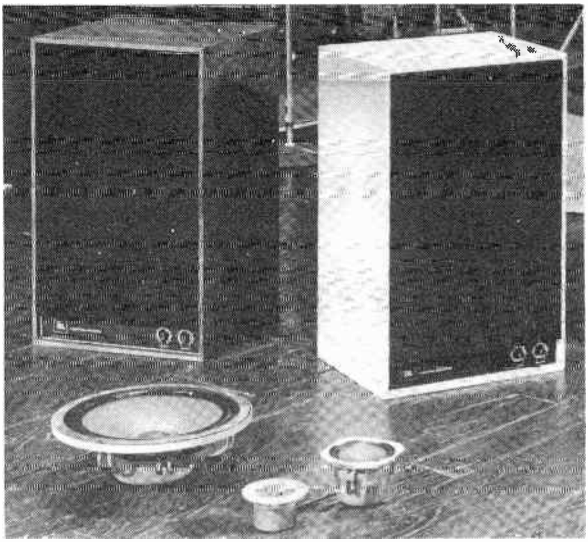
One more concern in monitoring is mono/stereo/quad compatibility. The majority of fm radio and all a-m radio listeners listen in mono, and radio introduces much of the new pop music to the consumer. Thus, if an album sounds good in stereo or quad but poor in mono, it may not sell well because its greatest initial exposure will be in mono. To prevent this, mixes should also be listened to in mono to make sure that no out-of-phase components are present which could cancel out instruments and ruin the balance.



Courtesy Altec, a division of Altec Corp.

Fig. 8-7. The Altec 9846B, a biamplified speaker system.

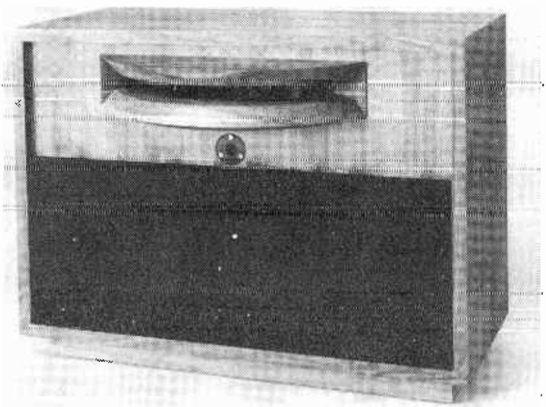
In addition, the instrumental balance will change when a quad or stereo mix is played back in mono. When the left and right signals mix in a room, a signal sent to both the left and right speakers will appear to come from between the two speakers (Fig. 8-6A) and its volume will be that produced by the sum of the power sent to each speaker. The total power of a signal in the center is twice or 3 dB ($dB = 10 \log P_1/P_2$) more than the power sent to either the left or right speaker alone, so the center signal is 3 dB louder than if one speaker were shut off. If, however, the left and right signals are mixed into mono before being sent to the speakers (Fig. 8-6B), the



Courtesy James B. Lansing Sound, Inc.

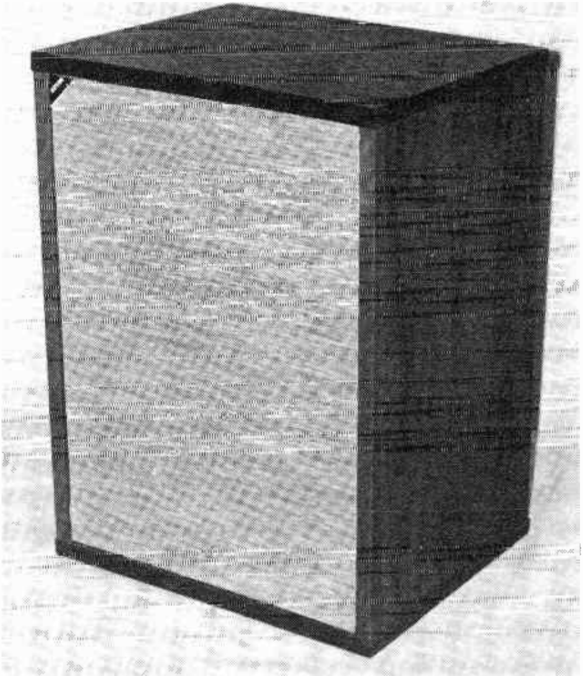
Fig. 8-8. The JBL 4310 Studio Monitor.

voltages are added rather than the powers and center information will be twice the voltage, or 6 dB ($dB = 20 \log V_1/V_2$) greater than a signal of the same level in one channel only. Thus, when played through a speaker in mono, signals which were in the center of the stereo mix will be 3 dB louder in mono than in stereo, while signals which were in one channel only do not change in level. There are devices which can be used to eliminate this center channel buildup



Courtesy Westlake Audio, Inc.

Fig. 8-9. The Westlake Audio Studio Monitor.



Courtesy Hopkins Sound Technology

Fig. 8-10. The Hopkins Model 1500.

by shifting the phase of one channel, but this is done at the cost of a loss of separation when the mix is played in stereo. The same compatibility problems arise when a quad mix is played back in stereo.

Some of the speaker systems used for studio monitoring are shown in Figs. 8-7 through 8-10.

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CHAPTER 9

Studio Session Procedures

The multitrack recording process can be divided into five different types of studio sessions: recording, overdubbing, mixdown, editing, and mastering. This chapter describes the procedures used in the first four types; mastering is discussed in Chapter 12.

RECORDING

Before a recording session starts, the tape machines to be used should be cleaned, demagnetized, and aligned for the type of tape to be used for the session. It is good procedure to record a 700- or 1000-Hz tone at 0 VU on all tracks of the tape to indicate the operating level of the tape. This tone is especially important to ensure proper decoding of the tracks if the Dolby system is used.

Once the number and types of instruments to be recorded are known, baffles can be placed to prevent leakage of the louder instruments into the mikes designated to pick up the softer ones and headphones can be distributed and plugged in at each musician's location. The setup should permit the musicians to see each other as well as possible so that they can give and receive visual cues. The arrangement of baffles and mikes depends on the type of sound the producer wants. If they are close to the instrument, a *tight* sound with good separation is achieved, while a looser, more live sound, as well as more leakage, is achieved with the mikes and baffles farther away. An especially loud instrument can be isolated by putting it in an unused vocal booth. Electronic amplifiers played at high volumes can also be recorded in a vacant vocal booth, or they can be isolated by building a box out of baffles to surround it and the mike on all four sides and the top. Another approach would be to cover both the amplifier and the mike with a blanket or other flex-

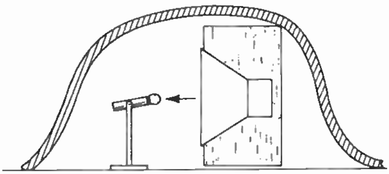

Fig. 9-1. Isolating an instrument amplifier by covering it with a sound-absorbing blanket.

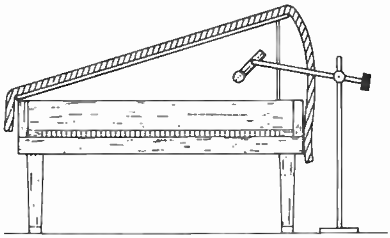

Fig. 9-2. Preventing leakage from getting into a piano mike.

ible sound-absorbing material (Fig. 9-1), making sure it does not interrupt the path between the amplifier and the mike. Separation can also be improved by placing the softer instruments in a vocal booth. For a piano, leakage is reduced by placing the mike inside it, putting the lid on a short stick, and covering with blankets (Fig. 9-2).

When using a directional mike, separation is often improved by putting a sound-absorbing baffle behind the performer. Since the back of the mike is relatively insensitive, the leakage has to reach the live side of the mike from either instruments or reflections from hard surfaces behind the performer (Fig. 9-3). The directional mike should be angled so that any reflected sound from the instrument being miked, due to music stands or other nearby hard surfaces, reaches the dead side of the mike (Fig. 9-4) to prevent peaks and dips in frequency response due to phase cancellation between the direct and reflected sound [1].

The microphones for each instrument are selected by either experience or experimentation and are connected to the desired mike inputs. The input used for each mike should be noted on a piece of paper so that the console input module corresponding to each instrument can be easily found. Some engineers find it convenient to use the same mike input and tape track for the same instrument at every session. Thus, one engineer might consistently plug the bass guitar into input 1 and record it on track 1, so that he knows which fader

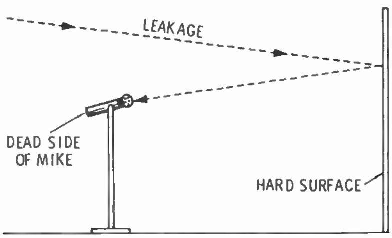

Fig. 9-3. A hard surface on the live side of a mike can put the undesired off-axis signals on-axis, by reflecting them.

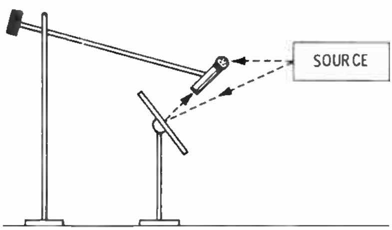

Fig. 9-4. Correct mike placement using a unidirectional mike.

controls the bass in both a record and a mix session without having to think about it.

Electronically amplified instruments with low-level, unbalanced high-impedance outputs, such as guitars, can be recorded without using their amplifiers, through the use of a matching transformer that converts their outputs to a low-impedance, balance signal which can feed a mike preamp. This is called picking up an instrument *direct*. Instruments are often picked up direct to avoid noise and distortion resulting from amplifiers with bad tubes or "blown out" speakers. Through the use of a *Y-cord* or parallel jacks, both the transformer and the instrument amplifier can be fed from the instrument simultaneously, enabling the engineer to record the direct sound alone, the amplified sound picked up by a mike, or a combination of the two. The direct sound is very clean, but more subject to string noise from guitars than amplified sound. On most guitars, lowest hum pickup and best tone for the direct connection occurs with the instrument volume control fully on. Since guitar tone controls consist of a variable treble rolloff, maximum control over the sound is achieved by leaving the tone controls on a treble setting and using a combination of console EQ and the different pickups on the guitar to vary the tone. If the treble is rolled off on the guitar, boosting the highs with EQ will increase noise that is picked up due to the high impedance of the guitar. The direct connection can also be used with the speaker output jacks of instrument amplifiers if a means of attenuating the signal is used so that neither the transformer nor the mike preamp is overloaded.

If drums are to be recorded, the drummer should tune them while the mikes and baffles for the other instruments are being set up. Each drum head should be adjusted for the desired pitch and for constant tension around the rim by hitting the head at various points around its edge and adjusting the lugs for the same pitch at each point marked by Xs in Fig. 9-5. After the drums are tuned, the engineer listens to each drum individually to make sure that there are no buzzes, rattles, or resonant after-rings when the heads are hit. Drums which sound great in live performance do not always sound that way when close miked. In a live performance, the rattles and rings are covered up by the other instruments and lost before the sound reaches the listener. Close miking picks up the noises just as well as it picks up the desired sound.

If tuning the drums does not bring the extraneous noises under control, masking tape can be used around the edge of each head on the drum, as well as across the head, to damp them out. Pieces of cloth, paper towel, or a wallet can also be taped to the head in various locations determined by experimentation to eliminate rings and buzzes The placement and pressure of this damping material varies

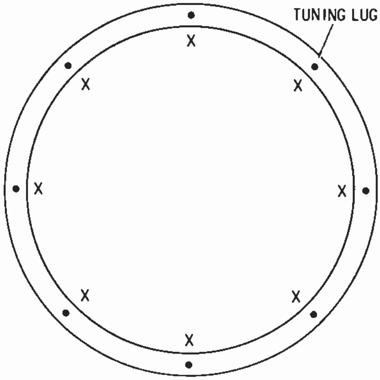

Fig. 9-5. Tuning a drum head. The pitch should be the same at each point around the rim (marked by Xs).

the amount of ring better than the dampers built into the drum, which apply pressure to only one spot on the head and therefore unbalance the head tension. The built-in dampers vibrate when the head is hit and are one of the chief sources of rattles.

For studio recording, it is best to remove the entire damping mechanism from the drum. The bass drum is damped by removing the front head and placing a blanket inside it, pressing against the head by the beater. By adjusting the pressure of the blanket against the head, the drum tone can be varied from a resonant boom to a dull thud. Bass drums are usually recorded with their front heads removed, while other drums are recorded with their bottom head either on or off. Tuning the drums is more difficult if two heads are used because the tensions of the heads interact in producing the pitch, but a more resonant tone can be obtained than with only one head. After the drums are tuned, their mikes can be put in the desired positions, making sure that they do not get in the drummer's way. If the mikes are in the way, they may be hit by a stick or moved out of position during a performance.

The engineer confers with the producer to find out how many instruments are to be used on the song, including overdubs, to determine how many tracks must be left open. This will influence the tracks to which the mikes are assigned, especially for the drums. If many instruments are to be recorded, the drums may be limited to two tracks, for example, one for the bass drum and one for the rest of the set. If there are plenty of spare tracks, five tracks may be used for drums: bass drum, snare, hi-hat, tom-toms, and cymbals, and floor toms and cymbals.

When all of the mikes have been set up, the engineer labels each input fader with the name of the corresponding instrument, either on a plastic write-in strip included above or below the fader on the console or on a piece of masking tape placed across the top or bottom of the faders. The mikes are assigned to the desired tracks, and the assignments are noted on a *track log* which is then attached to the tape box (Fig. 9-6). A rough headphones mix is set up so that the musicians can hear themselves, and the engineer asks them to play into their mikes one at a time. Starting with the EQ flat, the engineer listens for mike preamp overload and adjusts mike preamp gain, using the pad on the preamp or on the mike if necessary, to eliminate any distortion. The EQ is adjusted to obtain the sound the producer wants on each instrument, and limiting is used if needed.

XYZ STUDIOS	CLIENT:			P.O. No.:	SPEED: 7 1/2 15 30		ENGINEER:	REEL No.:
ARTIST:	PRODUCER:			DATE:	STUDIO: A B		NOISE REDUCTION: NONE DBX DOLBY BURWEN	
SELECTION 1	1. ELECTRIC BASS	2. BASS DRUM	3. DRUMS LEFT	4. DRUMS RIGHT	5. ELECTRIC GUITAR	6. ACOUSTIC GUITAR	7. ORGAN	8. PIANO
	9. LEAD VOCAL	10. VOCAL CHORUS	11. STRINGS 1	12. STRINGS 2	13. GUITAR SOLO	14. CONGA	15. MARACAS	16. (SPARE)
SELECTION 2								

Fig. 9-6. A track log illustrating the track assignments for a typical multitrack recording.

If the desired sound cannot be achieved with a minimal amount of EQ, different mikes are tried until the sound is acceptable. The engineer and producer listen for any extraneous sounds such as buzzes or hum from guitar amplifiers and squeaks from drum pedals and try to eliminate them. If several mikes are to be mixed onto one track, the balance between them can be set at this point.

After this procedure has been followed for each mike and each instrument, the musicians should *run down* or practice the song so that the engineer and producer can listen to the instruments to hear how they sound together before they are recorded. All the drums are listened to, then the bass guitar with the drums, then the entire rhythm section, and then all of the instruments together. Changes in EQ can be made to compensate for one instrument covering up another or to make them blend better. While the song is being run down, the engineer adjusts the recording levels and the monitor mix. The whole song should be performed so that the engineer knows where the loudest sections are to be sure the recorded levels will not overload the tape and, if compression or limiting is used, to be sure that the instruments do not cause more than the desired amount of gain reduction. Even though the engineer may ask them to play their loudest when the musicians are playing one at a time, each will almost invariably play louder when performing with others, requiring changes in mike preamp gain, record level, and compression/limiting threshold. Separation between the instruments can be checked by soloing each mike and listening for leakage. The relative positions of mikes, instruments, and baffles can be changed, if necessary, to reduce leakage.

The engineer checks the headphone mix by either putting on a pair of headphones connected to the cue system or routing the mix to the monitor speakers to make sure that all of the instruments can be clearly heard. If the musicians do not hear the sound they desire, the mix can be varied to intensify the sound of particular instruments. If several cue systems are available, separate headphone mixes can be made for musicians that want different balances. During loud sessions, the musicians often require high sound-pressure levels in their headphones in order to hear the headphone mix above the room sound which leaks through them. Since high sound-pressure levels can cause the pitch of instruments to sound flat as described in Chapter 2, musicians often have trouble tuning or even singing with them on. To avoid these problems, tuning should not be done through headphones. The musicians should be careful to only play as loud as necessary to feel comfortable so that the headphone levels do not have to be too high. The same situation exists in the control room with respect to high monitor-speaker levels; some instruments may sound out of tune even when they are not.

XYZ STUDIOS			ARTIST:	DATE:	REEL No.:
TAKE No.	COUNTER READING	TIME	TITLE	COMMENTS	
1	0:00			FALSE START	
2	0:20	3:30		GOOD SOLO, BUT ENDING WEAK	
3	3:50	3:29		GOOD	

Fig. 9-7. A take sheet.

Each performance is *slated* with both the name of the song and a *take number* for easy identification. A *take sheet* is kept to note the position of the take on the tape (Fig. 9-7). Comments are written on the take sheet to describe the producer's opinion of the performance, as well as whether is it a complete take, an incomplete take, or a false start.

During the recording, the engineer watches the level indicators and, if necessary, controls the faders to prevent overloading the tape. He also acts as another set of ears to listen to the performance. If the producer does not notice a mistake in the performance, the engineer should point it out. The engineer should try to be helpful, but he should remember that the producer's judgment of the quality of a performance must be accepted.

When a take is to be played back, the tape is rewound and the monitor system switched from the program to the tape playback mode. The musicians can listen to the performance either in the control room, over their headphones, or through the studio speakers. If the producer and artist decide that they like the performance, the engineer *leaders* it by splicing a piece of leader tape at the head and tail of the take so that it can be easily found for overdubbing and mixdown.

OVERDUBBING

Overdubbing or *sweetening* is used to add instruments to a performance subsequent to the recording of the basic tracks. In an overdubbing session, the same procedure is followed for mike selection, EQ, and levels as during the recording session. If only one instrument is being overdubbed at a time, the problem of leakage directly from other instruments does not exist, but leakage can occur if the musician's headphones are too loud or not seated properly on his head. The recorder is put into the master sync mode, tracks to be played back are set to the tape playback mode on the console monitor system, and tracks to be recorded are set to the program mode. The control room monitor mix should make the instruments being recorded somewhat prominent so that any mistakes can be heard plainly and the headphone mix can be adjusted to the taste of the musicians performing the overdubs.

The tape is run over and over, and either the same or different tracks can be used for each successive attempt. The advantage of using several tracks is that a good take can be saved and the musician can try to improve the performance, rather than having to erase the previous performance in order to improve it. When several tracks of an overdub have been saved, there may be parts of each track which are acceptable and can be combined to create a complete performance. This is done by playing the tracks back in the sync mode, mixing them together in the console, and recording them on another track of the tape (Fig. 9-8). The overdubbed tracks are turned on and off as necessary to transfer only the best parts of each performance to the composite track. Signals cannot be transferred to an adjacent track, however, because the crosstalk between the

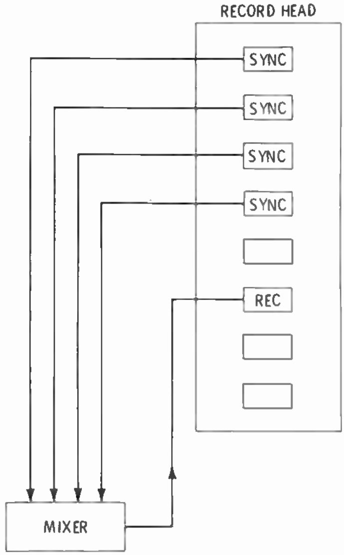

Fig. 9-8. Mixing setup.

adjacent recording and reproducing heads would cause high-frequency oscillations (Fig. 9-9). This procedure, called *bouncing tracks* or *ping-ponging*, can also be used to mix entire performances onto one or more tracks either to make the final mixdown easier or to open up their original tape tracks for additional overdubs.

For a particularly difficult overdub, it may be easier for the musician to work on one part of the song over and over until it is performed properly before going on to the next section. This is done using a single tape track and *punching in* to the record mode at the start of the section to be worked on, *punching out* of record at the end of the section, rewinding the tape, and punching in again until the section is performed properly. The next section is recorded in the same manner.

Overdubbing is often used so that a musician can play or sing along with himself to make his performance sound fuller or so that

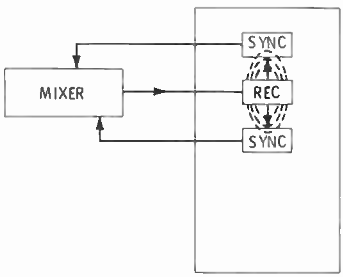

Fig. 9-9. Feedback path when attempting to bounce a signal to an adjacent track.

an effect can be created. The musician listens to his original performance and tries to match its phrasing as he overdubs. The two tracks are then played back together. This technique is called *doubling*. Additional tracks can be recorded in the same manner to make a vocal chorus out of one singer.

Some overdubbed instruments only play during certain sections of a song. On the sections where he is not performing, the musician should be careful not to make any noises which would be recorded on his track. If noises are recorded, they either have to be erased before mixdown or the track has to be shut off during the mixdown at the points the noise occurs.

MIXDOWN

After all the tracks required for a song are recorded, the multitrack tape must be mixed down to either mono, stereo, or quad for distribution to the consumer. The multitrack and the mixdown tape machines must be demagnetized, cleaned, and aligned before starting. After alignment is complete, the engineer should record 0-VU level tones at the head of the mix reel at 700 Hz, 10 kHz, and 50 Hz so that the disc mastering engineer can align his tape machine to play back these tones at 0 VU, resulting in proper playback EQ for the tape. If the Dolby system is used, the Dolby tone is used instead of 700 Hz. These tones can also be used to align the mixdown recorder at subsequent mixdown sessions for the same LP, so that one set of playback alignment tones is correct for all of the mixes.

Each input module on the console is switched to the line position, and the faders are labeled with the names of the instruments they control (either on their write-in strips or on masking tape). The engineer sets up a rough mix of the song by adjusting the levels and left to right to front to rear positioning the way he thinks the producer wants them. The producer listens to this mix and asks the engineer to make specific changes such as "make the guitar louder" or "add some bass to the voice." The instruments are often soloed one by one or in groups, and EQ changes are made. The engineer tries to translate the producers rough descriptions of sound character such as "fat," "thin," "round," or "wooden" into settings on the equipment. Compression and limiting are used on the individual instruments as needed either to make them sound fuller and more consistent in level or to prevent them from overloading the mixdown tape when they are raised to the desired level in the mix. As the mix begins to take shape, echo is added to give the close miked instruments a more live, spacious feeling and to help the instruments blend. Other effects such as tape delay, phasing, and so forth can be suggested by either the engineer or the producer.

If the fader settings have to be changed during the mix, the engineer marks the different levels on the fader scale with a grease pencil and learns when to move each fader from one mark to another. If there are more changes needed than he can handle alone, the producer or artist might help him by controlling certain faders. It is best if the producer does not have to handle any controls because he can then concentrate fully on the music rather than the mechanics of the mix. The engineer listens to the mix from a technical standpoint to detect any sounds or noises that should not be present in the mix. If noises are recorded on tracks not in use during a section of a song, these tracks can be shut off until needed. After running down the song enough to determine and learn all the changes, the mix can be recorded and the ending faded out with the master or submaster faders. Songs with exceptionally difficult control changes can be mixed in sections which can be spliced together afterwards.

The different takes of a mix should be slated as they are recorded, and a take sheet should be kept noting the differences between takes. When an acceptable take is recorded, it is leadered at the tail end (and preferably at the head as well) so that it can be easily found. The leader can be inserted roughly or tightly depending on whether the producer is in a hurry to go on to the next mix. If roughly, the leader is placed far enough ahead and behind the song to prevent cutting off the beginning or end of the mix. If tightly, the engineer cuts the tape just before the song starts to eliminate any slates or noise occurring before the start of the performance.

The beginning of the mix is listened to at high volume and the tape is moved back and forth over the heads by hand with the tape machine in the *stop-edit* mode to a point just before the first sound of the performance begins. The tape over the playback head gap at this point is marked with a grease pencil. If there is no noise directly in front of this spot, it is good practice to cut the tape a half inch before the grease pencil mark as a safety precaution against editing out part of the first sound. If there is noise ahead of the first sound, the tape should be cut at the mark and the leader inserted.

The tail of the song must be monitored at even higher volume because it is usually a fade-out or the overhang of the last note and is therefore much softer than the beginning of the song. The tape is marked and cut just after the last sound dies out to eliminate any low-level pops that may have been recorded when the bias fed to the record head was shut off and to get rid of the tape hiss from the blank tape.

Mixes should be made at consistent listening levels because the variation in the frequency response of the ear at different sound-pressure levels results in a mix sounding quite different at different monitoring levels. The level used should ideally be the same as that at which the listener will hear the record. Since most people listen to music at moderate volume, moderate monitoring levels (80- to 90-dB spl) should be used.

The mix should be tested for mono/stereo/quad compatibility to see what changes in instrumental balance will occur when the material is played in these different formats. If the changes are drastic, the original mix may have to be modified to make the two-channel version of a quad mix acceptable, as well as to make the mono version of a stereo or quad mix acceptable. The mix should be played over car radio speakers (4 or 5 inches round or 6 by 9 inches oval) to see how it will sound on a system with limited frequency response. If the mix is for a single rather than an LP, the entire mixdown session may be done at low volume over the car radio speakers.

EDITING

After all the mixes for an LP are completed, a master-mix reel is made up for each side of the record. The producer and artist decide on the sequence of the songs on each side on the basis of their tempos, their keys, which ones seem to flow best into one another, and which songs will attract the listener's attention best. The engineer edits the mixes out of their original reels and splices them together in sequence on the master reels, tightening up the leaders at the same time if this was not already done. The level set tones are included as the first band on the master reel for side one of the record.

The length of time between the end of one song and the beginning of the next can either be a constant 3 to 6 seconds, or an amount varying with the relevance of one song to another. Decreasing the time between songs can make a song seem to be a continuation of the previous one if they are similar in mood, or can make a sharp contrast with the preceding song if the moods are dissimilar. Longer times between songs let the listener get out of the mood of the previous song and prepare him to hear something that may be different, without accenting the contrast between them. If crossfades are desired rather than silence between songs, they are done at this point as described in Chapter 5.

The length of leader tape used determines the time between the songs. Paper, rather than plastic, leader is used because the plastic can cause static electricity pops. Blank tape could be used rather than leader, but it does not provide a visual division of the songs on the reel, making it more difficult to find a particular song. It also produces tape hiss between the songs, rather than silence.

When the sequencing is complete, several 7½ ips, 1/4-track dubs of the master reels are made for the producer, the artist, and the record company executives, so that they can hear the mixes and approve them before the record is pressed. A *safety copy* of the master mix tapes is made at 15 ips before the master mixes leave the studio. This copy serves as back-up protection in case the original mixes are lost or damaged. Although it is one generation removed from the master, and has 3 dB more tape noise, it can be used to cut the lacquer masters if necessary.

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2. Woram, John M. "The Sync Track." *db, The Sound Engineering Magazine.* Vol. 6, No. 5. May 1972, pp. 10-12.

CHAPTER 10

Interlocking Tape Machines

The number of separate channels of information that can be recorded on a tape of a particular width is determined by the track configuration of the tape machine. Once all of the tracks have been recorded, new material can only be added by erasing some of the existing material. The problem of working with a finite number of tape tracks is solved by recording additional tracks on additional reels of tape.

SYNCHRONIZATION

In multitrack audio work, different channels of sound are kept separate, but synchronized, by recording them directly above or below each other on the same piece of tape. Through the use of sel sync, additional sounds can be synchronized with the original tracks at a later date by recording them directly above or below the originals. Since the strip of magnetic oxide corresponding to each track is attached to the tape backing material, the time relationship of the recorded tracks cannot vary. When tracks are recorded on separate reels of tape, the tracks on any one tape are physically held in sync; but another means must be used to maintain sync between the different reels (Fig. 10-1). This is done by *interlocking* their tape speeds. Although as many reels as desired can be interlocked, only one additional reel is usually used.

MULTIPLE-REEL SYNCHRONIZATION

Synchronization does not require the tape speeds to be constant, but it does require the speeds of both tapes to be equal at all times. If one tape slows down or speeds up relative to the other, synchronization is lost; if both tapes slow down or speed up by the same

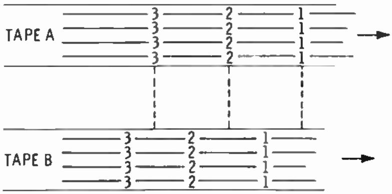


Fig. 10-1. Events 1, 2, and 3 on each track of each tape are held in sync by the tape backing. The events on the two tapes are out of sync with each other.

amount, sync is maintained. Neither hysteresis synchronous nor servo motors alone can maintain tape speed accurately enough for this application. Both of these speed control methods maintain capstan rotational speed constant and rely on the pressure of the pinch roller to transmit this into constant tape speed. Since varying amounts of slippage between the tape and the capstan can occur with this method of moving tape, fluctuations can be introduced into the tape speed. In addition, the tape speed is determined by the diameter of the capstan and the diameter decreases as the capstan wears. As a result, one capstan may move tape faster than another even though both capstan motors turn at the same speed.

In order to keep the tapes in sync, a means of measuring and correcting the tape speed must be used. This is done by recording a *sync tone* on one track of each tape. This track is called the *sync track* or *control track*. Any variation in the speed of the tape during record or playback of this tone will cause a change in sync tone frequency. Through the use of a *resolver*, the control track signal frequency of one machine can be compared with that of an *external sync reference* such as the sync tone generator, and its capstan motor speed can be varied to make the frequencies equal. The tape speed is controlled in such a way that each cycle of the control track signal is brought precisely into phase with each cycle of the external signal. This is called *phase-locked* operation (Fig. 10-2). The sync tone is

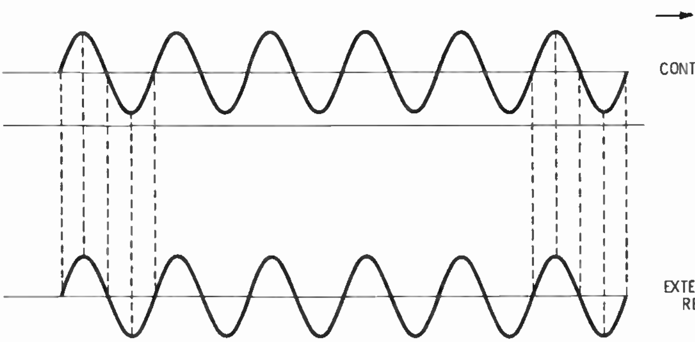


Fig. 10-2. Phase-locked operation. The resolver brings the control track signal precisely into phase with the external sync reference, and then maintains this relationship.

often thought of as providing *magnetic sprocket holes* on the tape, which are driven cycle by cycle by the external sync reference in the same manner that camera sprockets drive the holes in motion picture film.

If the external reference frequency is the same as that originally recorded on the control track, the information on the tape will play back at exactly the same tempo at which it was performed. For use with an hysteresis synchronous motor, the resolver produces a 60-Hz output signal which feeds a power amplifier which in turn drives the capstan motor. If the control track frequency is lower than the external sync frequency, the resolver output frequency increases to speed up the capstan. If the control frequency is higher than the external reference, the output frequency decreases to slow down the capstan.

In Fig. 10-3, the output frequency of resolver f_3 is varied to make control track output signal f_2 equal to external reference frequency f_1. If $f_1 = f_2$ when $f_3 = 60$ Hz, f_3 is held at this value. If f_2 is greater than f_1 when $f_3 = 60$ Hz, f_3 is reduced until $f_2 = f_1$. Then f_3 is held at this lower value. If f_2 is less than f_1 when $f_3 = 60$ Hz, f_3 is increased until $f_2 = f_1$. Then f_3 is held at this higher value.

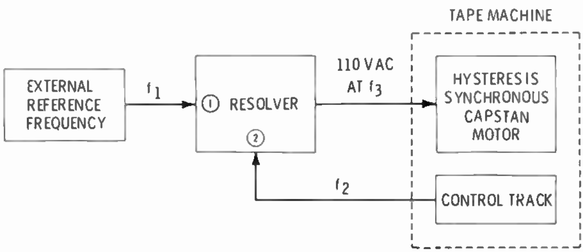


Fig. 10-3. Controlling tape speed with a resolver and a 60-Hz synchronous capstan motor.

Servo motors have resolvers built into them and constantly compare the output of their tach head with a reference frequency. If the sync track is fed to the servo motor instead of the tach head signal, the motor speed varies to keep tape speed constant, rather than capstan speed constant. Sixty hertz is the most commonly used sync tone frequency since it is available wherever the power line is accessible, but several other frequencies are also in use. Color-video tape recording uses a 29.97-Hz control track, while black and white video tape uses a 30-Hz track. Some film systems use a 14-kHz tone amplitude modulated by 60 Hz. Areas that have 50-Hz power lines often use 50-Hz tones for sync. Although machines using different sync frequencies can be synchronized by using a separate resolver and external reference frequency for each machine, it is more common in audio work for all machines to use the same sync frequency.

Several reels of tape can be kept in sync with each other by varying the speed of one on the basis of the speed of the other in a *master-slave* relationship. The 60-Hz power line is used to generate a sync tone by stepping down its voltage with a transformer and recording it one one track of the original reel of tape, which is used as the master. This control track can then be dubbed onto another tape, establishing a cycle for cycle time relationship between them. The additional reel is considered the slave and is made to follow the speed of the master by driving its capstan motor with a resolver using the master control track as the external frequency reference.

In Fig. 10-4, the master control track is connected to the resolver external sync reference input, and the slave control track is con-

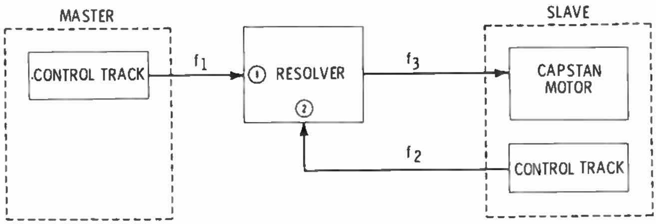


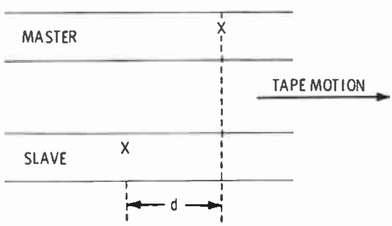
Fig. 10-4. A master-slave relationship.

nected to the sync tone input. Tone f_3 varies to keep the slave tape in sync with the master tape. The slave control track is either dubbed from the master control track or recorded simultaneously with the master control track from the same sync tone generator.

If the speed of the master varies, the resolver senses the change and maintains sync by varying the speed of the slave in the same manner. As many slave reels as desired can be synchronized, providing each reel contains a control track dubbed from the master, and is driven by a separate resolver.

The resolver can only synchronize tape speed, not program material. In Fig. 10-5, if phase lock occurs when the programs are out of sync by distance d, the resolver will maintain this out of sync

Fig. 10-5. Sync tones are regular and continuous waves. Thus the resolver can lock any two cycles together.



condition. For proper synchronization, some means must be used to ensure that sync between the programs occurs at the same time phase lock occurs.

If the programs are already in sync, the resolver will maintain it, but a separate method must be used to first get the tapes in sync. This is accomplished by using a common start button for the tape machines and placing a piece of tape with a control track on it ahead of the program material, so that the resolver can match the tape speeds before the program begins. By trial and error, starting spots can be found and marked on each tape which will fix the program in sync. Since the resolver can only maintain sync when the capstan is engaged, sync is lost each time the tape is stopped or rewound; it must then be established by running the tapes from the starting marks.

In an overdub session where punch ins are to be done, running the tapes from the beginning for each punch in wastes a lot of time. The problem can be avoided by recording a rough mix of the master tape onto the track of the slave tape. This track can be used to feed the headphones for the overdubs. The master tape can be stopped and the slave machine used without the resolver, since the rough mix is physically locked in sync with the slave control track by the tape itself. The musicians will synchronize their overdubs with the rough mix. When the overdubs are finished, the two machines can be synchronized for mixdown by using the resolver.

Since additional tracks of tape require extra console inputs for mixdown as well as more noise-reduction units, it is possible for the number of interlocked tracks to exceed the mixdown capacity of the studio. This can be prevented by leaving blank tracks on the master tape and making submixes of the slave tracks to be dubbed onto the master (Fig. 10-6) [1].

A second system for interlocking audio tape makes use of the SMPTE (Society of Motion Picture and Television Engineers) serial time code rather than a discrete-frequency tone. This code, designed for use in electronically editing video tape, uses an 80-bit digital word which is recorded at the beginning of each video frame. Each word assigns a different address to the corresponding point on the tape, expressed as the number of elapsed hours, minutes, seconds, and frames from the beginning of the program. Since there are 30 black and white video frames per second, new words are generated 30 times a second, producing a bit rate of 2400 bits per second. The code can be adapted to the color video standard of 29.97 frames per second by having the synchronizer count every thousandth frame twice (called the "dropped-frame technique").

The SMPTE code uses the digital phase modulation format called *biphase mark*. Each digital word is divided into time intervals called

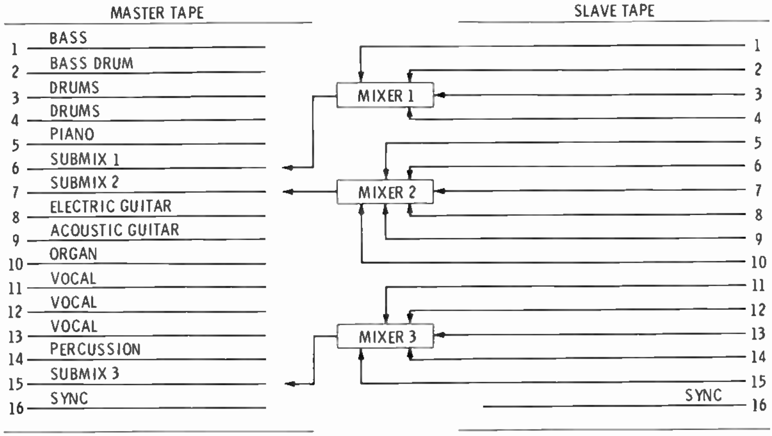


Fig. 10-6. Reducing the mixdown capacity requirements for interlocked tapes.

bit cells, and one bit is stored in each cell. A "1" is represented by a magnetic flux polarity reversal at the middle of the cell, while a "0" is represented by the lack of a reversal at this point. All cells have flux reversals at their beginning (or end).

In Fig. 10-7, the word shown corresponds to a readout of 16:47:31:23 on an SMPTE code reader. Bit groups A, B, C, D, E, F, G, and H are for optional binary control words which can be stored within the code to control auxiliary functions. The bits are left as 0's when no optional words are used.

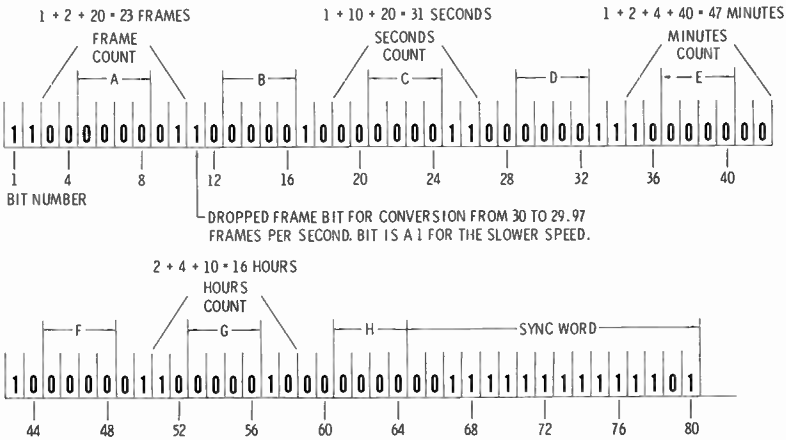
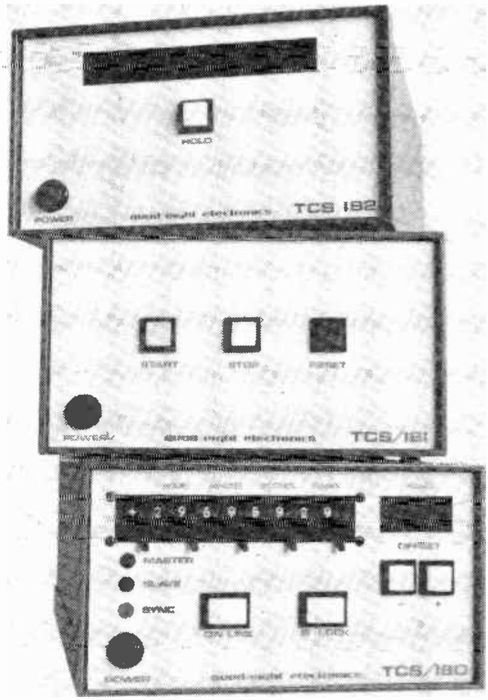


Fig. 10-7. The location of the 80 bits in the SMPTE digital word.

The last 16 bits form a *sync word* which is used in establishing the start of the next frame, as well as in adjusting the bit counting rate so that the address can be read when the tape is run at higher or lower than normal speed, or in reverse. An SMPTE *time code generator* can record the code on the control track of an audio recorder.

A frame-for-frame lock can be achieved between two tapes by using a *time code synchronizer* (Fig. 10-8). This unit has two oper-



Courtesy Quad/Eight Electronics

Fig. 10-8. Top to bottom: The Quad/Eight Electronics SMPTE time code reader, generator, and synchronizer.

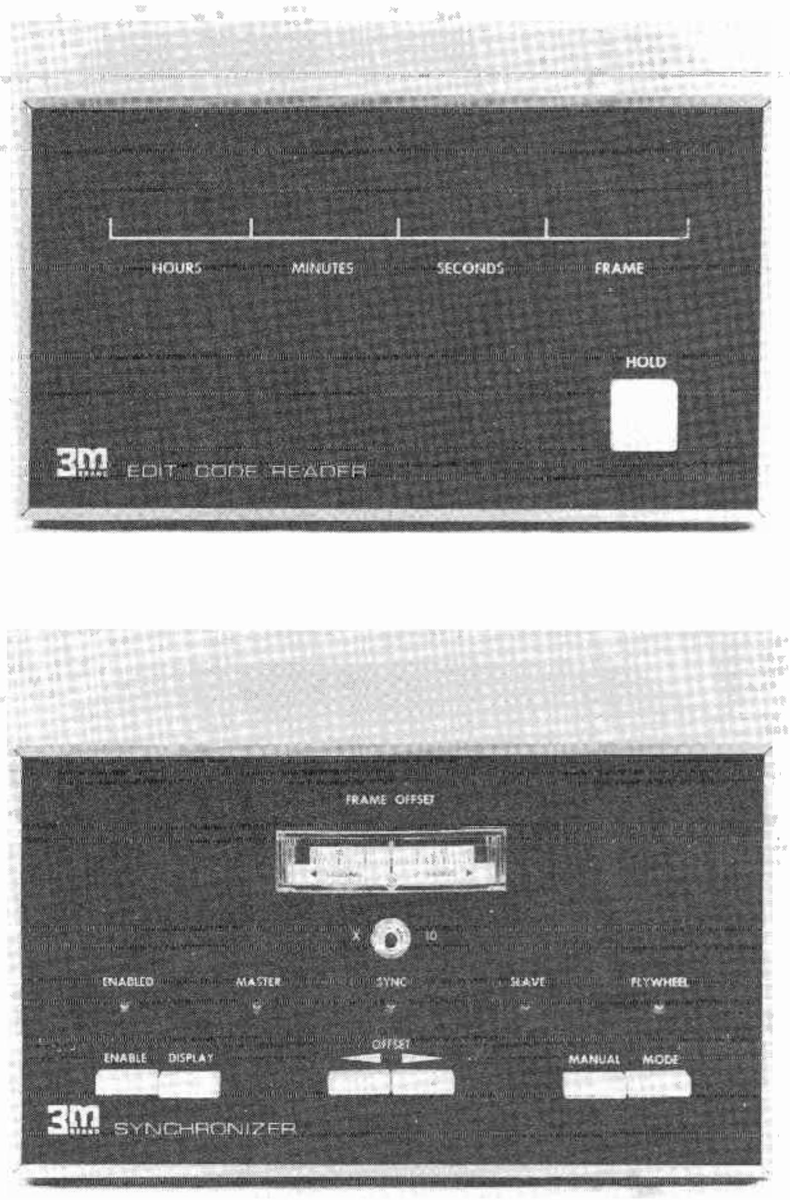
ating modes, the address mode and the phase lock or *flywheel* mode. In the address mode, the synchronizer compares the addresses contained in the code words and generates a dc voltage to vary the speed of the slave machine until both control tracks feed the same address to the synchronizer at the same time. This eliminates the problem of getting the tapes in sync. If the slave is behind the master, the synchronizer senses a lower address on the slave tape and speeds it up until it matches the master address. If the master is behind the slave, the slave is slowed down until the master catches up. For sync to be achieved, the codes on the control tracks of the master and slave tapes must be identical, and the relative positions of the tapes must be within the synchronizer *capture range*.

The 3M synchronizer (Fig. 10-9) has a capture range of ±30 seconds, so the slave address must be within 30 seconds of the master tape address when the master is started. The capture range is limited because the address sensing circuitry uses only the frames and seconds data for address locking. If the tapes are started more than 30 seconds apart, the synchronizer will lock the wrong minutes together [7]. A *time code reader*, which can be switched to read either the master or slave code, produces a numerical readout of the address in hours, minutes, seconds, and frames to enable the tape operator to *park* the master and slave tapes within the capture range of the synchronizer. The operator puts the master tape in the play mode, and the synchronizer automatically starts the slave when it senses the presence of the master code.

The tapes are brought into absolute sync and held there regardless of any tape stretching that may have occurred after the control tracks were recorded. When the master is stopped, the synchronizer stops the slave machine. Since the slave is stopped within the capture range, sync is quickly established when the master is started again. If the tapes are to be played again, they are rewound separately to the desired point and parked within the capture range. When the master is started, the slave is brought into sync again.

In the phase lock mode, the synchronizer operates like the resolver described earlier, using a 30-Hz signal derived from the frame rate as the sync tone. It ignores any differences in the addresses of the code words and adjusts the speed of the slave machine so that it matches that of the master. When the frame rate of the two codes is the same, the synchronizer locks the beginning of the corresponding master and slave code words together to maintain frame-for-frame sync between the tapes. In this mode, the slave tape can be deliberately *offset* by several seconds either ahead or behind the master, while maintaining synchronous speed. An offset display indicates the number of frames separating the two codes and whether the slave is lagging or leading the master.

The operation of synchronizers made by different manufacturers varies slightly. With the 3M synchronizer, the tapes are started in the address mode; when sync is achieved, the mode is automatically switched to phase lock so that offsets can be introduced. If no offset is desired, the automatic mode switching can be disabled, leaving the synchronizer in the address mode. The Quad/Eight synchronizer also starts the tapes in the address mode and automatically switches to phase lock when sync is achieved. If the unit mode selector switch



Courtesy Mincom Division, 3M Co.

Fig. 10-9. The 3M SMPTE edit code reader (top) and synchronizer (bottom).

is set to address lock, it will automatically revert to address syncing in the event any code difference arises between the tapes, so offsetting is not possible. The phase lock position of the mode selector prevents the unit from reverting to the address mode once sync has been achieved, permitting synchronous speed to be maintained in the presence of time code discrepancies caused by deliberate offsetting or by splices.

The Quad/Eight synchronizer can maintain a preselected offset through a stop/start cycle of the master tape, while the 3M unit requires that the offset be reintroduced by operation of the front-panel controls each time the master is started. Both units have optional voltage-controlled oscillators which can be used in conjunction with power amplifiers to control slave machines using frequency-controlled servo or hysteresis synchronous motors. Fig. 10-10 illustrates a typical setup for synchronizing two audio recorders using the time code method.

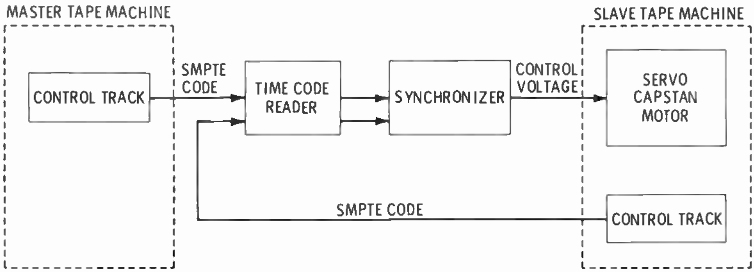
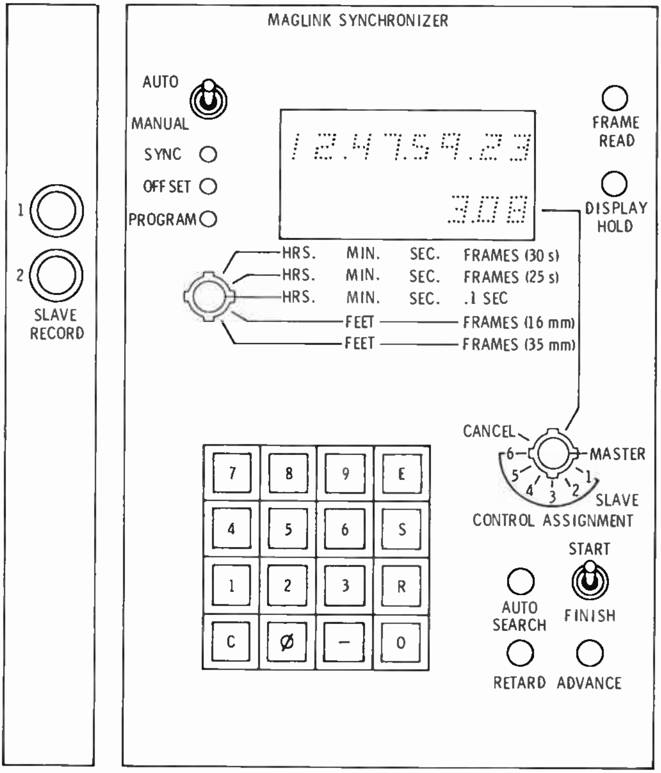


Fig. 10-10. Synchronizing two tape machines using the SMPTE code.

Automated Processes, Inc. has recently introduced their "Mag-Link" time code synchronizer (Fig. 10-11), which is somewhat more elaborate than those described above and uses a different time code. The SMPTE code has certain inherent drawbacks: (1) It creates more cross talk into adjacent audio tracks than results from regular audio programs; (2) Due to its high-frequency content, it cannot be read in fast forward or rewind without defeating the machine tape lifters (causing increased wear on the heads and on the tape) and using special wide-band playback preamplifiers; (3) Its readability can be impaired when dubbed through more than one generation; (4) Its capacity for optional control words is insufficient for some users.

By using a 300-bit-per-second binary address code, which is stored on tape through frequency shift keying (FSK), API avoids these problems. This is a modulation system in which the frequency of a carrier wave is varied between two possible values to represent the two binary states. The API code uses two FSK sine-wave carrier signals modulated at different bit rates and mixed together to keep the frequency of the code components below 3.5 kHz. This code does not create cross talk to adjacent tracks any more than program signals do.



Courtesy Automated Processes, Inc.

Fig. 10-11. The MagLink synchronizer.

A specially designed digital filter enables the carriers to be decoded in fast forward and rewind at speeds from 2 to 500 times that at which the data were recorded without dropping the tape lifters or using special preamplifiers. The control track requires a frequency response of 30 Hz to 4 kHz ± 6 dB with a signal-to-noise ratio of only 12 dB, so multiple generation dubbing will not cause code errors. The limited bandwidth requirement of the system leaves room for many optional control words to be FSK modulated on a carrier wave in the 7- to 15-kHz region and mixed in with the address code signal. In addition, API code resolution is 1/300 of a second as compared to 1/30 of a second for the SMPTE code. A converter is available to translate the SMPTE code into the API code for use with previously SMPTE encoded tapes.

The synchronizer is designed for use with a master and up to six slave machines. In the *manual* mode, the machines are operated independently by their transport controls. When the unit is in the *auto* mode, the slave tapes automatically cue themselves up to the same address as the master tape so that manual parking of the slaves within a capture range is not necessary. When the master is put into the play or record mode, the slaves are locked into sync with it. Sync will be maintained at play speeds within a range of 0.02 to 5 times at which the data was recorded. If the master is put into the fast forward or rewind mode, the slaves will automatically follow it to maintain sync. When the master is stopped, the slaves will again cue themselves up to its address.

The synchronizer contains two eight-digit position readouts. The master readout displays the position of the master tape, except in the search mode when it displays the address being searched for. In the manual mode, the slave readout displays the position of the slave selected by the CONTROL ASSIGNMENT switch. In the auto mode, the slave readout displays the offset between the master and the selected slave.

The READOUT CONVERSION switch determines the significance of the displayed digits. The readout can be in any of the following five units: 1) hours, minutes, seconds, and frames at 30 frames per second (or 29.97 frames per second by using the dropped frame technique which is selectable by an internal switch on the rack-mounted electronics of the system); 2) hours, minutes, seconds, and frames at 25 frames per second for use with European video systems; 3) hours, minutes, seconds, and tenths of seconds; 4) feet and frames of 16-millimeter film; 5) feet and frames of 35-millimeter film.

A DISPLAY HOLD button freezes the reading on the display when pressed so that cue locations can be logged. The last two digits on the right of the readout change rapidly when the tape machines are in motion, so they are blanked out to prevent them from being a distraction. A FRAME READ switch unblanks these digits so that they can be read when desired.

A sixteen-button keyboard is used to select tape positions and offsets in terms of the readout conversion units selected. When a number is entered with the keyboard, the slave readout displays that entry rather than the offset of the selected slave. Numerical entries must be followed by pressing a command button such as ENTER, OFFSET, or SEARCH to have any effect on tape position. After a command button is pressed, the slave readout resumes display of the offset of the selected slave. The CLEAR button is used to cancel the keyboard entry if a wrong number is accidentally entered.

The SYNC light illuminates when the speed of the selected slave is synchronized with the master, while the OFFSET light illuminates when the address of that slave differs from that of the master. Offsets can be introduced between the master and a selected slave machine by using the RETARD and ADVANCE buttons. The slave will maintain the offset when the button is released. Alternatively, a specific offset can be introduced by entering the desired amount into the synchronizer memory with the keyboard and pressing the OFFSET button. The slave will then assume a position offset ahead of the master by the selected amount. If the MINUS button is pressed before entering the amount of offset, the slave will be offset behind the master by the selected amount. When the slave achieves the desired offset and is running the same speed as the master, both the sync and the offset lights illuminate.

MagLink can be used to find particular locations on a selected slave by entering the desired address with the keyboard and then pressing the SEARCH button. The slave will wind to the desired address in fast forward or rewind and then stop there. If the CONTROL ASSIGNMENT switch is in the *master* position, the master and all of the slaves will wind to the selected address. The AUTO SEARCH button automatically enters the current address of the tape into the memory without having to use the keyboard. When the SEARCH button is pressed at a later time, the tape will wind to the address at which the AUTO SEARCH button was pressed. This feature is useful in overdubbing or mixing, where the tape is to be rewound to the same point over and over.

An additional feature of MagLink is that its memory can be preprogrammed to synchronize any selected sections of the slave tapes to any desired sections of the master. The point on the master tape at which the slave tape is to be in sync is selected with the keyboard and recorded in the preprogramming memory by pushing the ENTER button with the SYNC COMMAND switch in the *start* position and the control assignment set to *master*. The address on the slave tape which is to be locked in sync with the master address set above is then entered with the sync command switch set to *start*, and the control assignment set to the appropriate slave. The address at which the slave is to unlock from the master is preset by assigning control back to the master and entering the unlock address with the sync command switch in the *finish* position. A program entry identification number is then assigned to the lock-unlock commands by entering the number with the sync command switch in the finish position and the control assignment switch set to the appropriate slave.

Many lock-unlock commands can be preprogrammed, and they will be executed in the sequence they were entered. The position readout flashes if the memory capacity of the unit is exceeded, but, as mentioned before, the capacity can be expanded by connecting additional memory cards. When the master tape is started, the slave winds to its first preset "start" location and stops. When the master reaches its first "start" location, the tapes are locked into sync. When the master reaches the preset "finish" address, the slave unlocks, winds to its next preset "start" location, and waits there until the master reaches its next "start" location. The slave is actually stopped ten seconds ahead of its "start" address and put into the play mode ten seconds before the master reaches the master "start" address. This gives the slave time to get up to speed and lock into sync with the master before the tapes reach their sync start points. The PROGRAM light illuminates when the preprogrammed commands are being executed. Depressing the enter button with the control assignment switch in the cancel position clears the preprogramming memory.

Preprogramming is used mainly in video tape work for electronic editing, but it can also be used for audio work. Electronic editing refers to a procedure in which the tapes are not physically cut, rather sections of the different reels of tapes are combined by electronically switching from one machine to another at certain predetermined points on the tapes. The results are monitored and the edit points adjusted until they are correct. The tapes are then played again, and the results of the switching are recorded on another machine to produce the finished tape.

In audio work, this procedure can be used for editing and combining sections of different takes of a song to produce a complete performance. The advantage of electronic editing for audio is that the edited section can be heard exactly as it will be in its final form without having to cut the tape. Other applications for preprogramming are adding sound effects or instrumental parts recorded in random order (on one proper place) onto a third tape that is syn-

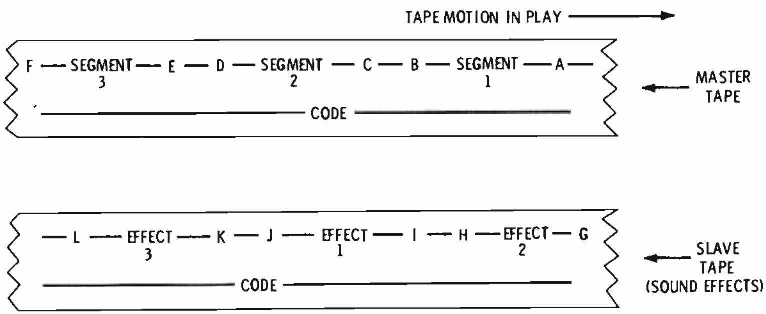


Fig. 10-12. Preprogramming sound effects on slave tape.

chronized with the master. Fig. 10-12 shows sound effects 1, 2, and 3 recorded in random order on the slave tape which have been preprogrammed to synchronize with master tape segments 1, 2, and 3, respectively. When the master tape is put into play, the slave is cued up to point I. When the master reaches point A, the slave is locked to it and sound effect 1 plays in sync with master tape segment 1. When the master reaches point B, the slave tape unlocks, rewinds, and cues up to point G while the master plays the tape between points B and C. When the master reaches point C, the slave is locked to it and sound effect 2 plays in sync with master tape segment 2. When the master reaches point D, the slave unlocks, winds forward, and cues up to point K while the master plays the tape between points D and E. When the master reaches point E, the slave is locked to it again, and sound effect 3 plays in sync with master tape segment 3 until the master reaches point F where the slave unlocks and stops.

The RESET button on the synchronizer control panel sets the code generator back to zero and also clears the system memory. The code can be started at any desired count by entering the starting point with the keyboard. In addition, a feature called "Jam Sync" enables the code generator to synchronize its count with that of an existing API control track. This permits the count to pick up from where it previously stopped, rather than rerecording from the head of the tape which would waste time and possibly change cue addresses. Thus, even though the tape machine may be stopped between takes during a recording session, the code can still be consecutive throughout the reel. If the code was reset to zero at the head of each take, the synchronizer could be used to find sections of that take, but not of previous takes. MagLink does not require a consecutive count in order to maintain sync, but the consecutive count does make editing easier.

Like the other synchronizers described, MagLink requires a connection to the slave machine capstan motor. It is connected directly for servo motors and through an accessory VSO for synchronous motors. Connections are also made to the remote controls of both the master and the slave machines. The record buttons on the MagLink front panel are used to activate the record mode on the corresponding slave machines.

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CHAPTER 11

Automated Mixdown

In multitrack recording, the final musical product is not realized until the mixdown stage. Each track of the tape has its own volume, echo send, front to rear and left to right pan pot, as well as several EQ frequency and amount of boost or cut selectors. Multiply these functions by the 16, 24, or more tracks of instrumentation currently used in recording, and add to it the echo return volume, panning, and EQ controls, compression/limiting, and other signal-processing functions, and it becomes obvious that the mixdown process is more than one engineer can efficiently handle. As a result, mixdowns must be rehearsed repeatedly so that the engineer can learn which controls must be operated, how much they should be varied, and at what point any changes come, relative to the recorded tracks. It is not uncommon for 12 hours to be spent mixing a complicated piece of music before an acceptable mix is obtained. Mixes must often be rejected because the engineer simply forgot to make one important control change. The producer and engineer know how and when the control settings should be changed, but the memory and physical dexterity required to execute them can exceed human capabilities.

The solution to this problem is the use of equipment that can remember and re-create any settings and changes made by the engineer, while allowing him to improve on them one by one until he achieves the desired final mix. This technique is called *automated* or *computerized mixdown* and is made possible through the use of *voltage-controlled amplifiers* or *attenuators* (*VCAs*), *voltage-controlled equalizers* (*VCEs*), a *programming encoder/decoder*, and a data storage device called a *memory*. A VCA controls the level of audio signal as a function of a dc voltage applied to its control input, while a VCE changes the frequency response of an amplifier as a function of the dc voltages applied to its control inputs. In a console fully equipped for automated mixdown, VCAs and VCEs perform the function of faders, switches, and equalizers, while the fader, switch, and equalizer controls are used to vary dc control voltages.

AUTOMATED CONSOLES

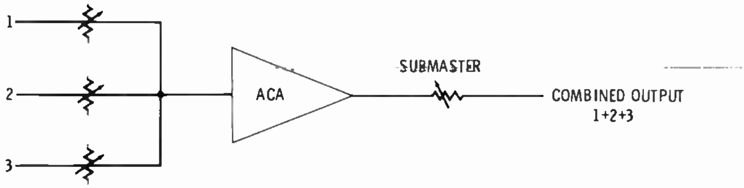
The VCAs permit the *free grouping* of signals. Since there is no leakage of the audio signal into the VCA control input or vice versa, the dc voltage at the control input of any number of VCAs can be controlled by a single variable-voltage source while maintaining complete separation of the audio signals being controlled. The advantage to this is that after the levels of several instruments are set relative to each other, their overall level can be controlled by a single fader rather than having to change each individual instrument. Some methods of this technique are illustrated in Figs. 11-1A, B, C, and D. In Fig. 11-1D, the output of each VCA is proportional to the sum of its channel and group control signals. Attenuation is 0 dB when the sum of the control voltages is zero volts. As the sum increases, so does the attenuation.

In a console not using VCAs this can be done in two ways: either the signals are mixed into one channel of the console and the mixture fed through a fader to control the overall level, or the signals are kept separate and are patched to a multichannel fader. In the latter method, the number of tracks controllable by one fader depends on the number of channels available in the multichannel fader. With voltage control, the level of four instruments placed in the four different channels of a quad mix can be controlled by a single-channel *submix* or *grouping* fader. By controlling the dc voltage feeding the grouping fader, the VCAs can all be turned on or off simultaneously with one switch. For example, if a horn section recorded on several tracks is to begin playing in the middle of a song but some horns begin to play slightly early, the engineer could leave their VCAs at full attenuation until the moment the horns are to play and then punch on all of them simultaneously with one switch. Alternatively, if one track had no noise and began to play at the proper time, a noise gate could be connected to it and the noise gate control voltage used to turn on all the other tracks when the noiseless track begins to play. Control inputs are grouped by providing several dc grouping busses on the console. All VCAs assigned to the same bus can be controlled by a single voltage source.

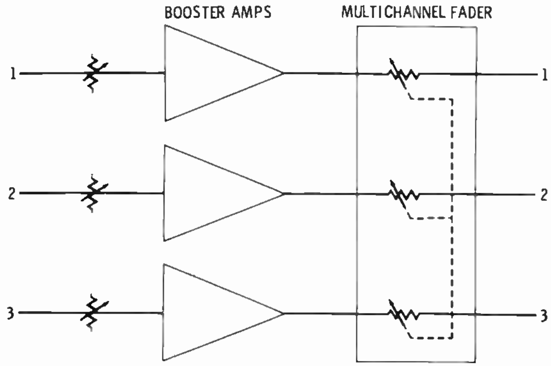
MODES OF OPERATION

Write Mode

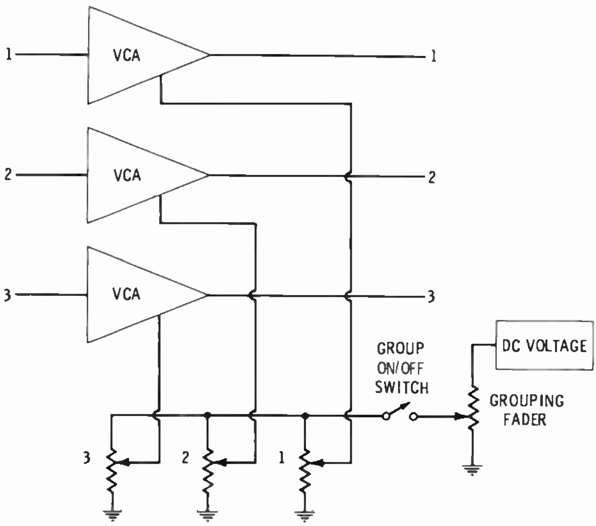
The automated console has three different modes of operation: write, read (or safe), and update. In the *write mode*, the program-



(A) By mixing them together and using a submaster.



(B) Retaining separation by using a multichannel fader.



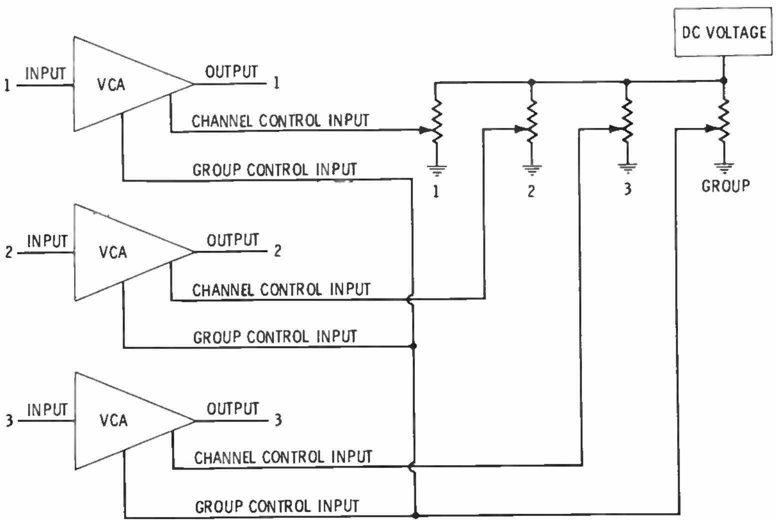
(C) Retaining separation by using VCAs and a grouping fader to control the level of the voltage fed to the individual dc control faders which in turn feed the VCA control inputs.

Fig. 11-1. Varying the levels of

mer scans the control input of each of the voltage-controlled devices of the console in a certain sequence and *encodes* the control voltage information into a form that can be stored for later use. The memory unit used for storage is usually an audio tape recorder, and the track on which the control information is recorded is called the *data track*. Since each control voltage represents the position of a console control, the data track contains a record of how the console controls were set during each scan cycle. In order for the settings to be reconstructed, the data must be synchronized with the program material, and it is therefore recorded on either a spare track of the multitrack master or on a separate tape machine that is interlocked with the master.

Read Mode

In the *read mode*, the console controls no longer supply dc voltages to the respective control inputs. Instead, the programmer reads the control information from the data track and feeds it through its *decoder* section which reconverts it into dc voltages and connects these voltages to the appropriate voltage-controlled device inputs. As the master tape and data track are played, the dc voltages produce a mix identical to the one that produced the data track. If after listening to the tracks mixed automatically in the read mode, the engineer and producer decide certain functions need to be com-



(D) Retaining separation using VCAs with several control inputs.

several signals with one control.

pletely changed, the write mode can be initiated on these functions only, while retaining automatic control of the others by leaving them in the read mode. The dc voltages created by the decoder and by manual operation of the console are then encoded and stored on a second data track. By alternating back and forth between two data tracks, improvements can be made until the desired mix is achieved.

If only part of the mix needs to be reprogrammed for a certain function, the write mode can be entered on that function at the point the changes are to be made. In order to provide a smooth transition from automatic to manual control with no jumps in level, the manual control must be set so that the dc voltage it creates equals that re-created by the decoder. This is done by adjusting the manual control to null a meter (or indicator lights) connected to read the difference between the automatically and manually generated voltages for each function, or it can be done by adjusting the manual control to illuminate a level match indicator light. Once level match is achieved, the engineer can switch from read to write on a function without the levels changing. After the write mode is entered, the controls may be moved from the level match position to the new settings desired.

Update Mode

The *update* mode eliminates the need to match the manual setting to the automatic one when punching into an existing data track. In this mode, settings are changed by adding to or subtracting from the automatic control voltage rather than completely rewriting them. The advantage to this is that if some involved level changes were made correctly except for the loudness or softness of the track, the overall volume could be changed without having to perform all the other level changes. The changes are reconstructed from the previous data track, raised or lowered in level by the manual update control and recorded on another data track. A level match between the new and old control voltages is achieved by setting the manual control to a position indicated by the manufacturer of the automated console to cause no increase or decrease when the update mode is entered. Once in the update mode, the control is operated to change the programming as desired and then returned to the update level match setting to resume the levels of the previous mix. At this point, either the read mode can be initiated on this function, or the update mode can be retained with the manual control left at the neutral point.

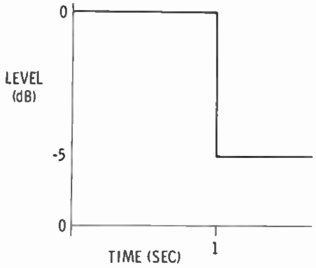
Since at least two data tracks are required in order to store and update a mix, only fourteen tracks are left for audio on a 16-track tape. If a separate mono and/or stereo single mix are desired in addition to the stereo LP mix, even fewer tracks can be used for recording the music. The solution to this is to either provide for a larger number of tracks than is needed for the music, leaving some tracks blank (such as using a 24-track machine and recording music on only 20 tracks) or to synchronize a second audio recorder with the first for storing the data by recording a *sync track* on one track of the music tape and one track of the data tape. This allows instruments to be recorded on all but one track of the music tape regardless of how many separate mixes are to be stored.

SCANNING

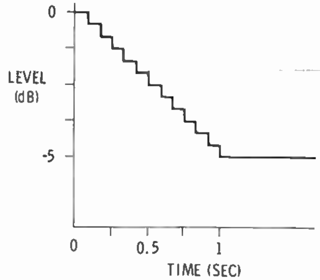
The programmer senses all of the control voltages used in the console and encodes them into a form which can be stored and decoded at a later date. The number of control voltages needed in an automated console makes it prohibitively expensive to monitor each control input on a continuous basis. Instead, the programmer samples the dc level applied to each control device in a preset order. Each device is sampled for the same amount of time, usually a small fraction of a second. This sequential sampling is called *scanning* or *multiplexing* and produces a single channel of output data which can be broken down by the decoder to reproduce the original dc levels. The time required to scan all of the control inputs in the console once is called the *scan rate* or the *updating rate*. As used here, the term *updating* refers to the repeated sampling by the encoder of control voltages one time during each cycle to create new data for storage. Updating also refers to the repeated modification of the reconstructed control voltages in response to each new cycle of data fed to the decoder.

The updating rate must be fast enough to provide smooth and accurate encoding of the engineer's actions. For example, if the engineer fades out a level control at a rate of 5 dB per second and the encoder scans the corresponding control voltage only once each second, the programmer will sense and encode 5-dB steps rather than a continuous fade (Fig. 11-2A). If the scan rate is increased to 50 milliseconds, the programmer would process steps of 0.25 dB which more closely approximates the continuous fade (Fig. 11-2B)[2]. The decoder output can be filtered to remove these small steps from the reconstructed dc voltage so that level changes are smooth. The higher the scan rate, the less filtering needed because very small steps are perceived as a continuous change. The scan rate is limited by the storage capacity of the memory unit as well as by the time needed by the processor to encode or decode a voltage.

The programmer processes two different types of functions. *Switching functions* have only two states and can be used to turn signals on or off (Fig. 11-3A) where continuous control is not de-



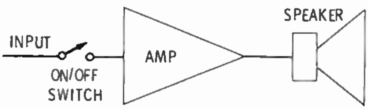
(A) Updating occurs once each second. The fade is encoded as a step change of 5 dB.



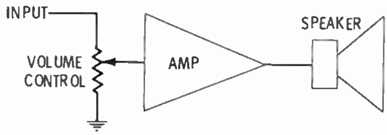
(B) Updating occurs every 50 milliseconds. The fade is encoded as 20 steps of 0.25 dB each.

Fig. 11-2. A 5 dB per second fade of one-second duration.

sired or used to control one step of a multiposition switch such as in an equalizer stepped frequency selector. A *dynamic function* controls devices that can be set to any position between a maximum and a minimum, such as level or gain, panning, or parametric EQ (Fig. 11-3B).



(A) A switching function.



(B) A dynamic function

Fig. 11-3. Two functions a programmer processes.

ENCODING AND DECODING

Since the signals being scanned are dc voltages, they must be encoded into another format before they can be recorded on an audio recorder. Two methods of encoding are currently in widespread use. The first encodes the analog dc levels as *digital binary words*. Since digital words by their very nature can express only step changes rather than continuous changes, switching functions are encoded perfectly, but the accuracy of the encoding of dynamic functions depends on the size of the steps. Expressed in dB, the step size is called *resolution*. Since longer words can represent more steps, step size can be decreased by adding more *binary digits* or *bits* to the word. The bit capacity of any system is limited however; so increasing the number of bits per word decreases the number of functions which can be programmed.

Since convenience and cost require that the words be recorded on an audio recorder rather than in a computer-type memory, the *bit rate*, i.e., the number of bits created by the encoder per second, cannot exceed the *storage density* of the recorder. In other words, the frequency bandwidth of the recorder must be equal to, or greater than, the bit rate. For a rate of 9600 bits per second, a data channel bandwidth of about 10 kHz is needed. The code is usually set up so that step size becomes larger where control settings are not as critical. For example, at attenuation settings below −70 dB the steps may be as large as 2 dB each while they may be 0.25 dB at −10 dB. Thus bits are not wasted on settings which are so low that they can barely be heard; they need to be present only to provide a smooth fadeout. Since switching functions are either on or off, they require only one bit of storage and many more of them can be automated than dynamic functions.

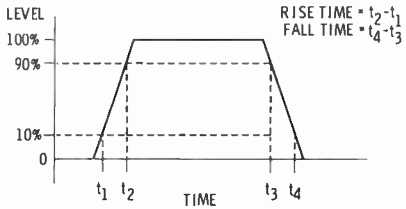


Fig. 11-4. Illustrating rise and fall times of a pulse.

Each bit of data is represented by a current or voltage pulse. In order to insure accurate digital processing, these pulses must have rapid *rise* and *fall times*. Fig. 11-4 illustrates that the rise time of a pulse is the time needed for the level to increase from 10% to 90% of the maximum value and the fall time is the time needed to decrease from 90% to 10% of the maximum.

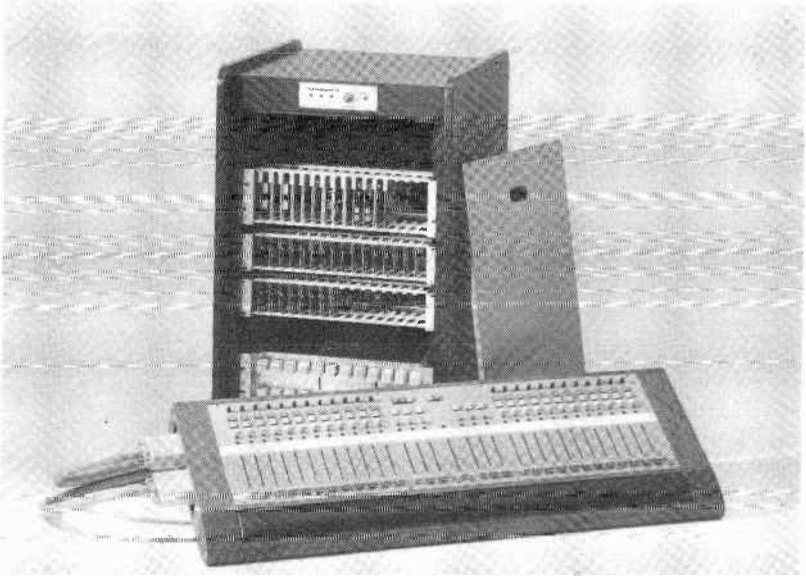
Fig. 11-5. A digital signal conditioner shapes the pulses played back from the tape to make sure that they have fast rise and fall times.

Although the encoder generates pulses of the proper shape, any nonlinearity of the electrical phase response in the tape recorder used for storage can distort this shape. To prevent possible errors, the decoder contains a *digital signal conditioner* which generates fast rise and fall time pulses from the distorted pulses on the tape before the data is decoded into analog form (Fig. 11-5). The generation of new pulses also prevents the errors that would result if tape noise was allowed to build up to the level of the pulses as a result of bouncing mix information back and forth between data tracks.

The second encoding method uses the digital technique in combination with analog representation. Instead of using the two value per digit binary code, this system uses a code which permits a greater number of values to be represented per digit. For example, a modulo 5 or quinary code digit can have any one of five possible values (0 through 4). Thus, a three-digit quinary code word can express 124 different values while a seven-digit word would be required to express this number of values with the binary code. The digit values are represented through the use of a single-frequency sine wave with five possible values of amplitude. The level of one cycle of the wave represents the value of one digit. Each code word consists of four cycles, the first of which is a level reference to guard against dropouts and other sources of data error, while the next three cycles represent the value of the word. The code words are separated from each other by four cycles of the wave at the reference level to aid in synchronizing the system.

Using magnetic tape as the storage medium subjects the data to the problems of dropouts and magnetic discontinuity at splices. Since the programmer depends on the sequence of the recorded data to determine which corresponds to which function, a temporary loss of signal at the input of the decoder due to a dropout or a splice could cause an error in the assignment of control voltages. To prevent this, a *parity checker* is included in the programmer which senses *error signals* such as the absence of input to the decoder or noise which could otherwise be misinterpreted as data. The parity checker causes the programmer to hold all levels at values determined by the last complete scan. The length of time that the program can hold previous values while awaiting valid data is called the *error signal holding time* and is expressed in the number of dB change in control settings per second during a data dropout. When valid data is again received by the decoder, control voltage updating resumes.

The automated mixdown system available from Quad/Eight Electronics is called *Compumix* and uses a biphase digital programming format which is compatible with SMPTE time coding. The system is designed for use with existing nonautomated consoles and consists of a *controller* and a *processor* connected by two multiconductor cables (Fig. 11-6). The controller varies only dc voltages, while all VCAs and audio lines are part of the processor. The processor is connected between the line outputs of the multitrack tape machine and the line inputs of the existing console. The faders of the existing console are set to provide unity gain for the input signals. If they were left at maximum attenuation, no signal would reach the console output. Similarly, moving these faders during the mix would cause changes in balance which would not be encoded

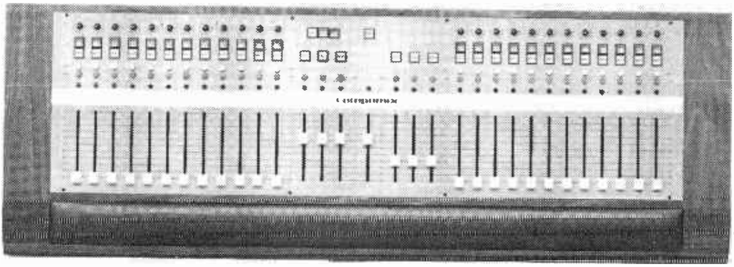


Courtesy Quad/Eight Electronics

Fig. 11-6. The Compumix automated mixdown system.

or stored. The reason for connecting Compumix in this manner is to minimize the changes that are needed in the existing control room system to install automation. A master keyswitch on the processor connects the processor inputs directly to its outputs for manual operation of the existing console, bypassing the automation electronics entirely.

The controller is a portable console measuring 47 by 16 by 3 inches (Fig. 11-7). It can be supported by its own detachable legs



Courtesy Quad/Eight Electronics

Fig. 11-7. The Compumix controller.

or placed on top of the existing console. This latter is the more usual position since the unit contains only level control and switching functions. Panning, EQ, echo send, and echo assignment must be set manually on the nonautomated console. In its basic form, the Model 2400 controller is a 24-input unit, expandable to 32 inputs (the Model 3200), or it can be modified to automate certain stereo or quad panning functions. Each input channel has a linear motion fader which controls the dc voltage fed to a VCA, an on/off switch for this voltage, and a control-voltage grouping bus selector switch which enables the voltage feeding that channel input fader to be varied by any one of the six submix faders provided. On/off switches are provided for each submix fader to shut off all VCAs assigned to that submix bus. A master fader is also provided for fading out the mix as a whole by simultaneously varying the dc voltages applied to all of the VCA control inputs. The controller also provides a second push button above each input channel on/off switch to control an auxiliary on/off switching function which can be set up by the individual studio to control any external switching desired. For example, the existing console could be modified to turn the echo send on and off in response to the voltage generated by this function.

Each input channel has a write and an update switch to initiate these modes, while the submix faders have only update switches. In the update mode, the −15-dB point on the input and submix faders provides a level match with the previous mix. A light emitting diode above each fader illuminates when the fader is set at this point. A master read switch initiates the read mode on all controls for automatic playback of a mix, while a master write switch initiates the write mode on all functions. Additional switches start the master tape machine and put the data track into either the safe or the record mode.

The Compumix processor capacity is 504 bits. A switching function requires 1 bit, while a level control function needs 8. The basic 24 input system with the 24 auxiliary switching functions uses up $24 \times 8 + 24 = 216$ bits, leaving 288 in reserve for expansion. The system can process a maximum of 63 dynamic functions if no switching functions are used. Note that the submix, master, and input channel on/off functions do not use any extra processor capacity since they merely change the dc levels applied to the VCAs. The processor cannot distinguish their use from the use of the input faders, so their effect is encoded in the input level control function.

Each function is updated every 50 milliseconds, and this scan rate remains constant regardless of the number of functions encoded. The bit rate is 9600 bits per second, requiring a data track bandwidth of just less than 10 kHz. The encoder resolution is ±0.1 dB for static control settings from 0- to 45-dB attenuation and infinite while the level controls are in motion. The LED level match indicator is accurate within ±0.25 dB, and the compatibility of the level control function from one Compumix system to another is ±0.5 dB. The processor is protected against dropouts and splices by a data validation system which will hold the last correct control settings within 0.25 dB for a 10-second data dropout. A data light flashes to indicate the operation of the protection circuitry in the presence of an error signal. Other lights are provided to indicate system overload, bypass, and sync. The Compumix processor will decode accurately in the presence of speed variations of as much as ±15%.

The Allison/Automated computerized mixdown system is a joint effort of Automated Processes, Inc., which designed the audio control equipment, and Allison Research which designed the programmer. This system differs from that of Quad/Eight in that the audio controls and electronics can be incorporated into the existing studio console rather than being connected as an add-on device between the multitrack tape machine and the console. The Allison/Automated programmer uses the combination of digital and analog encoding, rather than a strictly digital format.

This system is built around the Model 940 automated fader which contains a linear motion conductive plastic fader, a VCA, an auto/manual mode switch, a write switch with LED indicator, an update switch with LED indicator, a safe switch, a meter, and two null-indicator LEDs (Fig. 11-8). Although it is designed for individual channel level control, the 940 can be connected for use as a grouping or master fader by varying the control voltage fed to the other 940s. Its VCA can also be connected to external control voltages for channel muting or noise gating functions. Since the 940 requires the same console panel area as most standard faders (7 by 1½ inches), it can easily be retrofitted into existing consoles or wired into new consoles with no changes in console layout or circuitry, except for wiring in a connector that mates with it and adding the wiring that powers the unit and interconnects the VCA with the programmer. Other modules are available to automate echo send and panning.

In the manual mode, the audio signal passes through the fader and completely bypasses the automation electronics. In the auto mode, the meter illuminates and the audio signal flows through a VCA, while the fader controls a dc voltage applied to the VCA control input. The write switch causes the fader setting to be recorded on the data track, the safe switch allows previous fader settings to be played back, and the update switch enables the previous mix levels to be increased or decreased relative to the update level match setting which is −15 dB on the fader. Master safe and write switches can be wired into the system to control all of the 940s simultaneously.

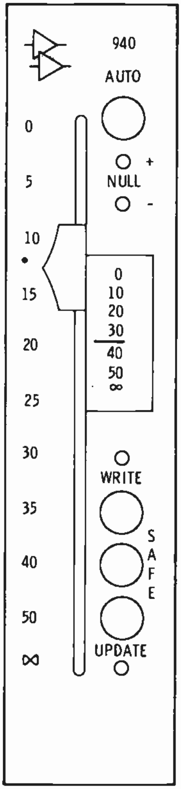


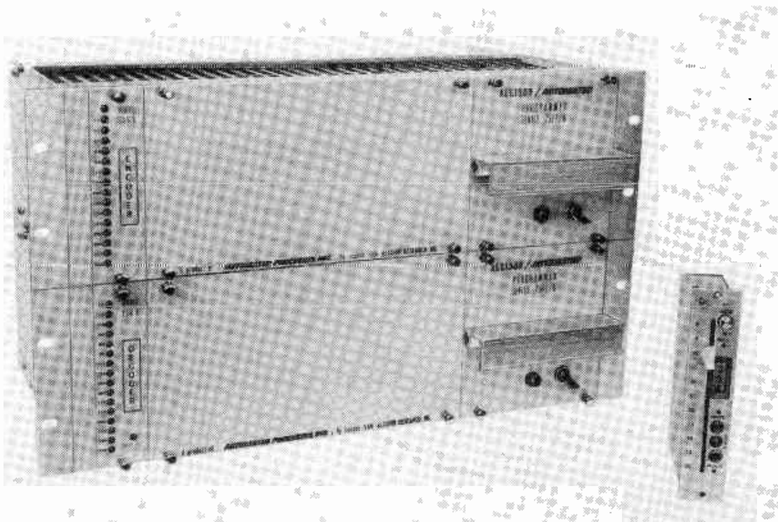
Fig. 11-8. The API Model 940 automated fader.

Courtesy Automated Processes, Inc.

When the 940 is in the safe mode, its meter displays the VCA attenuation settings of the previously stored mix. This enables the engineer to determine whether volume changes are due to instrument dynamics or to level control changes recorded on the data track. Current VCA attenuation is displayed in the write and update modes. Level match is achieved using the LED null indicators by adjusting the fader to the setting that causes both of them to illuminate. If the "+" LED is on, the current fader setting is higher than the previous one, while if the "−" LED is on, the fader level needs to be increased to match the previous setting. A mix can be changed by reading the old data, adjusting the fader to the exact previous level by watching the null LEDs, and then entering the write mode. After the desired change has been made, the engineer can match levels again and punch back into the read mode. Although using the update mode is easier than the above procedure, the gain available in update is limited to 15 dB above that of the previous mix because the update level match point occurs 15 dB below the highest setting on the fader. To achieve more than 15 dB increase in level, the fader settings must be matched with the meter and the write mode used.

The Allison/Automated Model 256 programmer consists of an encoder and a separately packaged and powered decoder (Fig. 11-9). The units are separate to enable encoding at one end of a telephone line and decoding at the other end for remote control applications, as well as for applications which only require the decoder, such as mastering a disc directly from a multitrack tape. The capacity of the programmer is determined by the number of plug-in switching cards used. The system is expandable in groups of 16, from 16 to 256 dynamic functions, or in groups of 90, up to 1440 switching functions, or a combination of the two. The function utilization format is based on 24-track capability, as follows: Functions 1 through 24, individual channel gain; 25 through 48, individual channel echo send; 49 through 72, individual channel left to right panning; 73 through 96, individual channel front to rear panning; 97 through 192, individual channel EQ with four functions per channel; 193 through 216, master levels and echo returns; 217 through 256, auxiliary console functions. Adoption of this set function layout is necessary to make tapes that are mixed in different studios compatible.

Since the programmer capacity is variable, the scan rate depends on the number of functions available and the sampling time per



Courtesy Automated Processes, Inc.

Fig. 11-9. The Allison/Automated Model 256 programmer.

function. Sampling time can be set by an internal switch on the master encoder to either the "A" rate of 500 microseconds per variable function or the "B" rate of 800 microseconds per variable function. Provision is also made for operation at other sampling rates. At the "A" rate, a 16-function scan updates each function every 8 milliseconds, while a 64-function scan updates every 32 milliseconds, and a 256-function scan updates every 128 milliseconds. Updating times for the "B" rate would be 12.8, 51.2, and 204.8 milliseconds, respectively.

Both the encoder and decoder have LED indicators which flash to indicate the number of functions being encoded or decoded. The decoder automatically adjusts its scan rate to the number of functions recorded on the data track. A 256-function decoder, for example, would cycle at a 16-function rate to decode a 16-function data track, and the LEDs would indicate only 16 functions being decoded. This feature prevents existing data tracks from becoming obsolete if the system capacity is expanded and also indicates the degree of automation on a tape that is unfamiliar to the engineer. The programmer will decode any sampling rate between 1600 and 400 microseconds per function and permits tape speed variations of +20%, −60% for "A" rate and +100%, −50% for "B" rate, without impairing the accuracy of the stored data.

The use of combined digital and analog encoding reduces the data track requirements to a bandwidth of 8 kHz for "A" and 5 kHz for "B" rates, with a signal-to-noise ratio of 35 dB, making synchronized cassettes a possible storage medium. The data track can be recorded at any level between −20 and 0 VU, and the decoder is insensitive to level fluctuations within this range. Any disturbance which could cause a decoding error, such as a splice or dropout on the data track, triggers a protection circuit which holds the last correct settings until error-free data is available. System resolution is ±0.5 dB for settings between 0 and −50 dB, and ±2 dB for settings between −50 and −100 dB. The accuracy of the reconstructed levels is within 0.5 dB of the encoded levels, and inaccuracies are noncumulative. All Allison/Automated programmers are compatible with each other within ±0.5 dB, to ensure accurate decoding of a mix on a system other than the one used for encoding.

In both of the previously mentioned systems, the mix data is completely regenerated each time it is bounced to another data track. This is done by decoding the old data to re-create the original dc levels which are needed to control the VCAs set to read, and then these dc levels, plus any changes due to the use of the write or update mode on any channel (Fig. 11-10), are re-encoded. This constant regeneration of the data can cause mixing problems if the encode/decode errors are cumulative. For example, if each encode/

decode cycle results in the decoded level being 0.1 dB higher than the original, bouncing back and forth between data tracks 30 times in the course of achieving the final mix would result in levels that had not been rewritten or updated since the first encoding, being 3 dB higher than they should be.

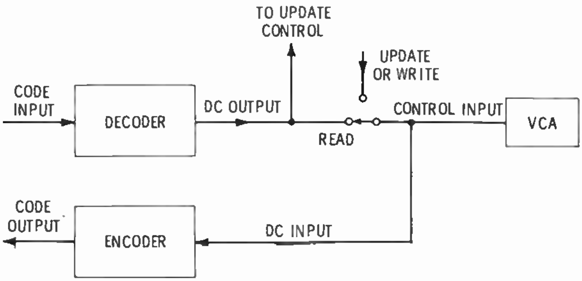
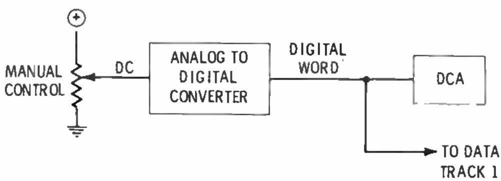
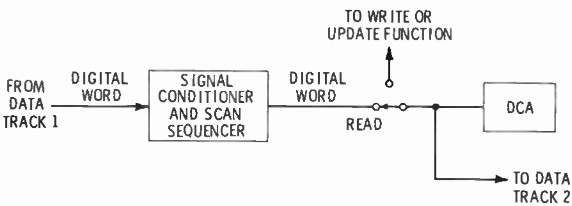


Fig. 11-10. The data for a function in the read mode is decoded and then re-encoded when recorded onto the other data track.

A third method of automation avoids any inaccuracies resulting from repeated analog to digital to analog conversions. Rather than generating dc levels to control VCAs, this system has manual controls to generate digital words which vary *digitally controlled attenuators* (DCAs). In Fig. 11-11A, the dc output of the manual control is scanned by an analog-to-digital converter which creates a digital word that controls the attenuator and is recorded on tape. Fig. 11-11B illustrates the read mode. The pulses from data track 1 are reshaped by the signal conditioner, applied to the proper DCAs



(A) Write mode.



(B) Read mode.

Fig. 11-11. Functions of digitally controlled attenuators.

by the scan sequencer and recorded on data track 2. When bouncing data from one track to another, the old data passes through a signal conditioner and a scan sequencer which feeds it to the appropriate DCA. The DCA is controlled by this digital word or by a digital word generated by the write or update controls. The word at the control input of the DCA is recorded on the new data track. Since the control data format is the same as the storage format, no encoding or decoding is needed, eliminating a possible source of control inaccuracy.

Automation allows the producer much more freedom than previously available. If he listens to a completed mix at home and decides that one instrument needs to be louder in a certain spot, he can go back to the studio and make this change without spending hours re-creating the recording session. Remixing songs for release as singles becomes relatively easy since the original mix is already set up and only the changes need to be programmed.

Before the availability of automation, only one master mix tape was made. This tape, already two generations removed from the live performance, was used to master the disc. It also was used to produce third-generation tapes to press discs in other countries as well as for use in producing fourth-generation reel-to-reel, cartridge, and cassette tape copies. The situation became even worse if the multitrack tape was mixed down into quad and this quad tape had to be mixed into stereo or mono.

Automation enables the engineer to make second-generation masters as easily as dubs. In fact, by sequencing the songs in the proper order, discs can be mastered directly from the multitrack tape, bringing the finished record one generation closer to the live performance (Fig. 11-12).

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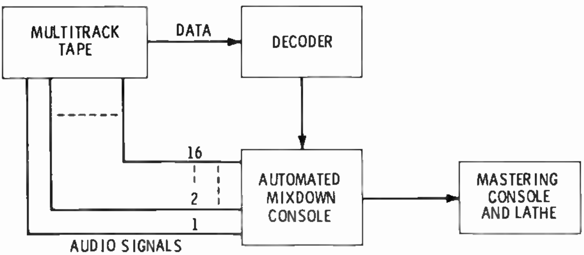

Fig. 11-12. Mastering a disc directly from a multitrack tape.

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CHAPTER 12

Disc Cutting and Pressing

High-quality records are easier and less expensive to produce in large numbers than tapes of equal quality. As most consumers listen to records rather than tapes, it is economically profitable to transfer the final approved mix of a performance from the master tape to a disc.

DISC CUTTING

As the master tape is played on a tape machine, its signal output is fed through a *disc mastering console* to the *disc cutting lathe*. Here the electrical signals are converted into the mechanical motion of a stylus and are cut into the surface of a lacquer-coated recording disc.

Unlike tape which maintains the chronological order and duration of recorded information by relating periods of time to physical lengths of tape, discs relate periods of time to the angle of disc rotation (Fig. 12-1). As the turntable rotates at a constant *angular velocity* such as 33⅓ or 45 rpm, the stylus gradually moves closer to the disc center, cutting a continuous spiral into the disc surface. The time relationship of the recorded material can be reconstructed by playing the disc on any playback turntable that has the same constant angular velocity as the one used to record the disc.

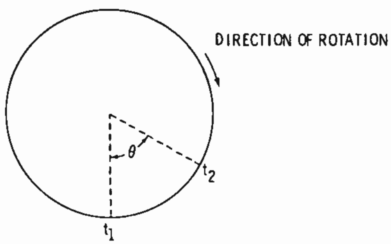
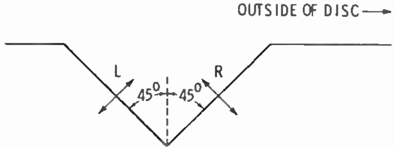
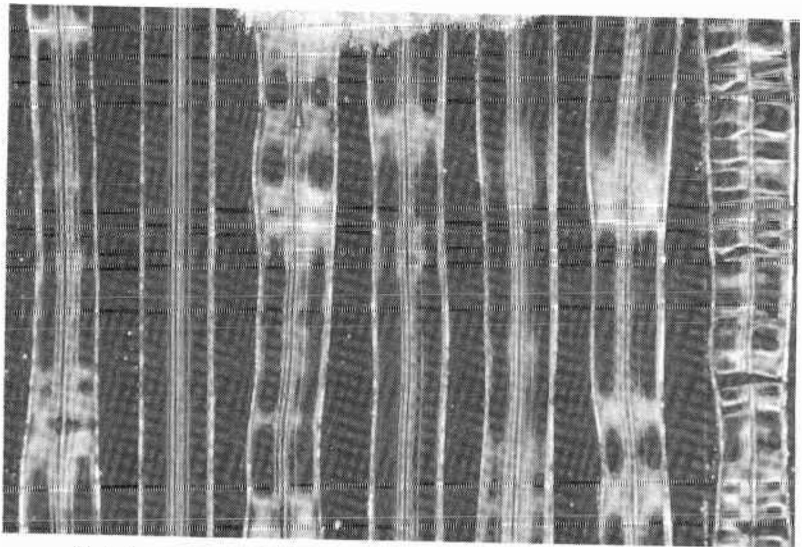


Fig. 12-1. A period of time is represented by a disc rotation of angle θ.

The system of recording used for stereo discs is the 45/45 system. The recording stylus cuts a 90° angle groove into the disc surface so that each wall of the groove forms a 45° angle with vertical. Left-channel signals are cut into the inner wall of the groove, and right-channel signals are cut into the outer wall (Fig. 12-2A). Fig. 12-2B is a photomicrograph of grooves showing recorded stereophonic material. The stylus motion is phased so that a signal which is in

(A) Left-channel signals are cut perpendicular to the inside groove wall, while right-channel signals are cut perpendicular to the outside groove wall.





(B) A photomicrograph of grooves modulated with stereo program material.

Fig. 12-2. Illustrations of stereo record cutting.

phase in both channels (a mono signal or a signal centered between the two channels) produces lateral motion of the groove (Fig. 12-3A); out of phase (channel difference information) signals produce vertical motion, i.e., changes in groove depth (Fig. 12-3B). Because this system is compatible with mono disc systems which use only lateral groove modulation, a mono disc can be accurately reproduced with a stereo playback cartridge. However, unless monophono cartridges are designed for stereo compatibility, they will reproduce only lateral motion and resist movement in the vertical direction. This resistance will damage the stereo information con-

(A) In phase. (B) Out of phase.

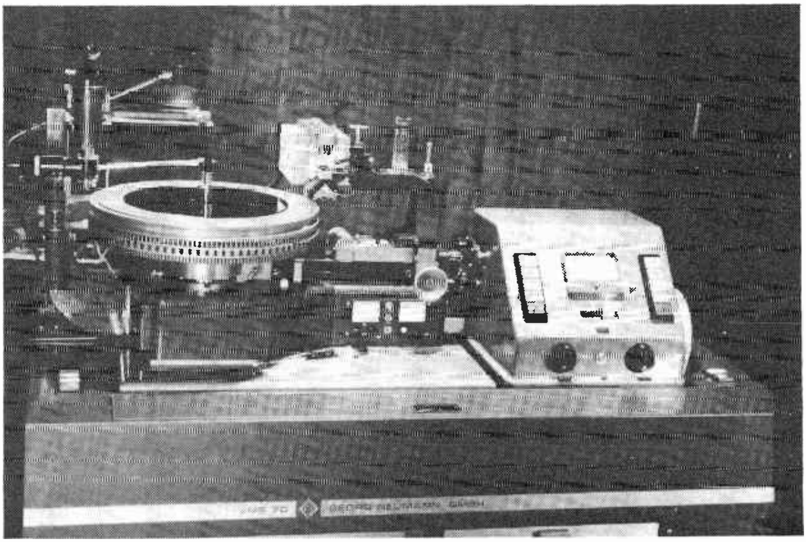
Fig. 12-3. Groove motion in stereo recording. The solid line is the groove with no modulation.

tained in the vertical motion of the groove if a stereo record is played with a mono cartridge.

Cutting Lathe

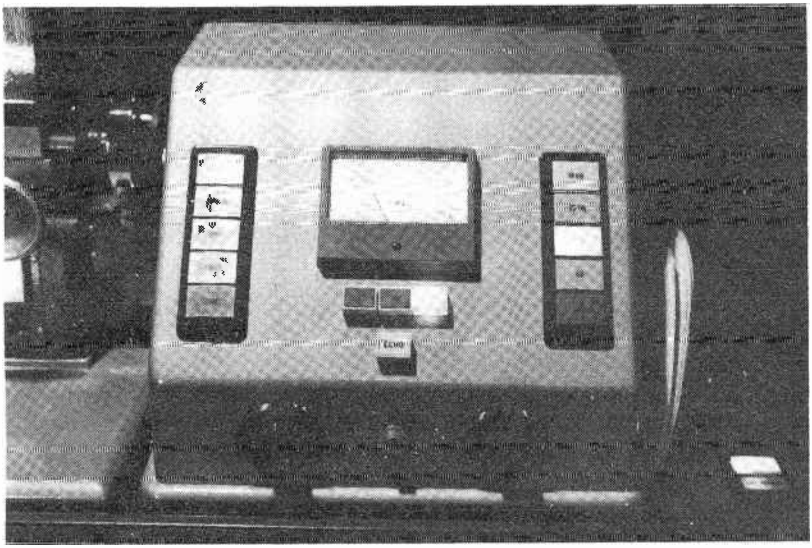
The main components of the modern disc cutting lathe are the turntable, the lathe bed and sled, the pitch/depth control computer, and the cutting head. The Neumann VMS-70 lathe is illustrated in Figs. 12-4 through 12-7. The tube entering the front of the cutter head provides helium cooling to permit extended operation at high cutting velocities.

The turntable is very heavy in order to reduce speed variations via the flywheel effect. It is driven by a special motor and linkage system to eliminate flutter and rumble from the recording. Three sets of stroboscopic rings on the outer rim calibrate the four switch-selectable speeds of the turntable: 16⅔, 22½, 33⅓, and 45 rpm. A



Courtesy The Master Cutting Room

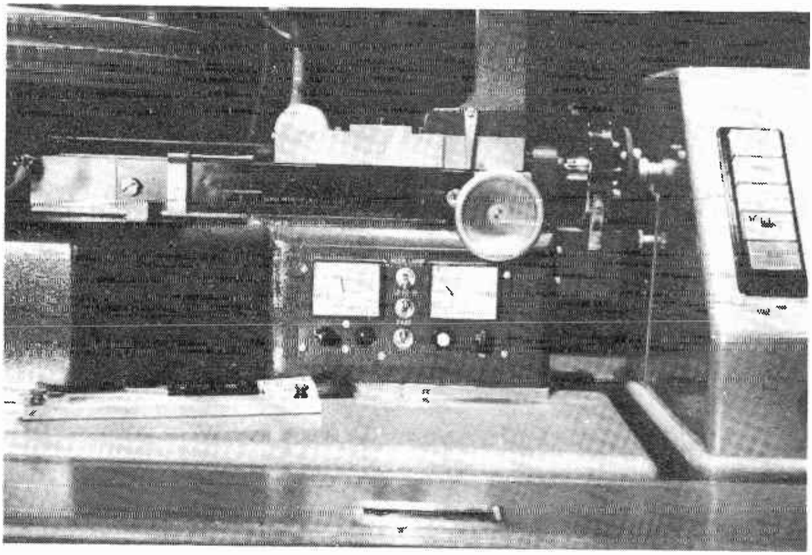
Fig. 12-4. The Neumann VMS-70 lathe front view.



Courtesy The Master Cutting Room

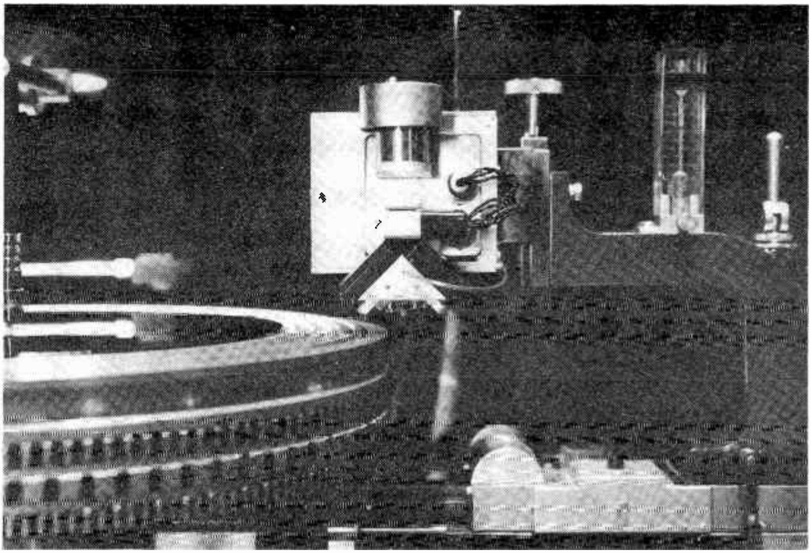
Fig. 12-5. The Neumann VMS-70 lathe pitch controller.

vacuum suction system secures the recording blanks to the turntable via holes in the turntable surface. The holes can be selectively opened or closed to provide proper suction to hold lacquer discs



Courtesy The Master Cutting Room

Fig. 12-6. The Neumann VMS-70 lathe bed and sled.



Courtesy The Master Cutting Room

Fig. 12-7. Neumann SX-68 cutter head.

from 7 to 16 inches in diameter. The suction is introduced through a flexible pipe connected to the hollow center post of the turntable.

Cutting Head

The *cutting head* translates the electrical signals applied to it into the mechanical motion of the recording *stylus*. The stylus gradually moves in a straight line toward the center hole of the disc as the turntable rotates, creating a spiral groove on the surface of the record. This motion is achieved by attaching the cutting head to a *sled*. A spiral gear known as the *lead screw* drives the sled in a straight track called the *lathe bed*. The speed of the cutting head motion toward the center of the disc determines the playing time of that side of the record.

The head speed is called the *pitch* of the recording and is measured by the number of grooves or *lines per inch* (lpi) cut into the disc. As the head speed increases, the number of lpi decreases, so the pitch and playing time also decrease. Several methods of changing pitch are possible: the lead screw can be changed for one with a finer or coarser spiral; the gears that turn the lead screw can be changed to change the speed of lead screw rotation; or the lead screw rotational speed can be varied directly by varying the speed of the motor driving it. This latter method is used in the Neumann lathe and provides continuously variable pitch.

The space between grooves is called *land* (Fig. 12-8). Unmodulated grooves are equally spaced at all points. Adding modulation to the grooves produces lateral motion proportionate to the in-phase signals contained in the two channels being cut. If the cutting pitch is too high (too many lines per inch, making the grooves very closely spaced) and high-level signals are cut, it is possible for the groove to *cutover* or break through the wall of an adjacent groove, or for the grooves to overlap which is called *twinning*. The former is likely to cause the record to skip when played. The latter causes either distortion of the signal or an echo of a signal in the adjacent groove, due to the deformation of one groove wall by the information cut in the next. Groove echo can also occur even if the walls do not touch. It is a function of groove width, pitch, and level, and it decreases as the signal frequency increases. In addition, high-frequency echos decrease in level as groove diameter decreases [3].

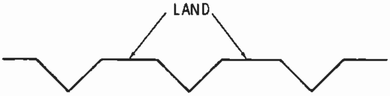


Fig. 12-8. The land portion of a recording.

These cutting problems can be eliminated by either reducing the cutting level or by cutting fewer lines per inch. A conflict arises here because in comparison with a softer record, a louder one sounds brighter, punchier, more present, and fuller. As a result, record companies and producers are concerned about the *competitive level* of their discs relative to those cut by others, so they do not want to reduce the cutting level. However, reducing the pitch shortens the playing time of the record.

The solution to these problems is to vary the pitch, cutting more lines per inch during soft passages and fewer lines per inch during loud passages. This automatic pitch control is achieved by adding an additional *preview playback head* to the tape machine feeding the lathe. This head is positioned at a distance ahead of the regular playback head on the machine. This gives the *pitch/depth control computer* in the lathe, which determines the pitch required for each portion of the program and varies the speed of the lead screw motor time to change the pitch as required. Since left channel signals are cut in the inside groove wall and therefore run no danger of cutting into an existing groove, a pitch change is not needed for loud left-channel signals until the moment they are cut. At that point, the pitch is decreased and the grooves expanded so that the following groove will be far enough away that a cutover cannot occur. Since the pitch correction is not needed until the signal is cut, the computer derives its level information from the left program channel. Right-channel signals, however, are cut in the outside wall of the groove, so loud right-channel signals require that the pitch change occur before the signal is cut, to make room for the new groove so that it does not cutover into the preceding groove. In order to provide the computer with right-channel signal level information before the signal is cut, a *preview playback head* and its associated playback electronics must be added to the tape machine. This head is positioned at a distance ahead of the program playback head of the machine (16.5 inches for a 33⅓-rpm disc and 15-ips tape). While pitch control requires preview information only for the right channel, depth control, to be described later in this chapter, requires preview information from both the left and the right channels, so a stereo preview head is used. When cutting a mono disc from a stereo tape, the sum of the left and right preview channels is used for pitch control information. The computer samples the left program and the right preview signal level information every one-quarter revolution of the turntable and adjusts the pitch to the value required by the highest of the current and the previous two level samples.

Pitch is divided into two categories; *coarse* which refers to between 96 and 150 lpi, and *microgroove* which is between 200 and 300 or more lpi. Microgroove records have less surface noise, wider frequency range, less distortion, and greater dynamic range than coarse-pitch recordings. They can also be tracked with lower stylus pressure, resulting in longer life. This lower tracking force, however, makes the stylus more likely to skate across the record if the turntable is not level. The playback stylus for a stereo microgroove record must have a tip radius of 0.7 mil or less, as compared to 2.5 mils ± 0.1 for coarse-groove records. Early 33⅓-rpm and 78-rpm records were recorded with a coarse pitch. Virtually all current records are microgroove with 265 lpi being an average pitch. At maximum pitch, the playing time of one side of a 12-inch disc with no modulation in the grooves is 45 minutes. The duration of modulated 12-inch discs cut at average levels is 23 to 26 minutes per side when they are cut with a variable-pitch lathe.

Depth Control

In mono cutting, the depth of the groove remains constant since there is no difference between what would be the left and right channels of a stereo disc, and thus there is no vertical information. The depth of the groove in a stereo disc varies with the vertical excursions of the cutting stylus and is measured from the surface of the disc to the bottom of the groove. If the depth is too great, the stylus will cut through the lacquer surface into the metal base of the recording disc, causing distortion and possible damage to the stylus. If the depth is too shallow, the cutting stylus could rise off the disc during a highly modulated passage and the groove would stop. If this disc were played, the playback stylus would skip at the point the groove stopped and would jump either to another groove or off the disc completely.

Ideally, groove depth should not go below two mils for reliable tracking on all turntables. One mil is a standard compromise for minimum depth to cut louder records. Grooves less than ¾ mil deep are considered too shallow or *light* to provide reliable tracking and are likely to cause skipping.

The problem of light grooves can be eliminated by either decreasing the separation between the channels when using a constant-depth lathe or by using a lathe with automatic depth control. This automatic control is achieved through the same pitch/depth control computer and preview playback head described earlier.

For depth control, the preview head outputs are added together out of phase to produce a signal equal to the upcoming information to be cut vertically. This signal is then applied to the depth-control amplifier. Since the stylus point forms an approximately 90° angle, deepening the groove also makes it considerably wider, so the pitch- and depth-control amplifiers are interconnected to expand the distance between the grooves when the cut is deepened. Thus, when strong out-of-phase or random phase signals are present, the depth-control amplifier receives a greater signal than usual and deepens the cut to prevent the groove from becoming too light.

RECORDING DISCS

The recording discs used on the lathe are very flat aluminum discs coated with a film of lacquer, dried under controlled temperatures, coated with a second film, and dried again. The quality of these discs, called *lacquers*, is determined by the flatness and smoothness of the aluminum base; any irregularities in its surface such as holes or bumps will cause similar defects in the lacquer coating. The disc flatness is achieved by stretching the aluminum. This can produce a *cosmetic effect* of two flashes of reflected light per revolution of the disc because the lacquer is not completely opaque. The presence of this effect does not degrade the quality or recording capability of the lacquer.

The lacquers used for *mastering* (cutting the lacquer to be sent to the plating plant) are always larger in diameter than the final record, making it easy to handle the master without damaging the grooves. A 12-inch album is cut on a 14-inch lacquer, while a 7-inch single is cut on a 10- or 12-inch lacquer. Producers often cut a *reference lacquer* to hear how the master tape will sound after being transferred to disc. Long-play references are cut on normal size discs because most turntables cannot handle a disc more than 12 inches in diameter. The lacquers used for references are noisier and of poorer quality than those used for masters.

The recorded disc consists of several distinct sections as shown in Fig. 12-9: (A) the starting spiral, (B) the lead-in grooves, (C) the program, (D) the lead-out groove, (E) the spiral out, and (F) the locked groove.

The *starting spiral* is cut at a very low pitch of 6 to 10 lpi; it serves to catch the playback stylus as the stylus is lowered onto the record and feeds the stylus to the lead-in groove. Between one and three spiral-in grooves are recommended. This standard is especially important for a record changer since its tone arm falls at a preset distance from the center pin and the stylus must land in a spiral-in groove. If an insufficient number of spiral-in grooves are cut, the tone arm could fall outside of the groove, either failing to feed into it or jumping off of the disc.

The *lead-in groove* is unmodulated and is cut at the pitch preset as the maximum for the program material. This preset pitch, together with the starting spiral which feeds the stylus into the lead-in groove, stabilizes the tone arm motion. The lead-in groove must be at least one complete revolution long. The first modulated groove is cut at a diameter no greater than $11\frac{7}{16}$ inches for a 12-inch disc and $6\frac{9}{16}$ inches for a 7-inch disc. The last modulated groove is cut at a diameter no less than $4\frac{3}{4}$ inches for a $33\frac{1}{3}$-rpm record and no less than $4\frac{1}{4}$ inches for a 45-rpm record. The limitation on outer groove diameter helps standardize the location of the starting spirals. The inner groove limitation is a combination of standardization for the lift-off function of record changers as well as a prevention against severe high-frequency losses and distortion which result from the low groove velocities at small diameters.

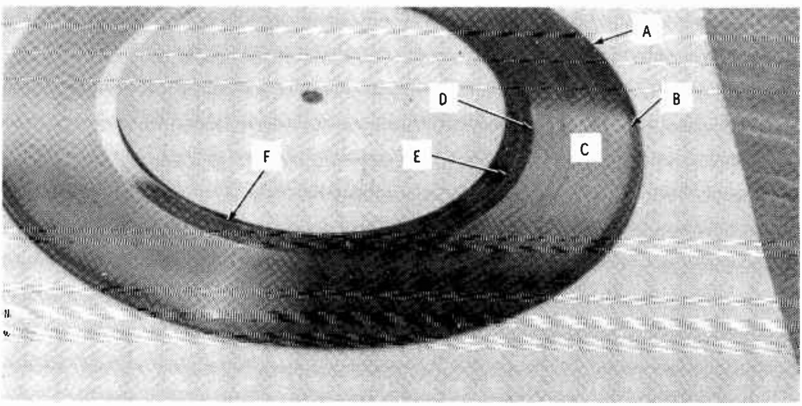


Fig. 12-9. The different grooves cut on a disc.

Inner spirals or *bands* are often cut between sections of the program to facilitate finding different selections by reading the selection number on the disc label and counting the sections of program between the spirals. The spiral grooves used for banding can be either modulated or unmodulated and are cut at the same pitch as the starting spiral.

After the program grooves are cut, at least one unmodulated *lead-out groove* is cut before the *spiral-out* action begins. The final spiral is used to start the automatic lift off and change cycle of record changers. This spiral leads into the last groove on the disc, called the *locked groove*. The locked groove leads back into itself and holds the stylus at the same groove diameter until the record changer cycle begins. If this groove were not locked, the stylus might continue toward the center of the disc, jump up onto the label and perhaps even across it, and damage the stylus. On a manual turntable, the locked groove holds the stylus until someone lifts the tone arm from the record.

The lathe can be programmed via plug-in modules to produce any desired disc parameters such as lead-in and end groove diameter; lead-in, spiral, and lead-out pitch; and cutter lift-off delay after reaching end groove diameter. These parameters are achieved through control of the lead-in screw drive motor, in conjunction with the position sensing of the sled in the lathe bed and a solenoid which lifts and lowers the cutting head onto the disc.

Stereo Cutting Head

The stereo cutting head consists of a *stylus* mechanically connected to two *drive coils* and two *feedback coils*, which are mounted in a permanent magnetic field, and a *stylus heating coil* wrapped around the tip of the stylus (Fig. 12-10). When a signal is applied to the drive coils, the alternating current flowing through them creates a changing magnetic field which alternately attracts and repels the permanent magnet. Since the position of the permanent magnet is fixed, the coil moves in proportion to the strength of the field created and the attached stylus moves with it. The drive coils are wound and mounted so that energizing either one alone causes the stylus to move in a plane 45° to the left or right of vertical, depending on which coil is energized. Feeding both coils an in-phase signal causes the stylus to move in the lateral plane, while feeding the coils out-of-phase signals causes stylus motion in the vertical plane.

The feedback coils are attached to the shank of the stylus and are, therefore, moved whenever the stylus moves. The motion of these coils in the permanent magnetic field creates a current flow in them which is an accurate representation of stylus motion. By mixing the outputs of these coils out of phase with the input signal in a technique called *negative feedback,* several advantages are achieved:

1. The need for heavy mechanical damping materials to control stylus motion is eliminated because the stylus responds more accurately to the drive signal. Without negative feedback, inertia can cause the stylus to overswing if it is not well damped. The lack of mechanical damping removes variations of damping characteristics due to age and heat from a hot stylus, which are characteristic of nonnegative feedback cutting heads. In

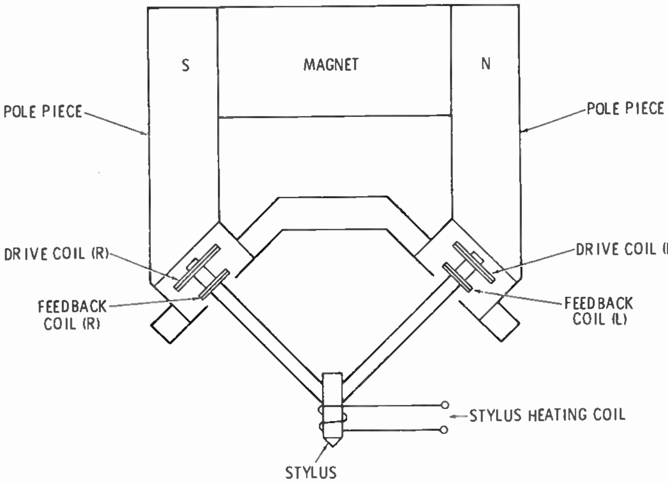


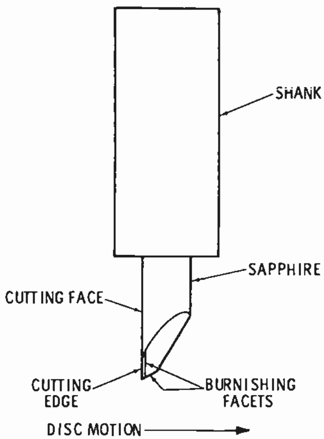
Fig. 12-10. Simplified drawing of a stereo cutter head.

   addition, negative feedback is more efficient because it controls the drive power while mechanical damping materials absorb it and dissipate it as heat.
2. Since the stylus follows the drive signal more closely, less distortion is produced.
3. The effect of irregularities in the hardness of the lacquer surface of the recording blank is reduced because the negative feedback senses them and changes the drive signal to compensate for them.
4. The signal-to-noise ratio of the disc is improved by approximately 16 dB at 10 kHz for a 6-inch groove diameter, producing an overall signal-to-noise ratio of about 70 dB when used in conjunction with a hot stylus.
5. The frequency response of the cutting head can be flattened or changed as desired by adding EQ to the negative feedback signal. The frequency range over which negative feedback can

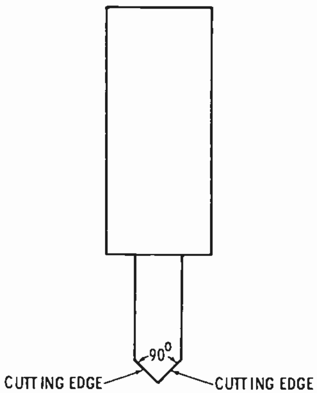
be used is limited by the mechanical resonance of the cutter head which shifts the relative phase of the input and feedback signals. This results in the feedback becoming positive at the higher frequencies. If the amount of feedback is not restricted at these frequencies, oscillations occur.

Recording Stylus

The recording stylus (Fig. 12-11) is made of sapphire (also known as corundum) because this substance is hard and it can be ground to very accurate dimensions. Although it is not as hard as



(A) Side view.



(B) View looking into the cutting face.

Fig. 12-11. A recording stylus.

diamond, the lack of grain in sapphire and its ruggedness make it superior for disc recording purposes. The sapphire is mounted in an aluminum shank so that it can be attached to the cutting head. The *cutting face* of the stylus is ground flat and oriented so that the disc rotates into it. The tip of the cutting face is ground to a 90° angle to form the *cutting edges*. The tip of the point is slightly rounded but must have a radius of less than 0.00025 inch. A *burnishing facet* ground into the stylus directly behind the cutting edge polishes the groove as it is cut, to reduce noise. The dimensions of this facet are carefully controlled because excess width erases the high frequencies of the disc.

The stylus heating coil is a small coil of wire wrapped around the stylus tip through which current is passed to heat the stylus. Cutting with a hot stylus produces several improvements in performance. Since the linear distance per revolution travelled by the stylus decreases as it moves closer to the center of the disc, while the time required to complete each revolution remains the same, the *groove velocity* decreases as the *groove diameter* decreases. As illustrated in Fig. 12-12, groove velocity equals path length divided by the time needed to travel the path. The path length between lines A and B is less at groove diameter D1 than at diameter D2. Since the disc rotates at constant angular velocity, the time it takes the stylus to travel an arc with angle θ is independent of groove diameter. As a result, the groove velocity at diameter D1 is lower than at diameter D2.

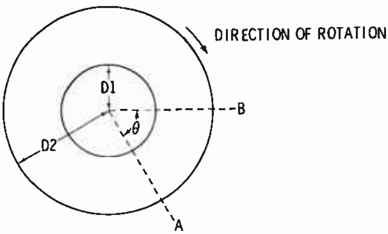


Fig. 12-12. Groove velocity decreases as groove diameter decreases.

As the groove velocity decreases, more program material must be recorded per revolution and more information must be crowded on the inner grooves of the disc. The effect of this crowding is that high-frequency response gradually decreases as the stylus moves from the outer to the inner grooves. Prior to the introduction of cutting with a hot stylus the high frequencies had to be boosted in gradually increasing amounts as the disc was cut in order to compensate for the loss. This *diameter equalization* was achieved by connecting the equalizer control to the lathe sled via pulleys, so that the EQ varied as a function of the diameter of the groove being cut. While compensating for the losses, this boosting also added to the noise and distortion content of the inner grooves.

Heating the stylus has virtually eliminated the need for diameter EQ because discs cut with a hot stylus exhibit losses of only 2 dB at 8 kHz on the inner grooves. This is a small loss as opposed to the 6- or 8-dB loss at the same frequency and groove diameter with a cold stylus, without diameter EQ. Some cutting rooms still use diameter EQ even with hot stylus cutting in order to recover the last 2 dB of high frequencies otherwise lost on the inner grooves.

The smoothing effect of the hot stylus produces a signal-to-noise ratio which improves as groove diameter decreases. The signal-to-noise ratio of a disc cut with a cold stylus, on the other hand, worsens with decreasing groove diameter. Discs cut with a hot stylus are 2 dB quieter than discs cut with a cold stylus at the outer grooves and 18 dB quieter on the inner grooves. In addition, the smoothing action virtually eliminates groove modulation noise (similar to the tape modulation noise mentioned earlier). A hot stylus cuts through the lacquer coating of the recording disc much easier than a cold stylus, facilitating the cutting of lacquer discs that have hardened due to age. A cold stylus would produce inferior grooves called *dry cuts* on a hard disc. The stylus heat also eliminates the *horns* caused by the elasticity of the lacquer (Fig. 12-13). Horns are raised edges on the sides of the groove which are easily broken and can cause the groove walls to break or crack when the finished record is removed from the press, resulting in increased surface noise. If horns occur on a disc, they are removed by polishing the molds used on the presses. Since horns limit the level that can be cut on a disc, higher levels can be cut with a hot stylus.

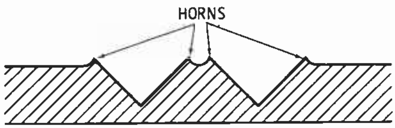


Fig. 12-13. Horns resulting from cutting with a cold stylus.

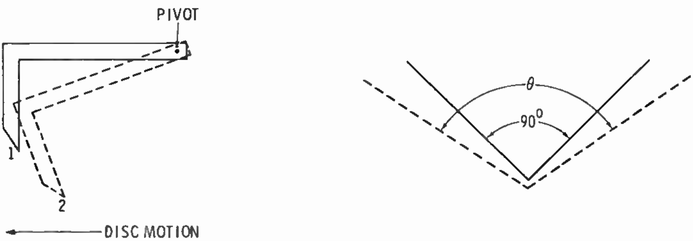
Stylus heat must be carefully set because too much heat can cause horns to form and the wrong amount can reduce the signal-to-noise ratio. Heater current is set by listening to the cutting head feedback outputs and adjusting the current for the minimum sputtering noise. Since groove velocity is constantly decreasing as the disc is cut, the heat applied per unit area progressively increases. As a result, lowest noise occurs only at the groove diameter at which the current was set. Ideally, heater current should decrease as the disc is cut. The signal-to-noise ratio of a lacquer master made with a hot stylus and a negative feedback cutting head is about 70 dB, but the plating process used in the manufacturing of the finished records degrades this figure by adding ticks and pops to the signal.

The material removed from the lacquer disc by the stylus is called the *chip*. A tube aimed at the stylus and connected to the lathe vacuum system removes the chip as the groove is cut; this prevents the chip from blocking the path of the stylus. The suction is always started first, and stylus heat is then applied as the cutting head is lowered onto the disc. If this is not done, the chip, which becomes very limp when stylus heat is used, may collect underneath the stylus before the suction can begin to act.

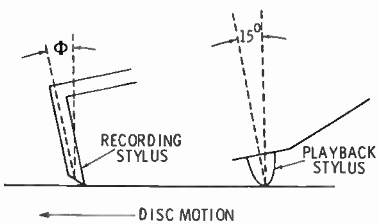
DISTORTION

Since the playback stylus *tracking angle* has been standardized at 15° from vertical with the stylus pointing into the disc rotation, the cutting head must be angled so that it produces a groove that is oriented in the same way. This is complicated by the fact that the lacquer springs back somewhat after being cut. As a result, cutting heads are operated at a greater angle of about 18° in order to produce a 15° groove. This standardization of reproducer tracking angles is necessary because the nature of the recording stylus assembly causes vertical information to be cut in an arc, rather than strictly vertical. As the stylus moves in this arc, the groove angle becomes wider than 90°. If the playback stylus tracking angle is not set to compensate for this increased groove angle, second-harmonic distortion would be introduced into the program.

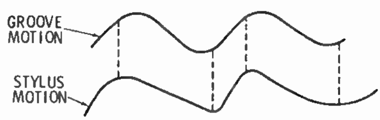
As shown in Fig. 12-14A, with no vertical information, the recording stylus is in position 1, and a 90° groove angle is cut. When the



(A) Cutting at increased depth causes groove angle greater than 90°.



(B) Cutting angle is greater than playback angle.



(C) Vertical tracking error.

Fig. 12-14. Tracing angles for cutting and playback styli.

cutter receives vertical information, the stylus cuts deeper into the disc. It does not move straight down, rather the support arm pivots as shown by stylus position 2. This produces a groove angle θ, greater than 90°. Fig. 12-14B shows that since the playback stylus vertical tracking angle is standardized at 15°, the recording stylus cutting angle is set to ϕ (slightly greater than 15° to compensate for lacquer spring back) so that the groove walls will present a 90° angle to the playback stylus. Fig. 12-14C shows the vertical tracking angle error with the stylus perpendicular to the disc. Peaks of the stylus motion occur before the peaks of groove motion, and dips in the stylus motion occur after dips in the groove.

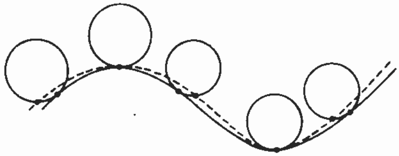
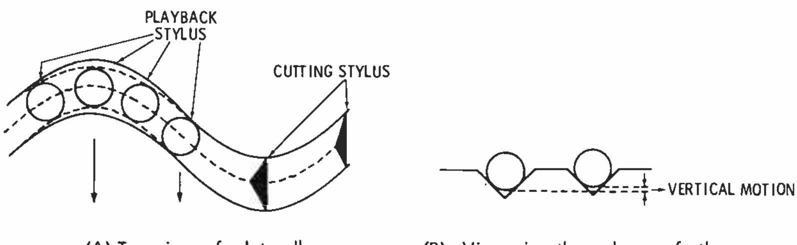


Fig. 12-15. Tracing distortion.

Other potential sources of signal degradation such as tracing distortion, the pinch effect, reaching the groove excursion limit, and mistracking can also occur during playback of the finished record. *Tracing distortion* results from the difference in the shape of the recording and playback styli. The recording stylus cutting edge comes to a sharp point while the playback stylus has a rounded point. Because of this, the playback stylus point of contact with the groove varies depending on the instantaneous amplitude of the groove modulation, and the path traced in playback is not the same as that recorded (Fig. 12-15). The point of groove wall contact for the play-stylus wanders as the modulation is traced, producing an output corresponding to the dotted path rather than the modulation. As signal level increases the distortion increases and as the wavelength of the recorded signal decreases and approaches the dimensions of the stylus tip. Thus, tracing distortion increases as the signal frequency increases and as groove diameter decreases, because both of these factors cause the wavelength of the signal to decrease (wavlength equals groove velocity divided by signal frequency, and groove velocity decreases as groove diameter decreases).

Tracing distortion refers to the inability of the playback stylus to follow groove modulations in the vertical plane. A similar problem in the lateral plane is called the *pinch effect*. Due to the triangular shape of the cutting stylus, the width of the groove measured perpendicular to the line cut by the stylus tip does not remain constant (Fig. 12-16A). The playback stylus is pinched where the groove narrows, and therefore it rides higher in the groove. As the



(A) Top view of a laterally modulated groove.

(B) View in the plane of the groove, showing the playback stylus rising in the groove where the groove narrows.

Fig. 12-16. The pinch effect.

groove widens, the stylus rides lower in the groove (Fig. 12-16B). This motion adds a vertical component to the signal output that was not present in the input signal. The pinch effect is greater for high-level and high-frequency signals, for these cause abrupt changes in groove direction and therefore cause the width of the groove to become narrower.

The *groove excursion limit* is reached when the radius of curvature of the groove modulation is equal to the tip radius of the playback stylus, preventing it from following all of the modulation. As shown in Fig. 12-17, the radius of curvature of the groove modulation is equal to the stylus tip radius, producing 50% second-harmonic distortion. Since the radius of curvature decreases both with increase in level as well as with increase in frequency, a limit exists on the level of high-frequency information that can be accurately reproduced.

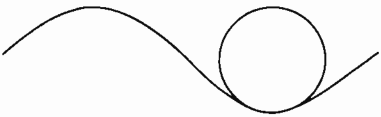


Fig. 12-17. Side view of a groove illustrating the groove excursion limit.

All three of these types of distortion would be reduced if the playback stylus point was very small. Distortion would be eliminated altogether if the playback stylus was shaped like a recording stylus. A stylus with a small tip radius, however, would ride on the bottom of the groove, producing noise and other types of distortion. However, a cutter-shaped stylus would tend to cut into the disc and erase high-frequency information.

A compromise solution is the use of a *biradial elliptical* stylus rather than the standard *conical* stylus. The conical stylus has a round tip with a radius of 0.6 mil. The elliptical stylus has an edge radius of 0.2 mil and a radius at right angles to the groove of 0.7 mil. The small edge radius follows groove excursions more accurately than the conical stylus, while the larger radius prevents the stylus from hitting the bottom of the groove. Distortion is reduced somewhat, at the cost of increased wear on the disc.

The *Shibata stylus*, designed for use in playing discrete quadraphonic discs, reduces disc wear by contacting the groove walls in a line rather than at only two points as do the conical and elliptical styli. The edge of the Shibata stylus is narrower than the elliptical stylus and can therefore trace even higher recorded velocities. Fig. 12-18 shows the cross-sectional areas of the conical, eliptical, and Shibata styli.

A more effective means of reducing tracing, pinch effect, and groove excursion limit distortion is through the use of a *tracing*

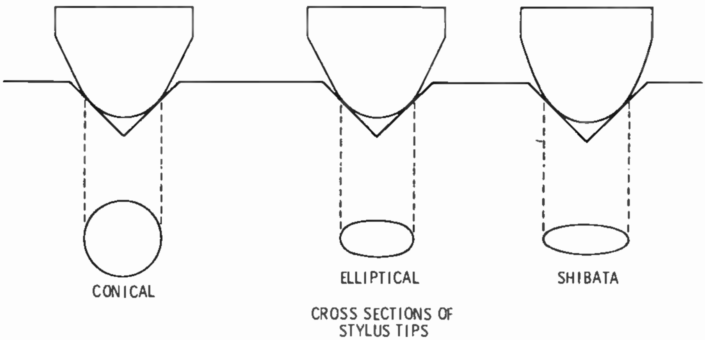
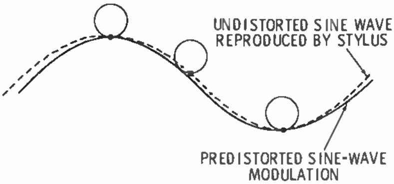


Fig. 12-18. Groove contact for the conical, elliptical, and Shibata styli.

simulator such as Neumann's Model TS-66. All three of the discussed types of distortion are predominantly of the second-harmonic variety. Their effect can be reduced by *predistorting* the signal to be cut on the disc in an inverse proportion to the distortion generated by the playback stylus (Fig. 12-19). The tracing simulator generates a voltage corresponding to the second harmonic of the program signal and adds it to the program out of phase with the distortion created in playback. This second-harmonic component is cancelled by the tracing, pinch effect, and groove excursion limit distortion generated during playback, resulting in a disc with substantially less distortion and greater high-frequency level capability.

Since the amount of distortion produced in playback is a function of groove velocity, which is in turn determined by the groove diameter for any particular turntable speed, a series of microswitches are connected in the lathe bed to sense the position of the cutter head. The microswitches predistort the drive signal the proper amount for that groove diameter.

Fig. 12-19. Reducing playback distortion by predistorting the groove modulation.



The tracing simulator can only optimize distortion for one tip radius at a time; it is usually adjusted for 0.6 mil which is the international standard for the radius of a conical stylus. The elliptical and Shibata styli with their narrow side radii would not completely cancel the second-harmonic distortion added to the signal by the tracing simulator. Therefore, these styli would not receive the full benefits of the distortion reduction.

Mistracking results from the loss of stylus contact with the groove walls during playback, due to recorded velocities in excess of that which the stylus can follow. The *recorded velocity* is measured in centimeters per second (cm/sec) and is determined by computing the distance that the playback stylus must move, laterally or vertically, per second from the unmodulated groove position to accurately track the groove modulation. Since recorded velocity is constantly changing, the value used is the instantaneous peak velocity.

Mistracking often sounds like a buzz or crackling on heavily modulated passages, or sibilance on vocals. Mistracking can cause instruments which should have sharp, clear sounds, such as bells, to produce a dull thud at the beginning of each note. Increasing the tracking force (pressure of the playback stylus on the groove) reduces mistracking. The maximum recorded velocity, listed by frequency, which can be accurately tracked at a certain tracking force is a function of its design and specifies the *trackability* of a stylus/cartridge assembly. The better the trackability, the lower the tracking force needed to prevent mistracking.

The pressure of the stylus results in some indentation of the groove modulations. As the radius of curvature of the recorded signal increases, the stylus indents the record surface more, resulting in less output. This *playback loss* becomes more severe as frequency rises and as groove diameter decreases, because both of these increase the radius of curvature. Typical finished records have playback losses of 3 to 4 dB at 15 kHz at minimum diameter, relative to the master tape. Stiffer record compounds are indented less and therefore have lower losses.

Most phono cartridges are designed using either magnetic or piezoelectric principles. The *magnetic cartridge* converts stylus motion into electrical current flow through the use of permanent magnets and coils. Designs are available using fixed magnets with moving coils, moving magnets with fixed coils, and fixed magnets with fixed coils in which the stylus motion varies the magnetic coupling to the coils. In all cases, the coils are angled 90° apart from each other and are oriented so that signals in the outer wall of the groove produce current flow in the right-channel coil, and signals on the inner wall produce current flow in the left-channel coil. The coils are phased so that lateral stylus motion produces an in-phase output from both coils, while vertical stylus motion produces an output that is out of phase.

The *piezoelectric* or *crystal* phono cartridge generates an output voltage proportional to the varying pressure applied to a crystal by the stylus motion. Crystal phono cartridges are much less expensive to manufacture than magnetic ones. They also produce higher output levels so that the extra stage of preamplification required by magnetic cartridges is not needed. The magnetic cartridge, however, is quieter and operates at lower tracking forces, creating less wear on the grooves. Crystal cartridges are mainly used in low-priced record players.

EQUALIZATION

Disc recording uses pre- and post-equalization just as tape recording does, but the equalization is set to the Record Industry Association of America (RIAA) standard rather than the NAB (Fig. 12-20). This standard results in the level of bass frequencies being decreased and the level of high frequencies being boosted during cutting, with the signals restored to flat by reciprocal equalization on playback.

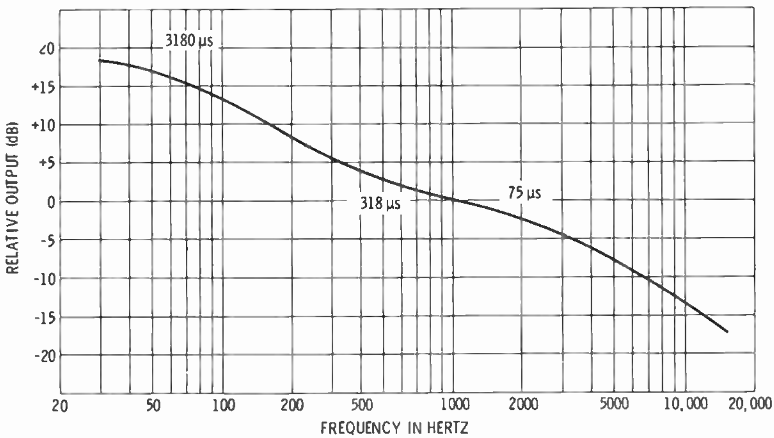
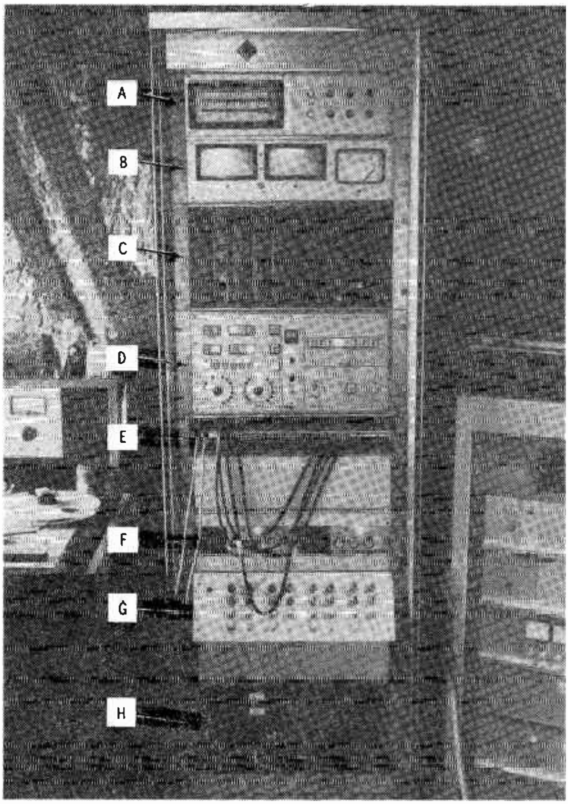


Fig. 12-20. The RIAA disc playback equalization curve.

Two improvements in signal-to-noise ratio result from this. First, since most of the energy of music is in the low frequencies, the large groove excursions accompanying them would limit the maximum level cut on the disc, as well as limiting disc playing time. Rolling off the lows and restoring them in playback permits the upper portion of the frequency spectrum to be cut at a higher level. Second, since relatively little music energy is contained in the higher frequencies, these frequencies can be boosted without overmodulating the grooves. Reducing the high-frequency response to normal on playback also reduces the audibility of disc surface noise. The RIAA high-frequency EQ improves disc signal-to-noise ratio by about 8 dB.

THE MASTERING CONSOLE

The disc-mastering console controls and monitors the signal cut on the disc and enables certain changes to be made. The Neumann SP-172 Program Controller is used as an illustration. Fig. 12-21 illustrates its equipment layout as follows: (A) Light-beam peak program meter. (B) Stereo VU meters (left and center) and correlation meter (right). (C) Left to right: channel gain controls; four Neumann equalizers for the program and preview channels; two high- and low-pass filters (each simultaneously controls the program and preview signals for its channel); the phase oscilloscope; the automatic banding unit; and the stereo master fader. (D) Left to right; the cutting controls; the console oscillator; the metering and monitoring controls. (E) The patch bay. (F) Compressor/limiters for the program channels (stereo interconnected). (G) Four addi-



Courtesy The Master Cutting Room

Fig. 12-21. The Neumann SP-172 Program Controller.

tional equalization channels to augment those in (C). (H) A Dolby A301 unit. Items (F), (G), and (H) are not part of the SP-172 console but are interconnected with it in this particular installation.

The console provides individual channel gain controls with a range of ±10 dB in 0.5-dB steps to adjust the balance between channels as well as the relative levels of the different selections on a disc. A stereo master fader is provided to fade out any sounds not faded early enough on the master tape. A stepped *relative disc level* control adjusts the level of the signals cut on the disc relative to standard recording levels without affecting the console meter readings. Standard recording level is measured at 1 kHz, and for mono is a peak stylus velocity of 7 cm/sec. For stereo, standard level is 3 dB lower per channel, at 5 cm/sec. This arises from the fact that the mono signal is cut laterally (Fig. 12-22) while the channels of a

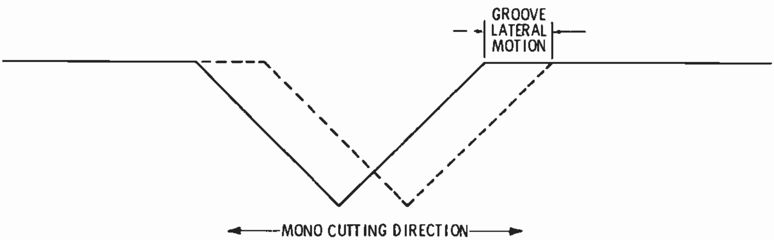


Fig. 12-22. A mono groove producing lateral motion.

stereo groove are cut at 45° to the vertical. Since there is a 45° difference in the direction of cutting motion between a mono and a stereo signal, the lateral motion of a stereo groove equals that of a mono groove when stereo motion in the plane 45° from vertical is 0.707 times the mono motion (Fig. 12-23). This results in 0.707 times the mono output for each channel of the stereo groove. In decibels, this is 3-dB less output per channel relative to the output of a mono groove.

Thus the stereo and mono standard levels produce the same lateral groove motion. The *relative disc level* control permits discs to be cut *hot* or *cold* with respect to standard level while still giving an on-scale reading on the meters. For example, if the record is to be cut 2 dB higher than standard level, this control is set for +2 and the gain controls are adjusted so that the VU meters read 0 VU on the loudest sections of the program. These loud sections will then be cut 2 dB above standard level.

The *preview offset* control is a stepped control which changes the gain only in the preview channels in order to adjust the sensitivity of the automatic pitch/depth control computer for optimum cutting of the disc. This control is usually set the same as the relative disc level control so that the computer will compensate pitch and depth for cutting at other than standard level. It can also be set higher than the relative disc level control to cause extra expansion of the grooves on highly modulated passages and to reduce the chance of a highly modulated groove affecting a neighboring groove.

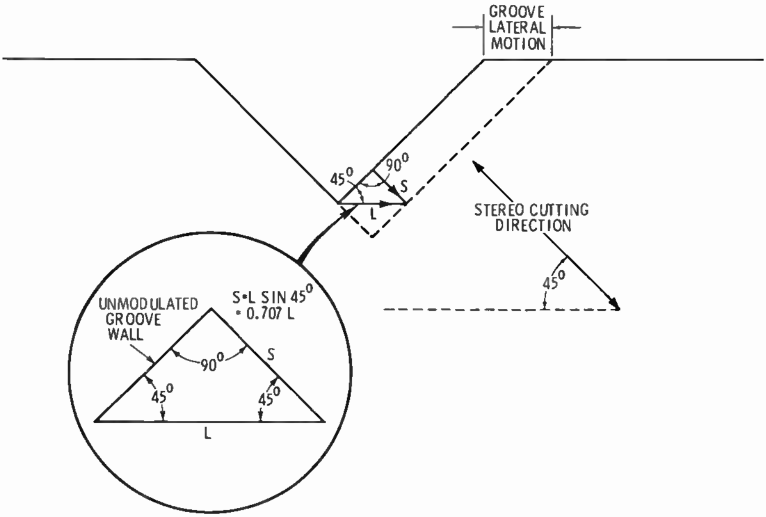


Fig. 12-23. Stereo groove motion with one channel modulated, producing the same amount of lateral motion as the mono groove in Fig. 12-22.

The console also provides equalizers to adjust the tonal balance of the program. Equalization can be used at this point to compensate for differences in the balance of selections mixed in different control rooms or mixed over different monitor speakers, or to compensate for deficiencies in the mixes themselves. In addition, EQ can be used to make the overall level of the program seem louder than it really is. Some record companies make a habit of boosting the midrange at about 5 kHz to achieve this effect. Four sets of equalizers are provided so that the same EQ that is used on the program channels can be added to the preview channels which feed the lathe computer. This is necessary to preserve the accuracy of the automatic pitch- and depth-control function.

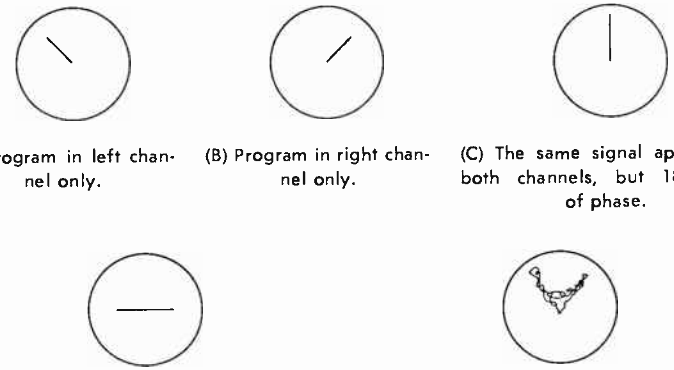
The Neumann equalizers used in the SP-172 console provide EQ of ±15 dB at 60 or 100 Hz and at 10 kHz in 11 steps; they provide EQ of ±8 dB at 0.7, 1.0, 1.4, 2.0, 2.8, 4.0, and 5.6 kHz in 9 steps. The console also provides 12 dB/octave high- and low-pass filters with switchable cutoff frequencies of 8, 10, 12, or 14 kHz and 60, 125, 250, or 500 Hz, respectively.

External compressor/limiters can be connected in each channel to either increase the average level of the program or to prevent overmodulation. The compression mode reduces the dynamic range of the signal. The overall effect is that the soft passages of the signal seem louder than they would otherwise be, while the loud passages do not overmodulate the grooves. The limiter mode merely prevents overmodulation without reducing the apparent dynamic range. The compressor/limiters for each channel are interconnected for stereo use to prevent the center shifting that would occur if gain reduction was not the same in both channels. Ideally, compressor/limiters should also be provided for the preview channels, but this is not always done.

A high-frequency limiter is provided to reduce the very high stylus velocities created by high-level, high-frequency signals. High velocities are very difficult for the playback stylus to track without distortion or skipping. They can also cause the cutter head to overheat or blow its protective fuses if they are of long duration. This limiter causes gain reduction in proportion to the amount of high-frequency energy present, rather than in proportion to the overall level as with the compressor/limiter. It can also be used to reduce sibilance on the vocals in a mix, but its threshold must be set carefully so that cymbal crashes or other high-frequency information do not trigger it accidently. The unit can be bypassed if no high-frequency limiting is desired.

Four types of visual monitoring devices are provided on the console. Two VU meters display average program levels for each channel corresponding to the loudness of the signal. Two light-beam display peak meters indicate instantaneous program levels corresponding to maximum cutting stylus excursions. The third display is an oscilloscope in which the trace is driven horizontally by one channel and vertically by the other. The face of the scope is rotated 45° so that the horizontal and vertical axes are at 45° from true vertical, thus representing the shape of the record groove.

The oscilloscope displays the instantaneous relative phase of the two channels (Fig. 12-24). When no signal is present, the trace is a stationary dot at center screen. A left-channel signal produces a trace 45° to the left, while a right-channel signal produces a trace 45° to the right. When a stereo program is monitored, the trace produces swirls and other movements between these two positions. Complete separation between the channels, such as would result from out-of-phase signals, produces a vertical trace. As separation decreases in the two channels (i.e., as more information appears that is common to, and in phase), the trace broadens. A mono signal produces a horizontal line. The movement of the trace thus represents the motion of the cutting stylus. The horizontal and vertical scope amplifiers are designed to compress the signals so that a full trace appears regardless of level differences between the channels. The mastering engineer watches the display and notes any sections of the program which have extreme separation. He must be sure that these vertical signals will not cause the groove to become too light, or skipping may occur on playback.



(A) Program in left channel only.

(B) Program in right channel only.

(C) The same signal appears in both channels, but 180° out of phase.

(D) A mono signal.

(E) A stereo signal.

Fig. 12-24. Phase oscilloscope displays.

Solutions to the problem of too much separation are to reduce the cutting level, to increase cutting depth, or to use an *elliptical equalizer* which reduces separation below its turnover frequency without affecting the high frequencies which provide most of the directional information. Decreasing the amount of vertical information also improves the mono/stereo compatibility of the disc. The Neumann elliptical equalizer has switchable turnover frequencies of 150 and 300 Hz.

The fourth indicator is the *correlation meter*. This is a center-zero scale meter similar in function to the phase oscilloscope except that it provides average rather than instantaneous phase information. Signal waveforms in the two channels which are of the same polarity at a given point in time are considered correlated, while those which are of opposite polarities are considered uncorrelated.

A positive reading on the meter indicates that the program has a predominance of correlated (in-phase) information and will produce more modulation in the lateral plane than in the vertical. A negative reading indicates that the program has a predominance of uncorrelated (out-of-phase or difference) information and will produce more vertical modulation than lateral. A zero reading indicates either equal amounts of correlated and uncorrelated information, or the lack of signal in one or both channels.

Identical signals in both channels (a mono program) produce a full-scale positive reading, while identical signals that are 180° out of phase produce a full-scale negative reading. A stereo signal with good separation causes a fluctuating reading which remains on the positive side of the scale. A negative reading indicates the possibility of shallow grooves and signal cancellations if the program is played in mono.

A *phase reversal* switch in the console compensates for the existence of a 180° out-of-phase condition between the channels by reversing the phase of one channel. An out-of-phase condition can arise due to miswiring of the console or the tape machine used to produce or play the master tape.

Other controls reverse the left and right channels and insert EQ and noise reduction devices into the cutting chain. Another control provides a 14-dB gain increase for the meters, which is for use only in playback of test records for phono cartridge and cutter head frequency-response calibration. (Test records are cut 14 dB below standard level to prevent overheating of the cutting head coils by the sustained tones and to prevent excessive groove velocities at high frequencies.)

In addition, the console has a *mono mix* switch which combines the left and right signals in a one-to-one proportion to form a mono signal. This function is used when a mono single is to be cut and the producer does not feel it is necessary to make a separate mono mix of the song. Often the stereo mix used for the album becomes the single mix through the use of this function.

With the 45/45 cutting system, mono is produced without having to change the cutting head. This is accomplished by feeding the mono signal to both channels of the cutting head. When a separate mono mix is provided for cutting a single, the mastering engineer presses the *mono button* on the console. This button connects the left channel of the tape recorder to both channels of the mastering console.

There are several reasons for this: If the mono mix is recorded full track, it will be reproduced by the top gap on the half-track stereo playback head of the tape machine. If the mono mix is recorded in the half-track mono configuration, it is equivalent to the left channel of a half-track stereo recording, and it will also be reproduced properly. If the mono mix is recorded on both channels in the half-track stereo configuration, only the left channel is reproduced. This last case is important, for if the azimuth of the record head used to make the master tape is not set for optimum phase response, reproducing and mixing the two channels (either in the console or on a mono record player) could cause severe frequency cancellation. Playing back only the left channel of a mono tape also eliminates the necessity of changing the headblock on the tape machine when both stereo and mono discs are being cut.

One of the monitoring capabilities of the mastering console is the ability to switch between tape machine program and preview outputs, console output, several playback turntables, and the cutting head feedback signal. The selected signal is displayed on the console meters as well as heard through the speakers. The cutting head feedback signal enables the engineer to hear exactly what the stylus is cutting and therefore makes audible any imperfections or holes in the lacquer coating which might impair cutting.

Additional features of the Neumann console include a patch bay, a five-frequency test oscillator, flashing illuminated push buttons (which signify lathe misfunctions, thus preventing the ruin of any lacquers), and remote controls for the lathe and the tape machines.

The actual mastering of a disc involves several steps. After the tape deck and Dolby units (if used) have been aligned to the level-set, the tones previously recorded at the head of the tape (or the tones recorded on an alignment tape) the master tape is played and monitored on the speakers and meters. Any desired changes in tonal balance are made at this point, using the equalizers. If compression or limiting is desired, it is also done at this point.

The mastering engineer adjusts the individual channel gain controls so that the VU meters read 0 VU on the loudest portions of the program. He also watches the peak meters for excessive peaks and watches the oscilloscope and correlation meter for random or out-of-phase conditions.

If high peak signal amplitudes or phase problems are noted, the engineer can make a *test cut* of that portion of the program. He examines the groove under a 156-power microscope to be sure that the peaks do not cause cutovers and that the random or out-of-phase signals do not cause the grooves to become too light. The test cut is also played to make sure that the signal neither distorts on playback, nor causes the playback stylus to skip. Overmodulation can be eliminated by reducing the peaks with a limiter or by lowering the channel gain (or relative disc level). Inserting the elliptical equalizer or decreasing the gain (or relative disc level) will prevent the groove from becoming too light, as will advancing the preview offset; this will cause the depth of the cut to increase when random or out-of-phase signals are cut. The settings used are written on a card and filed in case additional masters or references need to be cut at a later date.

When an lp is being cut, the setup is slightly more involved. The EQ, compression, and level settings must be determined for each selection individually for optimal results. A decision must be made at this point as to whether each selection on the lp should be treated separately or as part of the whole. If separately, then each selection should be cut at the highest possible level, without introducing distortion or skipping, to produce the best signal-to-noise ratio and loudest playback level. If the selections are to be treated as part of a whole, the transitions between volume levels of adjacent selections should be smooth so that the lp flows easily from one to the next. The problem of transitions is the result of the different nature of each selection. For example, a voice accompanied by a single soft guitar will sound much louder than the same voice accompanied by a loud guitar, electric bass, piano, and drums, even though both mixes read 0 VU on the meters. In the latter case, the energy which produces a 0-VU reading is being created by five sources rather than two, and the proportion of energy provided by the voice is less than in the first case. If both selections are cut at the highest level possible, the voice will appear loud and large in size on the first selection and relatively small on the second one. The apparent change in size of the voice can be reduced by cutting the the first selection at a lower, softer level or by compressing the second one so that it can be cut at a higher, louder level. By listening to the ending of one song and the beginning of the next, any necessary changes in levels can be made.

MASTERING

The mastering engineer sets a basic pitch on the lathe. This pitch is determined by the duration of the program to be cut on that side of the disc and whether the music is loud continuously or has quiet sections as well. A lacquer is placed on the lathe, and compressed air is used to blow any accumulated dust off the lacquer surface. Chip suction is started, and a test cut is made on the outside of the disc to check groove depth and stylus heat. The start button is pressed, the lathe moves into the starting diameter, the cutting head is lowered onto the disc, the starting spiral and lead in are cut as preset, and the tape machine is started automatically. As the side is cut, the engineer changes the console settings as previously determined. A photocell mounted on the tape deck senses the white leader tape between selections on the master tape and signals the lathe to automatically expand the grooves to produce bands. After the last selection on the side, the lathe cuts the lead-out groove, spirals out to a preset diameter, cuts the locked groove, and lifts the cutter head off the lacquer.

The master lacquer is never played because the pressure of the playback stylus would damage the recorded sound track. Reference lacquers, also called *reference acetates* or simply *acetates*, are cut to hear how the master lacquer sounds. Thus, any damage to the sound track is confined to the reference acetate, preventing any damage to the master lacquer from being transmitted to the finished record. The damage consists of high-frequency losses and increased noise and is aggravated by each successive playing. The damage done per play is proportional to the stylus pressure used. At a tracking force of one to two grams, most references remain usable for eight or ten plays.

Since the producer wants to hear any EQ, level, or dynamic range changes introduced in the mastering process, he orders a reference disc before ordering the master lacquers. After listening to the reference several times on a system with which he is familiar, he can either approve it and order the masters, or order a new reference with different EQ, etc.

The producer does not have to be present in the cutting room when the references are cut, although some do prefer to attend so that they can hear the changes produced by the EQ and decide how they want the reference cut. Other producers either ship or bring their tape to the cutting room with instructions on how they want it cut and order a reference disc. If the first reference is not acceptable, they order a second one specifying new EQ or level settings. This procedure is repeated until an acceptable reference is produced. The approved control settings are then used for the production of all of the master lacquers.

The advantage of giving cutting instructions beforehand is that the producer does not have to make changes with an unfamiliar speaker system. Occasionally, a producer may not realize that there is a difference in sound quality between home speaker systems and cutting room speaker systems, due to the superior equipment and conditions available in the recording studio. In this case, the producer will be unaware that frequency balance heard in the cutting room speakers must be deliberately distorted in order to produce the desired sound effect on home speaker systems. His attempts to coordinate the two speaker systems will be continually frustrated, and he may be wise to forego attendance at cutting room sessions.

After the control settings are approved, the master lacquer can be cut at any time. Since a master lacquer must be plated within three days after it has been cut in order to prevent the groove walls from drying out, it is not cut until the record company tells the producer the disc is ready to be pressed.

The record company assigns each side of the disc a *master* or *matrix number* which the cutting room engineer scribes between the grooves of the ending spiral on the lacquer (Fig. 12-25). This number identifies the lacquer and any metal parts made from it, and eliminates the need to play the record to identify it. If a disc is remastered for any reason, some record companies retain the same

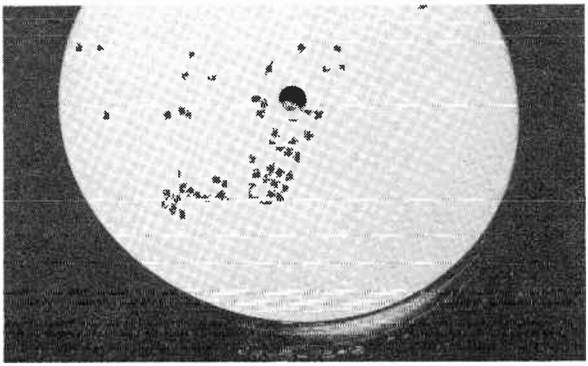


Fig. 12-25. The master number on a disc.

master numbers, while others add a suffix to the new master, such as "RE 1," to differentiate it from the previous one.

The engineer inspects the lacquers for any cutting defects and then ships them to the plating and pressing plants selected by the record company. Most record companies order three sets of masters so that the record can be pressed in three plants: one on the east coast, one in the midwest, and one on the west coast. Because each plant services the markets closest to it, the cost of shipping the finished record can be reduced.

While references are often recorded on both sides of a lacquer, masters are recorded only on one side of the disc. Cutting each side of a record on a separate lacquer allows the lacquers to be shipped with their blank sides touching and makes the production of molds for the record easier.

When the master arrives at the plating plant, it is washed to remove any dust particles and is then electroplated with nickel. When the electroplating is complete, the nickel plate is pulled away from the lacquer. This damages the master, so it can only be plated once. If something goes wrong at this point, the plating plant must order a new master from the cutting room.

The nickel plate pulled off of the master is called the *matrix* and is a negative image of the master lacquer (Fig. 12-26). The bottom of the groove is the highest part of the matrix, and the land is the lowest part. The excess nickel at the edges of the matrix is ground

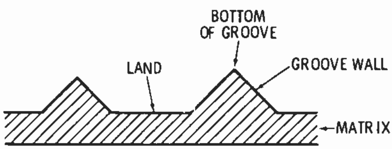


Fig. 12-26. The matrix is a negative image of the master lacquer.

down, and the matrix is attached to a metal or wood-backed plate to make it easier to handle. Since the groove walls containing the signal are now raised above the surface of the disc, they are very vulnerable and care must be taken to prevent damage to them.

This negative image is electroplated to produce a nickel positive image called a *mother*. Because the nickel is stronger than the lacquer disc, several mothers can be made from one matrix. Since the mother is a positive image, it can be played to test it for noise, skips, and other defects. If it is acceptable, the mother is electroplated several times, producing the *stampers* which are negative images of the disc and are used to press the records. The stampers are sometimes chrome plated to seal the nickel surface and make them last longer; however, the chrome tends to plate more on the groove edges than on the walls and increases the number of clicks and pops on the record. The plants that chrome their stampers feel it is better to have a few more clicks and pops than to have quieter records which deteriorate in signal quality as more records are pressed and the stampers wear out.

The plating process described above is called the *three-step process* to differentiate it from the *single-step process* which uses the matrix as a stamper. The single-step process is used when less than two-hundred records are to be pressed. The matrix, mothers, and stampers are called *parts*, and the plating process is referred to as *making parts.*

STAMPING

The stampers for the two sides of the record are mounted on the top and bottom plates of a hydraulic press (Fig. 12-27). A lump of the *vinylite* record compound, called a *biscuit* due to its shape, is placed in the press, sandwiched between the labels for the two sides. The press is closed and heated by steam to make the vinylite flow around the raised grooves of the stampers. Since the pressed record is too soft to be handled when hot, cold water is circulated through the press to cool it before pressure is released. When the press opens, the operator pulls the record off the mold. The pressing process causes excess compound to flow to the outer edges of the disc, so the disc is oversize. This excess, called *flash*, is trimmed off after the disc is removed from the press. The edge of the disc is then buffed smooth. The finished 12-inch record is actually $11\frac{7}{8} \pm \frac{1}{32}$ inches, while a finished 7-inch record is actually $6\frac{7}{8} \pm \frac{1}{32}$ inches.

Record molds are designed to make the thickness of the finished disc greater in the label area and on the edge than in the groove area (Fig. 12-28). This is done so that when the discs are stacked after being pressed, their groove areas do not rest on each other.

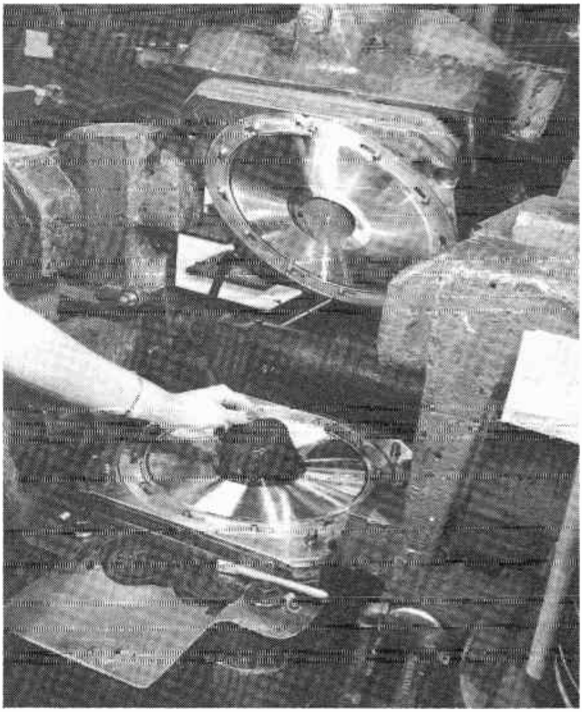


Fig. 12-27. A record press.

The raised outer edge, called the *groove guard*, must be the same thickness as the label area, or *dish warp* will occur when the still soft discs are stacked. A disc with dish warp is saucer shaped rather than flat.

Other problems can occur at the pressing stage. *Pinch warp* can occur if the record cools too much before being removed from the mold, making it stick to the stamper. The operator can pinch the disc or even leave finger prints on it in the process of pulling it off, causing warps in the disc.

Nonfill occurs when the vinylite does not flow around all of the grooves of the stamper. It is characterized by a swishing type of *once-around* noise, i.e., it occurs once each revolution of the disc. If

Fig. 12-28. The profile of an lp.

the disc is held up to a reflecting light, the noise area becomes visible as a dull greyish area on the otherwise shiny surface. Nonfill can occur when insufficient steam is fed to the presses to heat them and so the vinylite does not become hot enough to flow properly. Or, it can happen because the presses are running too fast for the vinylite to flow evenly through the entire mold.

The presses take more time to heat up when they are first started, so the first pressings are more likely to exhibit nonfill than the later ones. Occasionally, all of the presses in a plant demand maximum steam at the same time. To satisfy this demand, a source of steam many times greater than that normally needed would be necessary. Since the cost of providing this extra steam is prohibitive, a few cases of nonfill are unavoidable and have to be found and rejected by quality control.

The center holes of the discs are formed by the presses .Their location is determined by the mounting of the stampers on the press plates. The standard for 33⅓-rpm records is a 0.286, +0.001, −0.002-inch diameter center hole, concentric with the recorded groove within 0.005 inch. For 45 rpm, the center hole is 1.504 ± 0.002 inches, concentric within 0.005 inch.

Since vinylite is actually a clear plastic material, carbon black is added to it as a *filler* in order to hide air holes and bubbles which would otherwise be visible and an unnecessary source of concern to the consumer. Some companies also add an antistatic compound to their vinylite to prevent the disc from attracting dust. This reduces surface noise resulting from dust in the grooves. While the antistatic compound is practical, it is expensive and the amount added must be carefully controlled for too much will make the records noisy.

The hardness of the plastic has an effect on the frequency response of the disc because it affects the amount of playback loss. Stiffer compounds result in more high-frequency output due to less deformation of the groove modulations by the playback stylus. Since vinylite is stiffer than lacquer, the finished record sounds brighter than a reference disc cut with the same settings. The difference in frequency response is on the order of 1 to 3 dB at 10 kHz.

When a record is being readied for production, only a few discs are pressed initially. These are called *test pressings* and are used for checking the finished product for clicks, pops, skips, and nonconcentric center holes which can cause wow. The pressing plant checks several of these first and, if they consider them acceptable, sends them to the record company which then sends one or two of them to the producer for his approval. Quality control by the producer at this point is very important; any defects not noted and corrected on the test pressing will occur on the finished product. Unacceptable records often slip by the pressing plant quality checkers, and record companies are sometimes slow in checking for defects. The producer and engineer responsible for the record are most familiar with the master tape, and they are therefore best able to determine if any problems exist on the record. Since the presses do not have a chance to reach optimum temperature during the short run made for test pressings, these records usually have somewhat more surface noise than the finished pressings.

The record companies assign each record a *release date,* at which time *disc jockey (DJ) copies* are mailed to radio stations. The *commercial* records are shipped to record distributors slightly after this date. The DJ copies of albums are the same as the commercial copies except that the label attached to the disc usually contains timing information not included on the commercial copies. This label is often a different color too.

On DJ copies of singles, the "plug" side or *A side* is usually recorded in stereo on one side and in mono on the other side. Most commercial singles are now issued with a stereo A side and a different stereo B side. Although the consumer gets two songs, the record companies do not have to worry about the radio station diluting the promotion for the plug side by playing the wrong song. Before stereo singles became popular, the plug side was pressed in mono on both sides of the DJ copies. The stereo side is intended for use by fm stereo pop music stations, while the mono side is for use by a-m stations, which can only broadcast in mono.

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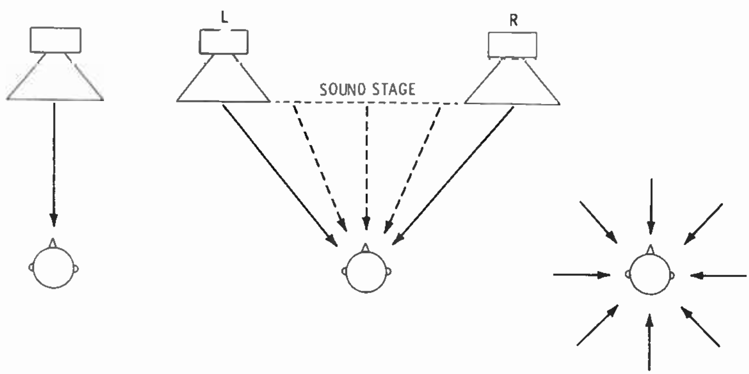
CHAPTER 13

Quadraphonic Disc Systems

Both mono and stereo sound produce the effect of looking into a sound field rather than being inside it as is the case when listening to a live performance (Fig. 13-1). In a live sound event, the sound field is formed by direct sounds which can originate from any direction around the listener and reflected sounds which can also originate from any direction. Since people exist in three-dimensional space, perfect reproduction requires that the reproduced sound field also define three dimensions: left and right, front and back, and up and down.

The point source created by mono reproduction is accurate only in re-creating the sound field which would be created by a single instrument or a group of instruments at a distance, as observed from within a nonreflecting environment such as an outdoor open field or



(A) A mono reproduction system.

(B) A stereo reproduction system.

(C) A live sound event.

Fig. 13-1. Forms of sound presentations.

an anechoic chamber. Sounds reaching the listener from directions other than that of the speaker are a function of the listening room acoustics, not of the recorded sound field. Differences in loudness between instruments in a mono recording can create an impression of distance or depth as can adding reverberation to instruments, but the proximity of an instrument is limited by the distance between the listener and the speaker.

Stereo reproduction projects a sound stage between the two speakers allowing the left and right dimension to be accurately presented. As with mono, however, the recorded direct and reflected sounds reach the listener from the front with the resulting perceived effect of listening to the instruments close up, but still from an unnatural environment. The sounds reaching the listener from behind, above, and below are still more related to the nature of the listening room than the nature of the recorded sound field. Direct/reflecting speakers aim much of their output toward the walls of the listening room so that sound reaches the listener from all directions, placing the listener inside the sound field, but this field is again more a function of the listening room than of the recording producer's intent.

Accurate reconstruction of a sound field is approached when an additional speaker is placed behind the listener because he is then within a plane defined by the three speakers. With three channels, both left and right and front and back dimensions can be presented, allowing real depth to be reproduced and allowing the localization of sources closer to the listener than the speaker distance.

Since most performance halls are rectangular rather than circular or triangular, a fourth channel and speaker added to define a rectangle around the listener helps re-create the original (or imaginary) performance hall in the listening room. The fourth speaker adds resolution to the sound field and reduces the disruption of the furniture layout in most rooms by replacing the center rear speaker with speakers in the two rear corners.

Height could be introduced to the sound field by adding a fifth speaker above and a sixth one below the listener. However, the improvement in sound field accuracy would be marginal since few sounds normally originate directly below a listener and overhead sounds can be simulated by feeding the signal to all four speakers equally.

Since more consumers listen to disc recordings than to cassette, cartridge, or reel-to-reel tapes, the popularity of four-channel or quadraphonic (quad) recordings depends on whether they can be recorded on discs. While tape can easily store four channels on parallel tracks, discs are inherently a two-channel medium, so that the four channels must be processed to make quad discs possible. To prevent the confusion and inconvenience that would result if separate stereo and quad versions of every lp had to be issued, record companies have decided that quad discs must be *compatible* with present stereo and mono playback systems and produce the same high-quality sound as stereo or mono discs.

In addition, playing a quad disc on stereo or mono equipment must not impair the recovery of the original four channels when reproduced with quad equipment at a later date. This enables consumers to buy quad discs and listen to them in stereo until they convert their playback systems to quad. A compatible quad system will enable the record companies to phase-out stereo lps the way mono lps have been replaced by compatible stereo ones and eventually release only compatible quad recordings.

A set of requirements for an ideal quad system have been layed out by Scheiber (1971) as follows:

1. Basic four-channel performance:
   (a) The ability to record sounds occurring at any point in 360°, and to reproduce each sound from the correct location in playback;
   (b) nondegradation of signal quality, including noise, frequency and nonlinear distortion as consistent with highest standards in the state of the art.
2. Compatibility:
   (a) four-channel compatibility: nonobsolescence of playback equipment, using standard components and construction wherever possible;
   (b) stereo compatibility: the ability to reproduce the four-channel program on all standard two-channel 'stereo' equipment, with all sounds in the four-channel program heard in their proper left-right positions;
   (c) mono compatibility: monaural playback possible on all standard equipment, without losing, or altering the relative level of, any sound in the four-channel program.
3. Economy
   (a) adaptability to standard practices for software manufacture;
   (b) full playing time within a given format, as compared with the equivalent stereo recordings;
   (c) usable with all major recording media and, preferably, broadcast. [16]

It should be noted in reference to item 2c, that current fm stereo broadcasts are considered to be mono compatible even though center located signals play back 3 dB louder in mono than signals coming from the sides. It seems that a 3-dB increase in mono playback level, while not desirable, should also be permissible for signals located between two speakers in a quad mix.

FOUR-CHANNEL DISC SYSTEMS

Two methods of storing four-channel sound on disc have been developed: The *matrix* system and the *carrier* system.

Matrix

Matrix systems require an encoder to combine the four original channels into two, and a decoder to recover four channels from the two encoded ones. The 4-2-4 conversion is achieved through the use of phase and/or amplitude changes in conjunction with summing and differencing networks, as defined by the system matrixing equations. Unfortunately, the circuitry cannot completely isolate the original signals from each other, resulting in decreased separation between the channels. By varying the matrixing parameters, separation can be increased between certain channels at the expense of further decreased separation between others. Matrixes are designed so that the level of the desired signal is greater in the proper channel than its cross talk is in the other channels. This allows *gain-riding logic* circuits to increase the apparent channel separation by increasing the gain on the desired channel and decreasing the gain on the ones which contain the same signal at a lower level.

Carrier

The carrier system stores the extra two channels on the disc by using them to modulate a signal above the audio-frequency range. This requires a bandwidth increase from 20 kHz to 45 kHz on the disc, but it has the advantage of providing as much separation between channels as available on standard stereo discs. As a result, this system is often called *discrete*, although it will be shown later that it also uses a form of matrixing.

QUAD SYSTEMS

The following convention is adopted for the quad discussions in this chapter: L_f, L_b, R_f, and R_b represent the original left-front, left-back, right-front, and right-back channels, respectively; L_t and R_t represent the left and right channels to be cut on the compatible quad disc; L_f', L_b', R_f', and R_b' represent the left-front, left-back, right-front, and right-back channels recovered from L_t and R_t; C_f, C_b, C_L, and C_R represent the four phantom source positions created by feeding a signal equally to the front, back, left, and right pairs of speakers, respectively (Fig. 13-2).

The CBS SQ (Stereo-Quadraphonic) system is an example of a matrix system. Observation of the following three psychoacoustic phenomena led CBS engineers to the selection of their matrix parameters. The phenomenon of "front source dominance" occurs when the same signal is fed to all four speakers in a quad listening setup, but the signal is loudest in one of the front speakers. The sound appears to come from the front speakers alone, and the back speakers contribute only to the perceived loudness, not to source localization. This is a result of the acoustic shadow (cast by the head and outer ear) which blocks sound from the rear, especially at high frequencies. The rear speakers are heard as reflection of the sound from room surfaces. Due to the time delay involved in traveling to these surfaces and back to the ear, the Haas effect comes into play, making the source of the sound appear to be in the direction of the first wave to reach the ear, i.e., the front speakers. The second phenomenon is that of "back image contraction," in which the stage created by stereo speakers behind a listener appears to be about one-third as wide as that created by speakers the same distance apart but in front of him (Fig. 13-3). The third phenomenon is called "quadrature image shift." When a signal is applied equally to two speakers with the signal to one speaker phase shifted by a constant amount, the image formed tends to shift toward the speaker with the leading phase signal. At 0° phase difference, the sound is perceived as a well-defined point source between the speakers. As the phase difference is increased, the perceived image spreads out, until at 90° difference it completely spans the space between the two speakers. [5]

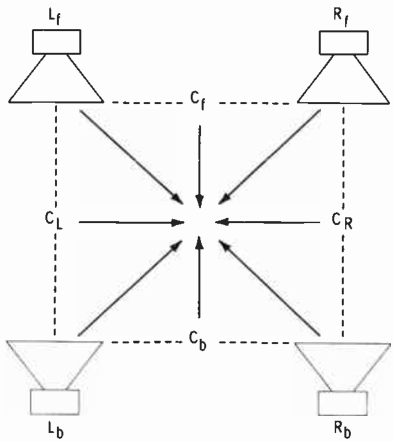


Fig. 13-2. The four sound sources in a quad system and the four principal phantom source images.

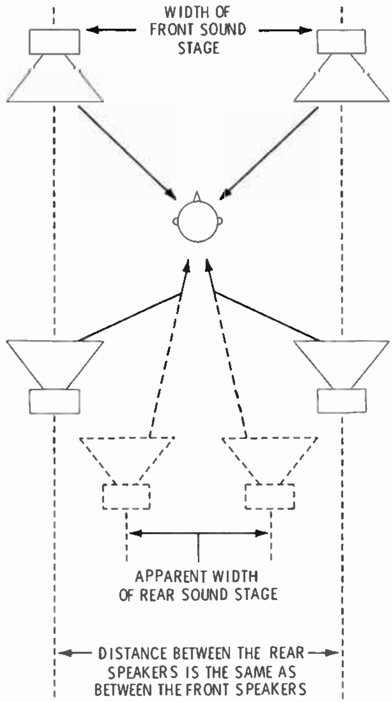


Fig. 13-3. The back image contraction phenomenon.

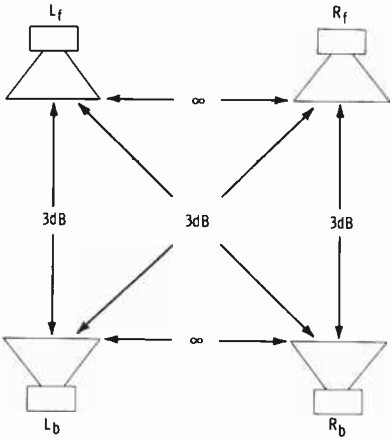
The SQ matrix is defined by the following equations. The term "j" indicates a 90° phase shift in the term following it:
encode:

$$L_t = L_f - j\ 0.707\ L_b + 0.707\ R_b$$
$$R_t = R_f - 0.707\ L_b + j\ 0.707\ R_b$$

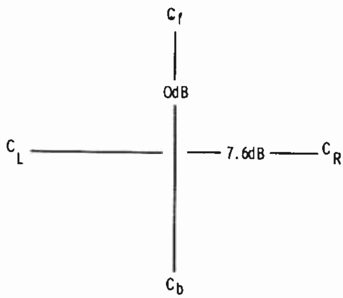
decode:

$$L_f' = L_t$$
$$L_b' = -0.707\ (-jL_t + R_t) = L_b + j\ 0.707\ L_f - 0.707\ R_f$$
$$R_f' = R_t$$
$$R_b' = 0.707\ (L_t - jR_t) = R_b + 0.707\ L_f - j\ 0.707\ R_f$$

Each output of the decoder (L_f', R_f', L_b', and R_b') contains the desired signal (L_f, R_f, L_b, and R_b, respectively) at the proper level, plus two *side-effect* components at a level 3 dB below the desired signal. The desired signals are all in phase as is necessary for proper reconstruction of image position in the quad sound field. The side effect signals in each of the front-channel outputs consist of only rear-channel information, and the side effect in each of the rear-channel outputs consist of only front-channel information. Infinite channel separation exists between the L_f and R_f components in the front speaker pair as well as between the L_b and R_b components in the rear speaker pair. Separation between either speaker in one pair and the speakers of the other pair is only 3 dB as a result of the side effect signals (Fig. 13-4). Images located at C_f or C_b in the discrete



(A) Between channels.



(B) Between phantom sources.

Fig. 13-4. Separation in the SQ system (with no logic or blending).

program are decoded as having 0-dB separation with the left and right signals in phase in the desired channels and 180° out of phase in the undesired channels. The in-phase signals produce a sharply defined image while the out-of-phase signals seem to come from all directions along the stage created by the speakers producing them. Separation between C_L and C_R signals is 7.6 dB. A signal positioned or moved anywhere in a 360° circle around a listener remains at a constant loudness through the encode/decode process so that the matrix does not change the musical balance of a program.

The front-source dominance effect enables sounds originating in the front channels to be positively localized in the front despite the presence of rear side effect signals only 3 dB lower in level. CBS acknowledges that this effect is lost if the listener faces the side, and so an optional logic circuit is provided to reduce the gain of the rear channels when they duplicate the front-channel information. Alternatively, the decoder can be modified to increase center-front to center-back separation by 6 dB by decreasing front-channel separation to 20 dB and rear-channel separation to 8 dB. Since the L_b

and R_b signals are louder in the rear channels than in the front, the front source dominance effect does not occur with them and they are localized in the rear.

The rear-channel information appearing in the front channels as side effects is decoded so that the phase of the L_b component in the L_f' channel leads the R_b component by 90°, and the phase of the R_b component in the R_f' channel leads the L_b component by 90°. Due to quadrature image shift, rather than appearing at center-front, the L_b components appearing as side effects in the front channels produce an image shifted toward the L_f' channel. The R_b components in the front channels produce an image shifted toward the R_f' channel. Since they come from the same side of the listener as the main image, the side effect images blend with the main image directionality rather than pulling them toward the center (Fig. 13-5). The higher level of the main back images results in only rear signals being perceived. Side effect images are also formed to the rear of the listener in response to front-channel information due to the decoder's symmetry, assuring that sounds are perceived as coming from the proper speakers whether the listener faces the front or the rear pair.

The L_t signal is cut as the left channel of the disc and the R_t signal as the right channel. Since the L_t channel contains no R_f signal and the R_t signal contains no L_f signal, the modulation produced by L_f and R_f signals alone is identical to the standard stereo 45/45 modulation. An L_b signal appears at 0.707 times its original level in both L_t and R_t with the component in L_t leading that in R_t by 90°, causing the stylus to move in a clockwise circle. Similarly, an R_b signal appears at 0.707 times its original level in both L_t and R_t but leads by 90° in R_t, so it causes the stylus to move in a counter-

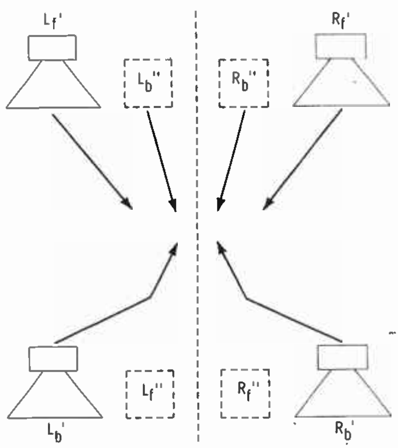


Fig. 13-5. Side effect images L_b'', R_b'', L_f'', and R_f'' are offset in the direction of their main images.

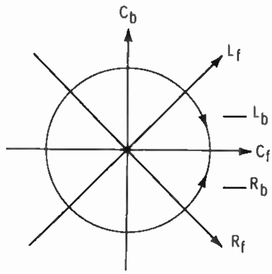


Fig. 13-6. Stylus motions for an SQ-encoded program.

clockwise circle. The lengthwise motion of the groove under the stylus causes these two circular motions to form a double helix. Center-front information causes lateral stylus motion while center-back information causes vertical motion (Fig. 13-6).

When played in stereo, the L_f and R_f channels appear as they would in quad, i.e., in front in the left and right speakers, respectively. The L_b and R_b signals appear displaced from center toward the left and right respectively as they should be (rather than dead center) due to the quadrature image shift phenomenon which shifts the image toward the phase leading speaker. This shifting results in the same stage width for the rear-channel signals playd in stereo as results from the back image contraction phenomenon when the recording is played in quad (Fig. 13-7).

The L_f, R_f and C_f images are sharply defined, producing an impression of nearness while the phase difference between the L_b and R_b signals contained in the L_t and R_t channels produces spreading of the L_b, R_b, and C_b images. This results in an impression of distance

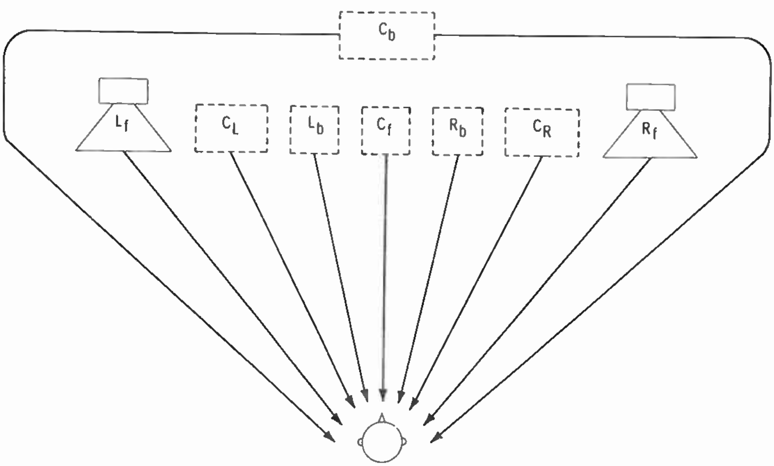


Fig. 13-7. The stereo stage created by an SQ-encoded disc played in stereo.

which gives the stereo image a feeling of depth similar to that created by the quad sound field. The loudness of a signal panned anywhere along a 360° circle around a quad listener remains constant when the recording is played back in stereo, so the instrumental balance does not change.

Mono listeners hear the sum of the L_t and R_t signals, resulting in the four channels being heard at equal levels. As in the playback of stereo program material in mono, center-front sources are 3 dB louder. Since center-back sources are present at equal level and 180° out of phase in L_t and R_t, any sound assigned to this position in the quad mix will cancel out and not be heard in mono. This prohibits soloists from being located in this position if mono compatibility is desired. Complete cancellation only occurs if the signal is centered precisely between the back speakers, so sounds panned in a circle appear continuous. Center-left and center-right signals are changed in magnitude when the recording is played in mono, with C_L at −5.3 dB and C_R at +2.3 dB, but the sum of the power contributed by them is the same as in quad playback.

While the standard SQ encoder processes signals located along the L_f/R_b speaker diagonal so that they are decoded in the proper location, input signals along the R_f/L_b diagonal are not decoded properly. In order to process R_f/L_b splits, a second encoder must be used with its inputs and outputs connected in reverse sequence so that the diagonal seen by the encoder is L_f/R_b. The outputs of the two encoders are added together to produce the L_t and R_t signals (Fig. 13-8). Separation between diagonal pairs is 7.6 dB.

To eliminate the need for a second encoder, CBS developed what is known as a *forward-oriented encoder*. This unit is capable of splits along both diagonals, but diagonal pair separation is reduced to 4.8 dB. This feature is achieved at the expense of accurate positional decoding of rear-quadrant signals. Center-back signals are encoded and decoded as center-front signals with this unit so that signals cannot be panned 360° around a listener. The forward-oriented en-

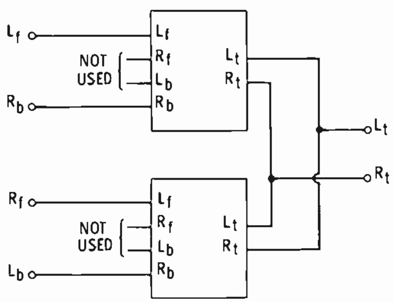


Fig. 13-8. Connecting two SQ encoders to permit encoding along the L_fR_b and R_fL_b diagonals.

coder has an advantage in broadcasting quad programs which have soloists in the center-back position; mono listeners will hear the soloist at full level (since he is encoded as a center-front signal). A center-back soloist would not be heard at all in mono if a standard SQ encoder is used.

A *backward-oriented encoder* is available to produce extra-wide stereo effects in the rear channels. This unit encodes and decodes center-front signals as center-back signals. Since the important front center image is lost to the mono listener, this encoder is usually only used for special effects occurring in the rear channels.

The SQ encoders were originally designed to encode a previously mixed four-channel tape into two channels for recording on disc. As a result, the spatial characteristics of the original mix had to be preserved by a single type of encoder. If information was desired in the center-back position and only a standard encoder was available, then R_f/L_b diagonal information could not be properly encoded. Correct encoding for all positions and movements requires that each signal be processed separately by the proper encoder. This can only be done at the multitrack mixdown stage.

To facilitate this, CBS has developed their Model 4211 SQ encoder which is capable of forward- and backward-oriented encoding modes, as well as standard encoding, and their Model 4212 SQ position encoder. Each channel of the multitrack tape is fed to a separate input on the position encoder. A pan pot in conjunction with a selector switch for each input makes either 360° or figure-eight panning patterns possible by directing the signal to the encoder inputs that will best encode its position and movements. A third module, the Model 4213, plugs into the 4211 to provide four-discrete outputs with content determined by the settings of the position encoder pan pots so that both SQ encoded two-channel and discrete four-channel tapes can be recorded simultaneously. The four-channel tape is used to produce quad reel-to-reel and cartridge tapes. The accuracy of the encoding can be checked by monitoring the signal fed to the two-channel recorder through an SQ decoder (Fig. 13-9).

The SQ system meets all of the requirements laid out by Scheiber, except for complete mono compatibility due to the cancellation of center-back information and economy because accurate encoding of certain positions must be done at the multitrack level using a position encoder which is separate from the encoder itself. Interfacing the position encoder requires an additional console connection for each pan pot used. A more economical system would encode all positions accurately from either the four console outputs or from a previously mixed discrete four-channel tape.

A second matrix system is the Sansui QS. This system provides 3-dB separation between signals in adjacent speakers and infinite

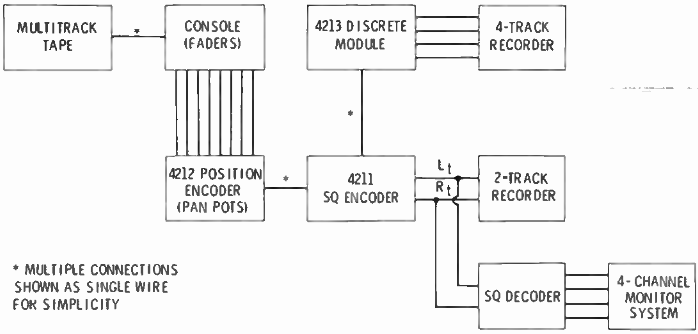


Fig. 13-9. SQ position encoding at the multitrack stage.

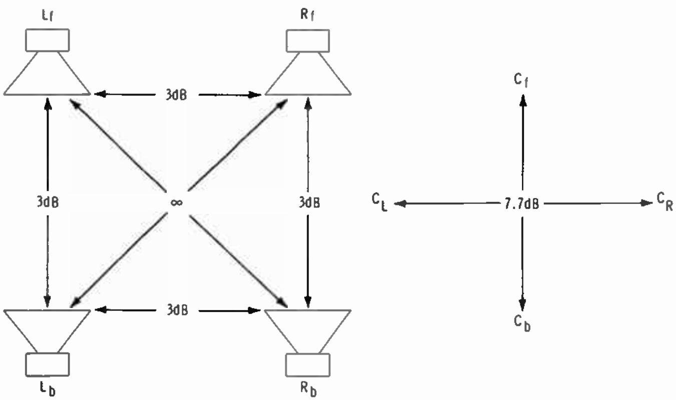
separation between signals in diagonally opposite speakers. Separation between center-front and center-back or between center-left and center-right images is 7.7 dB (Fig. 13-10). The matrixing equations for this system are:

encode:

$$L_t = L_f + 0.414\,R_f + jL_b + j\,0.414\,R_b$$
$$R_t = 0.414\,L_f + R_f - j\,0.414\,L_b - jR_b$$

decode:

$$L_f' = L_t + 0.414\,R_t = 1.172\,L_f + 0.828\,R_f + j\,0.828\,L_b$$
$$L_b' = -j\,(L_t - 0.414\,R_t) = -j\,0.828\,L_f + 1.172\,L_b + 0.828\,R_b$$
$$R_f' = 0.414\,L_t + R_t = 0.828\,L_f + 1.172\,R_f - j\,0.828\,R_b$$
$$R_b' = j\,(-0.414\,L_t + R_t) = j\,0.828\,R_f + 0.828\,L_b + 1.172\,R_b$$



(A) Between channels. (B) Between phantom sources.

Fig. 13-10. Separation in the QS system without Vario-Matrix.

Since the cross-talk components in this system are at equal levels on either side of the main speaker, they create an image between them which tends to reinforce the main image (Fig. 13-11). One of the cross-talk signals is in phase with the main signal while the other is phase shifted by 90° so that the image shifts slightly toward the phase-leading speaker or speakers.

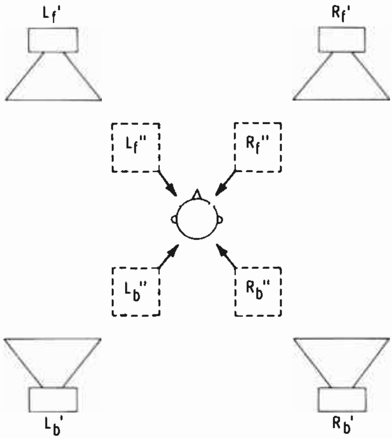


Fig. 13-11. The cross-talk images reinforce the main images.

Rather than adding standard gain-riding logic to the decoder to increase the separation between channels in quad playback, Sansui has developed what they call the 20-dB or "Vario-Matrix" system to augment their standard decoder. This circuit changes the phase and amplitude parameters of the decoding equations on the basis of the front-back and left-right signal distribution.

As an example, consider the decoded distribution of a signal originally applied only to the L_f channel of the encoder. This signal appears at full level in the decoder L_f' output, 3 dB down in the L_b' and R_f' outputs, and does not appear at all in the R_b' output. If the decoder matrix equation parameters for the L_b' output are shifted toward those of the R_b' output, separation between the signals appearing at the L_f' and L_b' outputs increases. When the L_b' equation is the same as the R_b' equation, the L_f'/L_b' separation becomes infinite. If an L_b signal is simultaneously applied to the encoder, the shifted decoding equation would result in its being decoded 3 dB too low in level, so the gain of the L_b' channel is boosted by 3 dB at the same time the matrix parameters are shifted to make up this loss. If signals are present in the other encoder inputs as well as in the L_f and L_b, the L_b' parameters are varied to a lesser extent, determined by the signal distribution. Since the matrix is symmetrical, the parameter variation is equally effective with all four-decoding equations, and it increases separation to at least 20 dB in all directions [18].

When played in stereo, the QS system provides 7.7-dB separation between the L_f and R_f, and between the L_b and R_b channels. The C_b, L_b, and R_b channels are reproduced out of phase in the left and right speakers so that they produce images which appear to originate from sources farther apart than the distance between the speakers. This effect compensates somewhat for the reduced L_f/R_f stage (Fig. 13-12). C_f and C_L signals are reproduced 2.3 dB louder in stereo than in quad. In mono playback, the out-of-phase components in L_t and R_t cause a 7.7-dB reduction in the level of the L_b and R_b signals and complete cancellation of C_b signals.

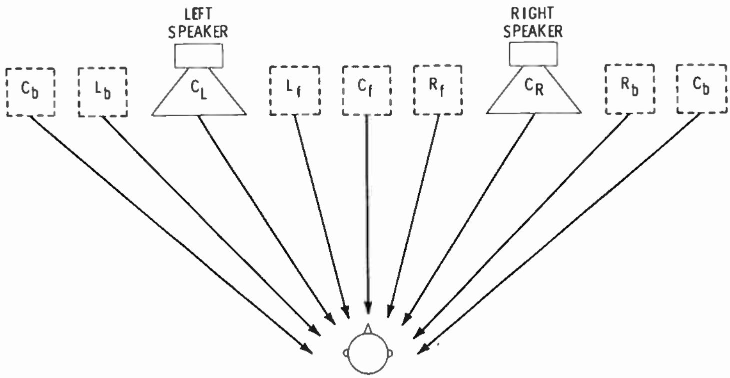


Fig. 13-12. The stereo stage created by a QS-encoded disc played in stereo.

The L_b and R_b signals require 30% more cutting depth on a QS encoded disc than the same signals on an SQ encoded disc. This means that if the rear channels contain loud signals, a QS disc must be cut 2.3 dB softer than the same program on an SQ disc.

The QS system meets the Scheiber requirements of an ideal quad system except for mono compatibility and the use of full cutting levels on a disc when loud information is present in the rear channels. While a wide stage is achieved in stereo through the use of out-of-phase L_b and R_b components and left-right directionality is preserved, the important L_f/R_f portion of the stage is considerably narrowed from the original. In addition, in placing the rear channels outside the speakers rather than between them, the QS system does not take advantage of the back-image contraction phenomenon in creating a stereo image similar to the quad image.

The JVC CD-4 (compatible-discrete 4-channel) carrier system provides a means of storing four channels on a disc without sacrificing channel separation. Each groove wall contains a main audio signal with a bandwidth of 15 kHz and a 30-kHz modulated carrier signal with a bandwidth extending from 20 kHz to 45 kHz. The carrier is recorded 19 dB below the main audio signal level and is frequency modulated below 800 Hz and above 6 kHz, while phase modulation is used for signals in the midrange.

The main signal on the inner groove wall consists of the sum of the L_f and L_b channels ($L_f + L_b$), while the carrier signal contains the difference of these two channels ($L_f - L_b$) (Fig. 13-13). The R_f and R_b sum and difference signals are recorded on the outer groove wall. The four channels are recovered by a *demodulator* which separates the main audio signal from the carrier using low-pass and

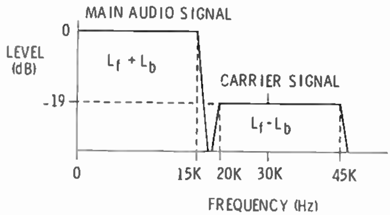


Fig. 13-13. The signals recorded on the inner groove wall of a CD-4 disc.

bandpass filters. The carrier is demodulated, producing a second audio signal. Through the use of an amplitude matrix, these two signals are added to form the front signal output and subtracted to form the back signal output according to the following equations:

$$L_f' = 0.5\ [(L_f + L_b) + (L_f - L_b)] = L_f$$
$$L_b' = 0.5\ [(L_f + L_b) - (L_f - L_b)] = L_b$$
$$R_f' = 0.5\ [(R_f + R_b) + (R_f - R_b)] = R_f$$
$$R_b' = 0.5\ [(R_f + R_b) - (R_f - R_b)] = R_b$$

There are no cross-talk components generated in this system. The separation is better than 25 dB between any two channels and is limited by the cross talk inherent in discs and phono cartridges.

The use of sum and difference matrixing ensures stereo and mono compatibility. The stereo listener hears only the main audio signals; he hears the front and back channels at equal level and from the same apparent location. The C_L and C_R images reproduce 3 dB louder in stereo than in quad (Fig. 13-14). A mono listener also hears the four channels at equal levels. Rather than C_b signals being cancelled in mono playback, C_b as well as C_f, C_L, and C_R signals are reproduced 3 dB louder. Any signal centered in the quad sound field (fed equally to all four channels) reproduces 6 dB louder in mono.

The problems of cutting 45-kHz signals on a disc are alleviated by cutting the disc at half speed, so that the 30-kHz carrier becomes only 15 kHz to the cutter head. This is done by slowing down both the lathe and the tape deck playing back the master tape. Slow-speed cutting is necessary due to the limited frequency response of present cutter heads. The carrier is cut 19 dB below the level of the main signal to prevent overloading the cutter head, mistracking of the playback stylus, and damage to the carrier by the playback stylus. This low level makes cross talk from the sum signal and surface noise noticeable in the difference signal. To eliminate these problems, an Automatic Noise Reduction System (ANRS) is used both in cutting and in playback. The ANRS is a frequency-sensitive compressor/expander system, somewhat similar in operation to the Dolby system. It reduces the cross talk and surface noise, resulting in signal quality comparable to that of regular stereo discs. The ANRS is used only on the difference signal so that the mono and stereo compatibility of the disc is not impaired. A system called *Neutrex* is used to reduce the playback tracing distortion caused by the narrow radius of curvature of the carrier modulations.

Quad playback of the CD-4 disc is done with a Shibata stylus in a cartridge which has a smooth response up to 45 kHz and has output ±15 dB between 20 kHz and 40 kHz, relative to 1 kHz. The Shibata stylus can track high-frequency groove modulations better than an elliptical stylus and has a groove contact area four times larger, thus reducing wear on the fragile carrier signal. The demodulator contains a limiter which compensates for any fluctuations in carrier level due to wear. A muting circuit automatically disconnects the difference-signal channels of the demodulator when the carrier is not present so that stereo and mono discs can be played through the system.

While the CD-4 system produces more accurate disc reproduction of quad recordings and has better mono compatibility, it does not

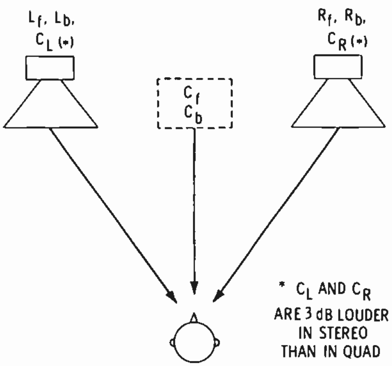


Fig. 13-14. The stereo stage created by a CD-4 encoded disc played in stereo.

meet some of the other Scheiber requirements as well as the SQ or QS systems. The frequency response of the CD-4 system is limited to 15 kHz at the high end, which represents a slight compromise in signal quality. The presence of the carrier takes up some modulation space in the groove, resulting in maximum cutting levels about 2 dB lower than those of regular stereo discs. Noise, distortion, and playing time are the same as for stereo discs, as is record life when played with a Shibata stylus. The CD-4 system requires that the consumer buy a new phono cartridge or modify his old one by adding a Shibata stylus (to enable it to track the carrier signal) in addition to buying a demodulator. This makes the consumer's current cartridge or stylus obsolete.

At present, the CD-4 system cannot be used for quad broadcasts due to FCC restrictions on the bandwidth of fm stations. Systems are being developed, however, which would enable CD-4 recordings to be broadcast in the event the regulations are changed. Due to the extra bandwidth required, CD-4 programs cannot be recorded on two-channel tape, and a 4-channel playback machine with a 4-channel preview head is needed for disc mastering, doubling the number of mastering console channels to eight.

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Appendix

The following pages are reproduced for those readers desiring further technical insight into the nature of magnetic tape and recording. The material was originally published in the 3M Company's "Sound Talk" bulletins and is reproduced with their permission. ("Sound Talk" is a registered trade mark of the 3M Company.)

Volume I
No. 1
1968

DESIGN CONSIDERATIONS TO INSURE INTERCHANGEABILITY OF RECORDING TAPE

As the variety of magnetic recording tapes increases, questions are being asked about tape similarities and differences. What are the basic differences between the popular types of tape? Is it possible to interchange the different types of tape without sacrificing performance? How can I achieve maximum performance from my recording system using a specific tape?

Because of the large variety of professional and home recording systems, each of them built to individual manufacturing specifications with different settings and adjustments, it would be impossible to list all the specific differences for each system. Quality magnetic recording tape is manufactured to established specifications and its performance is predictable and easily measured. Reviewing some of the individual properties and characteristics of recording tape will show us what the tape is capable of reproducing with specific inputs. Observing some of the similarities and differences in performance will show some of the requirements for optimization of the recording system to take advantage of the individual characteristics of each tape.

For convenience this paper will cover the three most popular types of magnetic recording tape.

We will consider "standard" recording tape as a reference and will use it as a basis of comparison for the other tapes.

Classification	*Example of Commercial Number*
Standard	#111, 102
Extra-play	#150, 190 & 200
Low Noise	#201, 202, 203

TAPE PROPERTIES

Magnetic recording tape receives and retains magnetic signals from the recorder head. The thin layer of ferric oxide, coated on a polyester or acetate backing is the substance that reacts to magnetic signals. The magnetic properties of an oxide coating are the basic factors which determine the differences between tapes. Certainly, the thickness of the oxide coating, the application and purity of the oxide, particle size and orientation, and the type and thickness of the backing are all variables, but for this discussion we will be concerned with the magnetic properties. Future *SOUND TALK* Papers will explore the types of backing and the physical parameters of tape more completely as separate subjects.

The parameters which identify the fundamental magnetic differences between tapes are the intrinsic magnetic properties of Coercivity, Retentivity, and Remanence. The intrinsic magnetic properties are the measurements of magnetic flux interaction with the tape's coating; and the coatings ability to receive and retain the magnetic signal.

COERCIVITY

As a strict definition, coercivity is a measure of the magnetic flux intensity required to return a magnetic material from saturation back to zero. Practically speaking, it represents the flux intensity or magnetic field strength required to record a magnetic signal onto the tape. A high coercivity tape requires a greater flux intensity (a higher signal and bias level from the recorder head) to record on the tape. An example is low noise tape which has a coercivity measurement of 315 oersteds (a unit of measure of magnetic field strength). In comparison, standard tape has a coercivity measurement of 270 oersteds which indicates a lower flux density is required to record on this tape. The extra-play tape has an even lower coercivity of 260 oersteds which shows that this tape will respond to an even lower flux density level. The coercivity of a tape is a function of the basic oxide particles used to form the dispersion that will ultimately become the coating. Coercivity, therefore, is a measure of the magnetic field strength required to establish magnetism in the coating.

RETENTIVITY

Now that we have magnetized a section of tape with a signal from the recorder head, the next tape parameter is concerned with how much of the signal, in terms of magnetic strength, is retained in the tape coating the instant it leaves the influencing field of the recorder head. This is known as Retentivity; which is the measurement of the number of flux lines (or gauss) per square centimeter of the coating cross section (width of tape and the coating thickness).

Although some tapes respond to a magnetizing signal output more readily than others, they all will retain the resultant magnetic impulse indefinitely. Retentivity is primarily a magnetic property of the coating dispersion (particle size, density, and composition) without reference to the tape size, and it varies with the particular coatings used for recording tapes. A typical retentivity measurement for standard tape is 920 gauss (a unit measure of magnetic induction or quantity value of magnetic flux). The dispersion used for a low noise tape has a retentivity of 790 gauss which is lower than standard tape. The extra-play tape has an extremely high retentivity of 1120 gauss. Each dispersion used in the manufacture of the three basic tapes has a different value of retentivity. This value however, defines one of the properties of the dispersion before being coated onto a backing. A more meaningful measurement to the user would take into account the result of applying the dispersion in a given thickness to a particular width of backing. Since the majority of recorders use a ¼" wide tape, the industry developed a parameter with the ¼" as a constant. This is known as remanence. Since the tape width is a constant, the two immediate variables are coating thickness and dispersion type.

REMANENCE

Remanence is the actual magnetic signal retention as applied to a specific tape cross section. For our purposes, we will regard remanence as the induced magnetic flux remaining in a ¼" wide tape after a longitudinally applied field is reduced in intensity from 1000 oersteds to zero. This is explained simply by saying that a ¼" wide tape will have retained the recorded magnetic signal and will exhibit a magnetic field of its own. The remanence property therefore, is what the playback head is magnetically exposed to.

As previously shown, the retentivity of the three basic oxide dispersions are all quite different. From this, one might expect different results in terms of playback. But, by carefully controlling the application of the coating, the remanence value can be established at a desired point. To assure proper interchangeability of the three tape types, the coating variables are structured so that the remanence value is the same for all three tapes. EACH OF THE THREE TAPES HAVE A REMANENCE MEASUREMENT OF 0.64 FLUX LINES PER ¼ INCH WHICH ASSURES A MAGNETIC COMPATIBILITY AND PLAYBACK INTERCHANGEABILITY BETWEEN ALL OF THE THREE TAPES. While the control of the remanence value allows the tapes to be interchanged, the differences in coercivity and oxide retentivity require slightly different magnetic signal input levels during recording to fully exploit the abilities of the different tapes. Some of the differences that the tapes exhibit and the corresponding machine adjustments for maximum performance will be shown in the following paragraphs.

TAPE CHARACTERISTICS

The differences in the magnetic properties are reflected in the particular characteristics which each tape exhibits. Assuming a tape speed of 7.5 ips, some of the characteristic differences can be easily shown in the frequency response curve for each tape (Figure 1). These curves were generated on a good quality professional machine and show the difference in both the low and the high frequency response. To establish these response curves, the recorder was adjusted for maximum performance using the standard type of tape. The record level, bias, and record equalization were set to achieve the best response possible from this machine with standard tape. The individual magnetic properties of each different tape became apparent in the differing output and response when each is run without readjusting the machine.

RESPONSE

The ideal response curve would assume a straight line from the low to the high frequencies, but is limited by the recorder electronics. Note that the standard tape (which had the optimum settings) is nearly level until

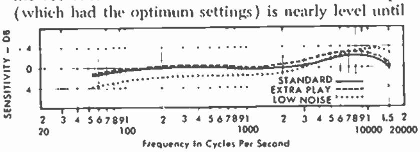


FIGURE 1. FREQUENCY RESPONSE — TAPE AND RECORDER

the high frequency roll off. The extra-play tape, with slightly lower coercivity, shows a slight increase in high frequency response but has a sharper roll off. The high coercivity, low noise tape shows a slight decrease in sensitivity at low frequencies but has a prominent increase at the high frequencies with less roll off. Figure 1 shows only a comparison of typical response for the three tapes, but from our discussion of coercivity, one will remember that each tape required differences in the input signal level. The desired frequency response for either high or low frequencies, *for each particular tape*, can be achieved by changing the bias and record equalization levels. Variations in the bias and equalization settings and the corresponding changes in output will be shown in later paragraphs.

It is possible to design a recording tape for maximum output at either high frequencies (short wavelength) or low frequencies (long wavelength) by formulating different coating dispersions. Variables would be coating thickness, coercivity, and retentivity. In the design of Audible Range Magnetic Tape the challenge lies in the ability to produce a tape that is capable of uniform output over the broad range of wavelengths from less than ½ mil to more than 30 mils. A factor which can affect response is the smoothness of the surface coating. A very smooth coating surface insures maximum contact with the recorder head therefore allowing a maximum of magnetic signal changes to interact on the tape. Minute variations in the coating surface, as found occasionally in low quality recording tapes, will create variations in the head-to-tape contact which will change the magnetic flux level and will affect the playback signal from the tape to playback head.

While discussing tape response, it would be well to mention that recording tape sees the recording in terms of wavelength and not frequency. This is understandable when one considers that there are two variables that affect the recording process. One is the frequency that is being recorded, the other is the relative speed of the tape passing the recording head. Suppose that a tape is travelling at 7½ ips, and that a 7.5 kHz signal is being recorded. This means that 1000 cycles of information are packed on each inch of tape. The distance encompassed by each complete cycle is 1/1000 inch. The wavelength of this recording is 1 mil.

Expanding the previous example, we find that doubling the frequency 15 kHz will cause 2000 cycles of information to be placed on each inch of tape. This renders a recorded wavelength of ½ mil, as each individual cycle takes up .0005 inch of tape. If we reduce the tape speed to 3¾ ips, and leave the frequency to be recorded at 7.5 kHz, we once again will be recording a ½ mil signal. If both the frequency and the tape speed are doubled, the tape will see no change in recorded wavelength.

Since the information is recorded on the tape coating magnetically, it could be viewed by applying a fine metallic powder to the tape and viewing it with a magnifying glass (Figure 2). Notice the variations in magnetic pole density, the low frequencies are widely spaced (long wavelength) and the high frequencies are packed very close together (short wavelength). When recording at the short wavelengths, the coating which becomes *magnetized for each cycle* of information; must faithfully establish each set of poles without disturbing the preceding pole. When the magnetic poles are very close together the coating's ability to receive and hold magnetization (coercivity and retentivity), without influence from adjacent magnetic fields is very important. The oxide dispersion must be carefully prepared and applied to assure correct coating caliper, coating density, coating surface smoothness, as any variations will create changes in the magnetic properties and the sensitivity.

FIGURE 2. MAGNETIZED PORTION OF RECORDED TAPE

SIGNAL TO NOISE

Another important tape characteristic is the signal to noise ratio. Tape noise is strongly influenced by the particular type of oxide coating. Specially designed low noise oxide and precise manufacturing control allows the production of tape with a greatly improved signal to noise ratio. A comparison of standard and low noise tape shows a difference of 6 db at 500 cps to the upper limits of the audible spectrum (Figure 3). Low noise tape, as you recall from earlier paragraphs, has a high coercivity coating. This is a basic property difference of the special low noise oxide. Tape noise is a very low level signal and may be masked by the recorded sounds but does become critical during quiet musical passages. This extension of the dynamic range is important for full fidelity enjoyment. For maximum purity of reproduced sound, the use of low noise tape is recommended. In addition to the benefit of noise reduction, the coating properties of this tape give greater fidelity and response in the high frequency region of recording.

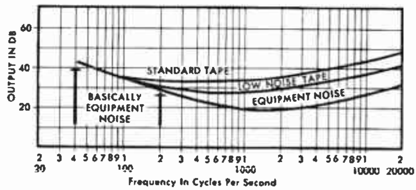


FIGURE 3. NOISE BY ⅓-OCTAVE BANDS

PERFORMANCE

Because of the different coatings available, magnetic tape is manufactured to meet a variety of recording requirements. Although the Audible Range constitutes a challenge because of the frequency bandwidth, an important factor is the production of tapes which are compatible to each other on the large variety of recording systems available. As previously shown, each particular type of tape has its individual magnetic properties which respond, in terms of maximum performance, to specific input levels.

BIAS

Figure 4 shows a typical example of optimized bias settings for each of the three tapes. Some of the recorders now in use have a fixed bias level which cannot be changed, but the majority of them offer some control over bias level. As part of the basic design, each recorder manufacturer establishes a bias level which is adjusted for a particular recording head and a laboratory tape. On any recorder care should be taken in making bias adjustments, and the recommendations of the recorder manufacturer should be followed. Professional recorders have specific adjustments for bias and equalization, and these adjustments can be made with more ease. Because of the large variety of recorders available, each with their own specifications, no attempt has been made to indicate bias level on the graph in Figure 4 in terms of actual bias current.

Bias level is indicated as a percentage of that which is proper for standard tape; the standard tape value being 100%. The percentage value relationship will hold generally true for all recorders. In a comparison of the bias current vs. output curves, note that the output in db is the same for each tape, but notice also the difference in bias level requirements. The high coercivity low noise tape requires additional bias for a given output in comparison to the standard and extra-play tapes which require less bias.

BIAS AND FREQUENCY

As can be seen from the graphs in Figure 4, the recommended peak bias for a given tape type is that point where the 2 curves cross. With standard and extra-play tape as bias is decreased below the recommended peak, the short wavelength output will be increased but the long wavelength response will suffer. If the bias is increased, the long wavelength response will be improved at the expense of the short wavelength. It is interesting to note that the high coercivity coating used in the low noise tape has essentially the same bias requirement for both low (15 mil wavelength) and high (1 mil wavelength) frequencies. This tape, although requiring greater input drive, allows a bias setting which compliments both the high and low frequencies.

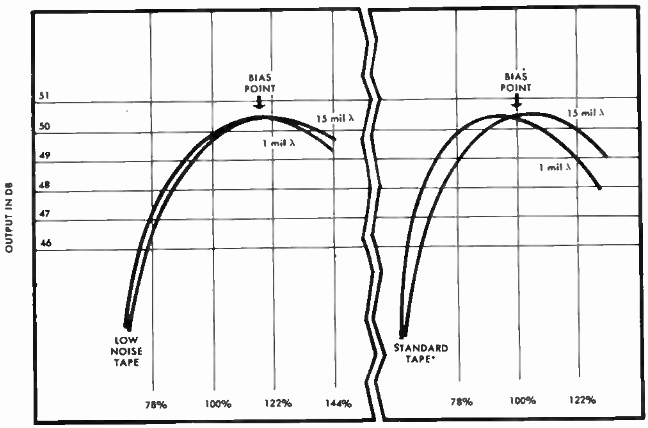
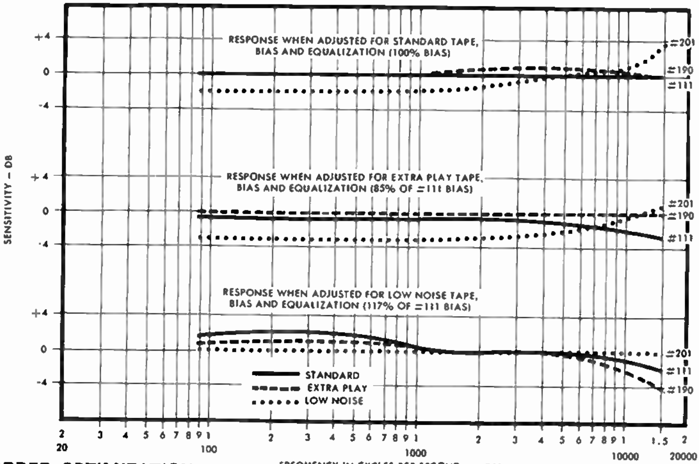


FIGURE 4. BIAS AND WAVELENGTH RESPONSE



FREQUENCY IN CYCLES PER SECOND TEST RUN AT 7.5 IPS

FIGURE 5. NORMALIZED TAPE RESPONSE CURVES

RECORDER OPTIMIZATION

Recalling the discussion of magnetic properties, it was stated that the coercivity measurement represents the flux intensity or magnetic field strength required to record a signal on a section of tape. To maintain a specific output level, a high coercivity tape requires a greater input signal level and, in comparison, the lower coercivity tape requires less drive. Notice, though, that in all cases the bias requirements for a given tape type do not constitute a major change in the recording system. To compensate for the differences in tape sensitivity, the equalization settings of the recorder can be adjusted so the frequency response curve will achieve the desired overall flat response. Figure 5 shows the result of optimizing bias and equalization for each tape type and the effect this has on the other two tapes. To achieve perfect results these can be adjusted, but because the amount of change in record level, bias, and equalization is only minor, the average outputs from the different tapes do not vary widely from each other. The three tapes can be interchanged without any severe decrease in overall performance.

Flat response can be attained, within the limits of the recorder amplifiers, with specific input settings. As shown in Figure 5, the plotted sensitivity range for the different tapes is about 3.5 db. The differences in bias and equalization for flat response for a specific tape will create slight response differences for the other tapes. The response curves exhibited by each tape show that a compromise setting can be used so that all the tapes will produce a similar and relatively flat response curve. The recommended compromise setting for tape interchange is at a bias setting of the standard tape (100%) or just slightly higher. With a bias setting of 105 to 110%, the three tapes will achieve a similar response with less than 1.5 db difference. With this compromise setting, the equalization can remain at a point that was proper for the standard tape.

SUMMARY

The three most popular types of magnetic recording tape do exhibit individual magnetic properties which are a function of the oxide dispersion forming the tape coatings. For maximum performance with a particular tape, the recording system can be adjusted for optimum bias, record level, and equalization. Because the individual differences are not extreme, a compromise setting can be used so the tapes can be interchanged without appreciable loss in performance. By using a recording tape which is properly designed and manufactured, an increase in overall performance can be attained without sacrificing tape-to-tape and tape-to-recording system compatibilities.

If at any time additional information on this topic is desired, it is available by simply writing to:

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No. 2
1968

HIGH FREQUENCY BIAS REQUIREMENTS FOR MAGNETIC TAPE RECORDING

Magnetic recording tape provides a superior method for the permanent recording of information, but it is limited by the natural phenomena of magnetic properties. Fortunately, the shortcomings created by this magnetic phenomena can be compensated for by the use of electronic measures. This bulletin will make no attempt to explore the mathematical or theoretical realms of magnetic recording, but will present a simplified explanation of high frequency bias, its requirements and limitations, and methods of adjustments.

Every magnetic medium exhibits a non-linear characteristic because the magnetization, resulting from an exposure to a magnetic field (such as that produced by the recording head), is not directly proportional to the strength of the field. This non-linear characteristic, if not corrected, would result in severe distortion of the audible recorded information. The use of a high frequency bias current; applied through the recording head, is the standard method of compensating for the non-linearities in the transfer of electro-magnetic signals onto magnetic recording tape.

The high frequency bias signal is usually generated by an oscillator circuit in the recorder electronic system and is added to the signals generated by the microphone or supplied by the recorder input circuits. The bias is a high frequency, usually 30 to 100 kHz (KiloHertz), which is above the range of hearing. Therefore, during playback of only the bias signal, one would not hear or identify any tones which would identify its presence. By adding the bias signal to the audio signal, a resultant signal is produced (Fig. 1). In most recorders, the two signals are simply combined without any form of modulation. The resultant signal is what the record head inductively converts from electrical signals into magnetic fields which influence the magnetic tape.

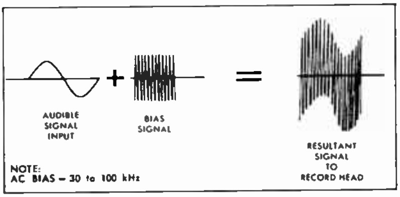

FIGURE 1. PRODUCTION OF RESULTANT SIGNAL

As previously stated, every magnetic medium exhibits a non-linear characteristic. This non-linearity is best illustrated by the Transfer Characteristic Curve which is mathematically derived from a family of hysteresis loops (Figure 2). The hysteresis loops and transfer curve indicate the degree of tape magnetization which results from an exposure to a magnetic field such as that produced by the record head. The transfer curve also indicates that the non-linearities exist only at the extremely low signal level (center portion of the curve) and at the very high signal levels (saturation areas) which are at the extreme ends of the curve. The remainder of the curve is relatively straight and allows linear and proportional transfer of magnetic signals.

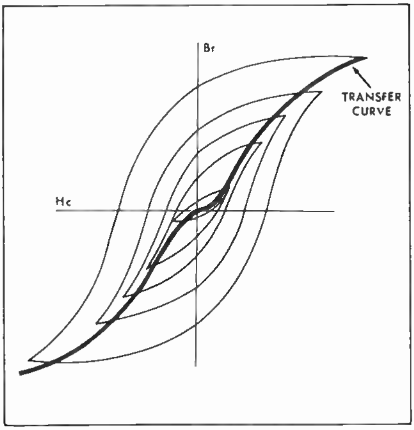

FIGURE 2. TRANSFER CURVE DERIVED FROM A FAMILY OF HYSTERESIS LOOPS

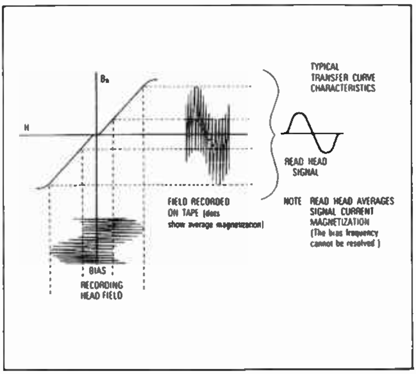

FIGURE 3. RECORDING TAPE TRANSFER CURVE

The transfer curve shown in Figure 3 illustrates the resulting tape magnetization from a magnetic signal generated by the record head. The curve is typical of those for recording tape and no attempt is made to show non-linearities and signal losses created by either the record head or recorder electronic systems.

As the magnetizing force increases (greater record head output in terms of magnetic flux field intensity) the resulting tape magnetization also starts to increase. Notice that the vertical segments of the transfer curve are relatively straight. It is within these straight segments of the curve that undistorted recording takes place. The straight segments indicate that a linear and proportional relationship exists between a given input and the resulting output. This relationship may change for different types of tape because of differences in the magnetic properties exhibited by various oxide coatings.

The straight portions of the curve continue until either saturation in the positive or negative directions occurs. At the saturation points, no effective additional tape magnetization will ocur even if the magnetizing force continues to increase. Recording into the saturation levels may produce distortion, tape noise, and reduce frequency response.

To visualize the recording process, the transfer curve illustrates the resultant signal waveform (sum of bias and input signals), and its transfer across the curve to form the recorded signal waveform (Figure 3). Observe that the signal with bias essentially bridges the "zero-point" and the low signal response portion. The bias position across the curve allows the signal changing portions of the input waveform to fall onto the linear segments of the curve.

The shift of the input waveform across the transfer curve to form the recorded signal waveform shows that the non-linear segment is essentially removed by the bias signal, and the recorded signal is relatively distortion free. Also, it can be visualized that either a low or high bias condition will drive the signal onto the non-linear segments of the curve and will cause distortion.

With a low bias condition, (Figure 4) the low level input signals may fall into the "zero-point" region and either may be severely distorted or not be reproduced. In a high bias condition, (Figure 5) the high frequency response will decrease. The high frequencies will distort sooner or go into saturation because of a phenomenon called "self-erasure" which will be discussed in a future *SOUND TALK*. Also, the signal-to-noise ratio may be reduced causing undesirable tape noise.

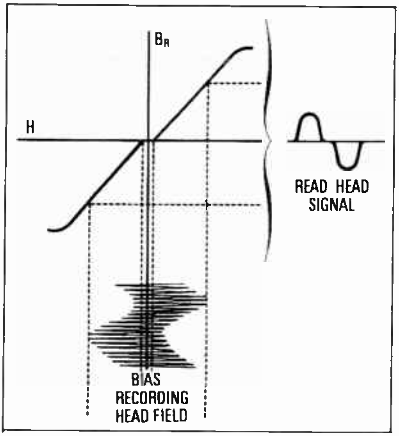

FIGURE 4. UNDER BIAS CONDITION

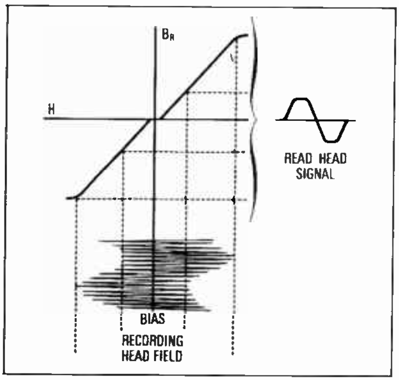

FIGURE 5. OVER BIAS CONDITION

The transfer curve is typical of most ~~magnetic~~ recording tapes but each particular type of tape will exhibit different slope, "zero-point" region, as well as different saturation peaks. The differences of the curve shapes are created by the individual magnetic properties exhibited by each tape type. As the shape of the curve changes so do the bias requirements.

A low coercivity tape has very steep linear segments and will require less bias current. On the other hand, a high coercivity tape has relatively shallow linear segments which require a greater bias current input. Because of the differences in tape magnetic properties the slope of the curve changes and the proper bias level required to eliminate distortion will change accordingly.

To evaluate the changes of bias requirements involved with different types of tape, the tape's wavelength response must be considered. Bias current is required to eliminate distortion but is also directly involved with frequency response and output. In terms of response and output, the bias requirement is related to tape construction such as; type and thickness of coating, quality of oxide dispersion forming the coating, and smoothness of the coating surface.

As a general rule, high frequency response can be improved by using a tape with a high coercivity oxide, relatively thin coating depth and a smooth (specially prepared) coating surface. These improvements of high frequencies may have an opposite effect for the low frequencies to the extent that they may not be reproduced with the same efficiency. This situation becomes apparent in the bias curves (Figure 6).

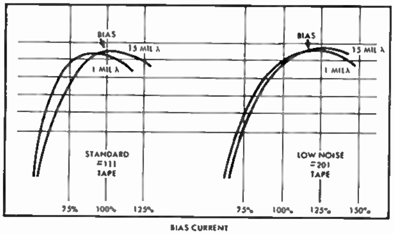

FIGURE 6. BIAS CURVES

Because of the slight differences that may occur in the reproduction of the different frequencies, the actual bias setting is of a selective nature. The ideal situation would be one where the bias setting is at the point of peak output for all frequencies. Note that a bias setting is easily accomplished for the low noise tape as shown in the bias curves (Figure 6). For this particular tape both the high frequency (1 mil wavelength) and the low frequency (15 mil wavelength) output peaks coincide with each other allowing the bias setting to be at the overall output peak.

In the case of the standard tape, the high and low frequency (short and long wavelength) output peaks do not coincide at maximum output. The bias setting could be at either output peak or at the mid point. In normal recorder adjustment however, the bias setting most often used is at the output peak of the longer wavelengths. This setting is justified because the greater percentage of recorded information is in the low or mid-range portion of the frequency spectrum. To compensate for any unbalance in response output, the equalization settings of the recorder are adjusted until the overall output frequency response is flat.

TYPICAL BIAS ADJUSTMENT PROCEDURE

The bias settings shown in the illustration indicate only a relative bias level comparison between two different types of tape. The percentage value relationship will generally hold true for most recorders. Specific information on bias adjustment or settings is impossible to enter into here because of the large variety of recorders in use. Most recorders have their own individual requirements and specifications for bias current (or voltage) adjustments. If a bias level adjustment is attempted – care should be taken to assure correct settings and the recommendations of the recorder manufacturer must be precisely followed.

As mentioned in the preceding paragraphs, the low-midrange frequency (longer wavelengths) output peak is generally used to obtain the most desirable bias setting. The normal adjustment frequency (for 7½ inches/sec. tape speed) is 500 to 1000 Hz. This audio signal is available from an audio, function, or signal generator which most electronic repair facilities have available.

The following adjustment of recorder bias is typical of many machines now in use. For stereo machines, the adjustment procedure must be repeated for both channels. Before attempting any adjustment, be sure that the machine is operating properly, the record and playback heads are clean and in good condition, and thoroughly review the manufacturer's service manual. The bias adjustment range, location, and function of controls, and the operation and scale of the output meters (VU meters) must be understood. Since the bias setting is determined by the type of recording tape, establish the basic type of tape most often used in your particular recorder. Prepare the machine for normal recording at 7½ ips with a low signal level input (approximately 20 db below tape saturation).

Set "Gain," "Record Volume," or "Level" adjustments low to avoid the possibility of recording in the saturation levels. Adjust the signal generator (1000 Hz signal source) for a low voltage output and connect to the recorder input terminals. If the recorder is a 3-head type, while recording the 1000 Hz signal, listen to the recorded signal. Slowly increase the bias current (or voltage) and observe any increase of output as indicated on the VU meters. An increase of intensity of the playback signal should also be heard. Continue to adjust the bias, starting at low output, until the maximum output signal is observed. Continue to increase bias until the maximum output signal is observed.Continue to increase bias until the output begins to drop, indicating an over-bias condition (Figure 6), and return the bias setting to the point of maximum output.

If the recorder is a 2-head type, the set up procedure is similar except that a series of short recordings, each with a change in bias current (or voltage), is made and played back. A simple method is to voice identify the recording segment and bias setting and record the 1000 Hz signal for 10 seconds, readjust the bias current and record and identify another segment. Repeat this procedure for low to high bias current (or voltage). Then play back all the recorded segments noting which one has the greatest fidelity and intensity, and set the bias accordingly.

The recommended bias setting for most recorders is where maximum output is indicated for the 1000 Hz signal. This setting coincides with the low frequency (long wavelength) output peak as shown in the response curve illustrations.

After the correct bias adjustment is obtained, a corresponding equalization control adjustment may, in some cases, be required to compensate for differences in overall response. Usually a simple listening test of recorded material will determine if the overall response is correct.

SUMMARY

High frequency bias current to the recording head is required because of the non-linear characteristic exhibited by most magnetic media. Its major purpose is to compensate for these non-linearities and allow distortion-free recording. Correct bias setting allows undistorted recordings on magnetic tape to the limits established by the record head or the recorder electronics. Proper bias adjustment also assures a better signal-to-noise ratio and optimum frequency response. Maximum performance and fidelity are direct results of high frequency bias.

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LITHO IN U.S.A. WITH 3M OFFSET PLATES

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No. 1
1969

POLYESTER AND ACETATE FOR MAGNETIC RECORDING TAPE BACKINGS

Within the magnetic recording industry, only audible range recording tape is available with both polyester and acetate as a backing material. It is interesting to note that the recording tapes used for other applications such as video, instrumentation, and computer rely almost entirely on polyester backing materials. For these applications polyester is the preferred substrate because of two basic properties, stability and strength. These properties are also of definite interest to the audible range recording industry, so this issue of SOUND TALK will compare the performance of both polyester and cellulose acetate film in terms of their use as backing materials. Some of the physical parameters which define the basic properties will also be discussed.

During its extensive life, magnetic recording tape may be subjected to a variety of environmental changes. The temperature and humidity, which are constantly changing, will affect the backing material used in all recording tapes. The tape backing, during these changes, will expand or contract. This change in physical dimension affects its wind-stability and ultimately its overall life expectancy. These finite variations created by environmental changes are reflected in the basic property of stability which, as we will describe, is different for the two backing materials. During the use of recording tape, the stresses and strains which the tape receives will vary. The ability to withstand these forces is determined by the physical strength of the backing material. The two materials being examined exhibit subtle differences that we can observe by various testing techniques.

STABILITY

A basic requirement of a backing material is to maintain its dimensional stability when subjected to changes of temperature and humidity. Differences which do occur and can be measured when comparing the two materials are expansion, cupping, and wind stability.

EXPANSION COEFFICIENTS

The thermal and hygroscopic coefficients of expansion of each material define some of the differences between them. The thermal coefficient of polyester is 2×10^{-5} in/in/°F. The coefficient of acetate is 3×10^{-5} in/in/°F. We must conclude that neither material is detrimentally sensitive to temperature changes. Even though polyester expands only two-thirds as much as acetate, the numbers are so small that the result is rather unimportant.

The significant difference in materials, however, lies in their moisture or hygroscopic coefficients. Polyester expands or contracts at the rate of 6×10^{-6} in/in/%R.H. Acetate on the other hand has a coefficient of 50×10^{-6} in/in/%R.H. Although these, too, are rather small numbers, notice that polyester is 8 times *less* sensitive than acetate to changes in relative humidity. As an example, let's use these expansion or contraction rates to determine how the length of two 7200 foot rolls of tape will change when the RH changes 60%. The thickness and width will also change the same percentage, but this change will be negligible when compared to the absolute change in length. One of the rolls will be 1 mil polyester and the other 1 mil acetate.

From the following computation, notice that a 60% change in Relative Humidity creates a change in length of the polyester tape of about 2½ feet.

Length (Poly.)

$= [6\,(10^{-6})$ unit change/%R.H.$] \times 7200$ ft. $\times$ 60% R.H.

$= 6\,(7.2)\,(6)\,(10^{-2})$ ft.

$= 2.59$ ft.

Using the same formula, notice that under the same conditions the acetate sample changed over 21½ feet.

Length (Acet.)

$= [50\ (10^{-6})$ unit change/% R.H.$]$ x 7200 ft. x 60% R.H.

$= 0.5\ (7.2)\ (6)$ ft.

$= 21.60$ ft.

At 15 ips, this would amount to a 2 sec. difference, a negligible change, in the running time of the polyester roll. The acetate, however, changes about 17 seconds.

CUPPING

The moisture sensitivity of acetate also shows up in another manner. A short, single strand of acetate tape, if exposed to a wide range of Relative Humidity, will not remain flat through this range. The tape will have a tendency to curl across its width. Observing a cross-sectional view of the tape end, as in Figure 1, you would note the amount of cupping is measured in degrees of arc (flat tape is taken as zero). Samples tested at 15, 50, and 85% R.H. showed the amount of cupping is related to the relative humidity level. A typical 1½ mil acetate tape measures 4° at 15%, 8° at 50%, and 16° at 85%. Thus the acetate may cup badly at high humidities while remaining relatively flat below 50%. The acetate's change in cupping throughout this 70% R.H. range was 12° while the polyester changes less than 1°. The reason for acetate's change is its absorption and loss of moisture that causes a differential in expansion and contraction when compared to the magnetic layer.

Another type of cupping apparatus measures the height of the arc. This distance can be ascertained by viewing the edge of the sample through a calibrated microscope eyepiece. Regardless of the measurement, cupping is considered detrimental because it disrupts head to tape contact.

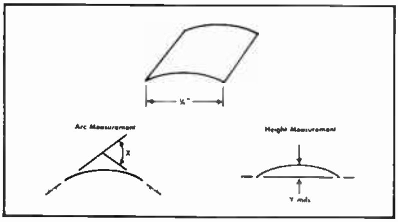

FIGURE 1. CUPPING

WIND STABILITY

Another important humidity effect is apparent in the tightness of a roll of tape. An acetate roll wound at moderate tensions with constant torque at 50% R.H. and normal room temperature will become very loose when the R.H. is raised to 95%. At 5% R.H. and normal room temperature the same roll becomes very tight. For a duplicator with a 7200 foot bulk roll, a loose wind makes the roll quite difficult to handle because the roll may fall apart. A tight wind may cause the roll to become dished, making it difficult to handle on the duplicating slaves. The tight wind could cause stresses within the roll that may result in permanent physical distortion.

A roll of polyester will not exhibit these changes after being submitted to the same range of conditions. Seasonal changes may create either loose and tight winds of acetate bulkpack rolls because of the day to day changes in relative humidity. Some users may consider environmentally regulated storage areas for acetate tape. The wind stability of polyester, with respect to changes in relative humidity, is much more predictable than acetate.

STRENGTH

Another important property of magnetic recording tape is its ability to withstand stresses and strains which occur during use. The simplest method of a comparison between the two backing materials is to subject them both to a series of tests which determine tensile strength during tension and their shock tensile strength to withstand a sudden application of stress. Still another strength parameter is the ability of the backing to withstand minor damage and aging and still remain in usable condition.

TENSILE STRENGTH

The tensile strength test is accomplished by attaching a tape strand to two fixtures or jaws, one is stationary and one is capable of being moved at a constant speed (figure 2). The tension imposed on the stationary jaw is measured by a force transducer as the movable jaw pulls on the sample. The output of the transducer provides an electrical signal which deflects the pen of a chart recorder in proportion to the force. The chart paper is moved in proportion to the distance the movable jaw travels, thus generating a force versus elongation curve. The curve shows the backing's strength when subjected to a constant rate of elongation.

Figure 3 is a graph of typical samples of 1½ mil polyester and acetate tested on the apparatus at 50% R.H. Even though this curve represents 50% R.H. the polyester would be essentially the same at any humidity because polyester's strength is virtually unaffected by the presence or absence of moisture. During this test a typical polyester sample elongates 100%. This is a deceiving figure, however, because magnetic recording tape is useless once it has stretched beyond 5%. The knee of the curve appears at approximately this 5% point. Below 5%, the backing's elasticity allows it to return to essentially the same length and shape as when it was under no tension. Beyond this knee at 5%, the film is permanently distorted. After the 5% point, the tape continues to lengthen without additional force for a short time, then more force is required to reach the breaking point. Finally, breakage occurs at approximately 100%. This percentage can vary from about 90 to 150%, but 100% is typical.

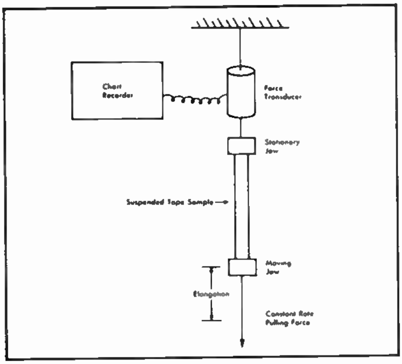

FIGURE 2. TENSILE STRENGTH TESTING APPARATUS DIAGRAM

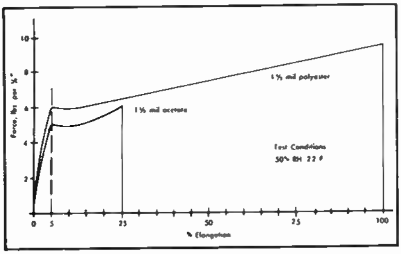

FIGURE 3. TENSILE PROPERTIES OF 1½ MIL BACKING

During the comparison test of a 1½ mil acetate tape sample, notice that the acetate also stretches, but not nearly as much as polyester. Breakage occurs after 25% elongation. Also, this breakage figure varies considerably depending on both the edge quality of the tape and the relative humidity. The permanent deformation point (yield point), is again approximately 5%; the same as polyester. But about 15 to 20% less force is required to permanently distort acetate. For 1½ mil polyester bases, the 5% point is reached at about 6 lbs. per ¼ inch. For 1½ mil acetate, it is only about 5 lbs. per ¼ inch. Notice that under the conditions of test (normal room temperature and 50% R.H.), acetate and polyester both stretch, but the acetate sample is permanently deformed at 20% lower force.

Figure 4 shows how 1 mil polyester and acetate tapes compare. The same general relationship exists between the two different 1 mil bases as with the 1½ mil bases as previously illustrated. The major difference lies in the forces required to stretch and break the tapes. Since the base materials will withstand an established constant force per cross-sectional area, the thinner caliper tapes, having reduced cross-sectional areas, will stretch and break at lower forces.

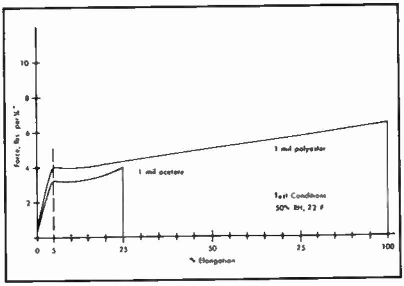

FIGURE 4. TENSILE PROPERTIES 1 MIL BACKING

The effect of temperature on the 5% yield point of 1½ mil acetate and polyester is shown in Figure 5. At low temperatures, both tapes require more force to reach permanent deformation than they do at room temperature. The higher the temperature, the lower the required force. Notice the two lines are parallel, indicating that both materials are affected to about the same degree.

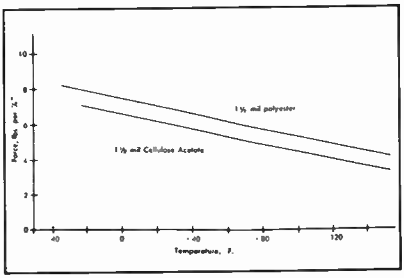

FIGURE 5. TEMPERATURE EFFECTS ON 5% ELONGATION POINT

Since acetate is hygroscopic (absorbs moisture), it has different tensile properties at different humidities. As an example, Figure 6 illustrates the tensile properties of 1½ mil acetate at three different humidities. The tests were made at 15%, 50%, and 80% R.H. with a constant 72°F. temperature. At 85% R.H., acetate stretches much easier and elongates much further than it does at 50% R.H. At 15% R.H., acetate becomes more brittle and will break sooner although it requires a slightly higher force to reach permanent deformation. At high humidities the acetate absorbs moisture which "plasticizes" the backing, allowing it to become more flexible; and, therefore, it is more subject to stretching.

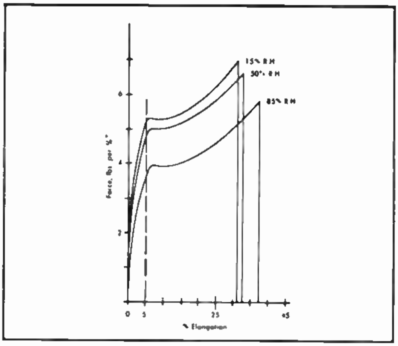

FIGURE 6. ACETATE TENSILE PROPERTIES AT DIFFERENT RELATIVE HUMIDITIES

SHOCK TENSILE STRENGTH

Shock tensile is a test that evaluates how tapes will react to sudden stresses which often can be the cause of breakage. A special instrument is used to apply the forces required during the test. Figure 7 is a simplified drawing of the stress application which corresponds to the Military Specification W-T-0070 testing methods. A weight (or pendulum), attached to a radial arm, is raised a number of degrees and allowed to fall and strike a tape sample. The weight, angle, and radius are determined so that the tape sample at the bottom of the arc is struck with 0.59 ft.-lbs. of energy.

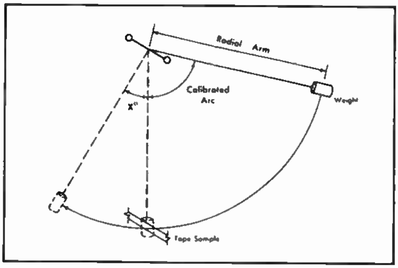

FIGURE 7. SHOCK TENSILE TEST APPARATUS DIAGRAM

The comparison of tapes tested on this apparatus is done by measuring the distance the weight travels after breaking the tape sample (angle X°). If the sample breaks without absorbing any energy, neglecting the frictional losses in the apparatus, the weight will swing to the same height (or angle) as that of its initial position. The difference in the angles, before and after striking the tape, allowing for frictional losses, yields a calculable amount of energy that is absorbed by the tape before it breaks.

The average energy absorption figures for acetate are 0.43 ft.-lbs. for 1½ mil thickness and 0.30 ft.-lbs. for 1 mil. As previously mentioned, because of plasticizing, these figures are extremely dependent upon the Relative Humidity. At lower humidities acetate will absorb less energy and at higher humidities it absorbs more. Depending on ambient conditions, the tape will tend to either plasticize or embrittle. The more moisture, the more plasticized it becomes and the more it stretches before breaking. The stretching action of the tape absorbs the energy of the falling pendulum. At 95% R.H., both 1 mil and 1½ mil acetate material will stretch enough to absorb 100% of the pendulum force and will not break.

During this test comparison, the polyester material of both 1½ mil and 1 mil thickness absorbed the entire pendulum force without breaking, regardless of the relative humidity.

"WEAK" EDGES

A "weak edge" can occur if a tape has been poorly slit during manufacture or damaged in handling. The edge is apparently minutely broken or nicked, and, therefore, has a weak point. To evaluate the effect of the damage to the strength of tape, we took a known good sample roll of 1½ mil acetate. Preliminary tests were performed to assure that it would absorb a minimum of 0.43 ft.-lbs. and stretch normally before breaking. The tapes' edge was then deliberately damaged with a 5 mil nick, using a sharp razor blade and a microscope with a calibrated eyepiece. The edge-nicked samples were then tested. The damaged tape absorbed only 0.07 ft.-lbs. of energy before breaking. No stretching was observed, and the break occurred where the edge nick had been placed. With 5 mils of edge damage, the tape absorbed less than 20% of the energy that it would have if it had good edges. Repeating the same experiment with 1½ mil polyester samples, it was found that the damaged samples did *not* break.

To additionally verify the effect of minor edge damage, static tensile tests were made on samples of the same acetate and polyester rolls. The polyester samples exceeded the 5% permanent deformation point. The acetate samples broke after they elongated about 3%.

Edge damage may be the result of substandard slitting, as occasionally is found in poor quality recording tape. Weak edges also can be the result of improper transport guiding, causing the tape to scrape a reel flange. Damage can also be caused by the bending over of a slightly exposed tape edge in a scattered wind, or by rough handling during thread-up and editing. In all probability, the main cause of acetate breakage is attributable to damaged edges — the direct result of improper handling.

AGING

Cellulose acetate film, as used in magnetic recording tape, contains a plasticizer which is necessary to provide the required flexibility at all relative humidities. It was discovered that old acetate tapes become brittle because of the gradual loss of the moisture and plasticizer over many years time. In an attempt to verify this effect, shock tensile tests were made on artificially aged rolls of tape. The rolls were aged by placing them in a 150°F. oven for 1,000 hours, which, in the chemical industry, is known to be the equivalent of about 2 to 3 years of normal aging. In a comparison between these "aged" rolls and "un-aged" control rolls from the same lot, it was discovered that the strength of the aged rolls was almost ⅕ *less* than that of the un-aged rolls. This verifies previous theories about aging and the loss of plasticizer, causing the acetate to become more brittle. It was found that moisture can be returned to the tape but that the acetate film would never be as flexible or as strong as it was originally. Since polyester does not contain a plasticizer, it does not exhibit this "aging" phenomenon.

CONCLUSION

Because of the greater stability and strength, polyester film is the preferred type of backing for many applications. Although acetate type backings are considered by some as preferable for editing purposes, the polyester must be considered better for operational reasons. For original mastering and duplicating masters, it is important that a recorded tape remain useable in spite of relatively rough handling. Physical distortion caused by changes in humidity that could be encountered during storage cannot be tolerated in conditions which require precise playback. In duplication, it is important that breakage does not occur either during start-up or during operation. Generally speaking, polyester backing materials offer greater reliability and a larger safety factor against problems that can be both time consuming and costly.

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LITHO IN U.S.A. WITH 3M OFFSET PLATES

Volume II
No. 2
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SPLICING TAPES AND THEIR PROPER APPLICATION

An ideal splice is one that, when properly made, will remain intact for an indefinite period of time. Its mechanical strength is the first consideration, but there are other areas that may be counted just as important. There must be an absolute minimum of "adhesive escape" around the edge of the pressure sensitive tape used to make the splice, and the splice itself must not cause an audible disturbance on playback. With these three basic considerations in mind, let's investigate the factors and precautions that become part of the design of a splicing tape by the manufacturer and the fundamental rules and possible pitfalls with which the operator must be concerned.

DESIGN REQUIREMENTS

When designing *any* pressure sensitive tape, the two obvious components are the backing and the adhesive coating. In the development of a tape suitable for splicing magnetic recordings, both of these components were chosen with great care.

The backing had to be tough and durable while being as thin as possible. For this reason, paper was not suitable; and plastic was chosen. Both acetate and polyester are currently being used.

Developing an adhesive coating suitable for splicing tape was even more involved. Here, three basic qualities must be carefully evaluated. These are known as (1) shear adhesion, (2) peel back or ASTM adhesion, and (3) thumb appeal.

Shear adhesion can be defined as the adhesive's resistance to being parted from the surface to which it is adhered when pulled in what is commonly called the shear direction. Figure 1 demonstrates this by showing a piece of splicing tape being tested for its shear strength.

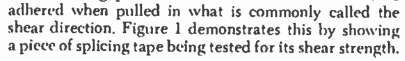

FIGURE 1. SHEAR ADHESION (CREEP)

Peel back or ASTM adhesion is, as its name implies, a measure of the adhesive's resistance to being peeled away from the surface to which it is adhered. Figure 2 graphically demonstrates how this test is performed.

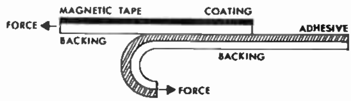

FIGURE 2. ASTM ADHESION (PEEL BACK)

The next property is "thumb appeal" or "quick stick." It is the quality of the adhesive to actually feel sticky. Oddly enough, it is not a particularly important quality as far as the strength of the bond is concerned, but it is a quality that is readily noticeable to the user. There seems to be an "old wives' tale" that has led some users to believe that "the stickier it feels, the better it will hold." This is not necessarily true when talking about splicing magnetic recording tape.

If the thumb appeal is high, the peel back adhesion might be improved to some small degree, but this advantage must be paid for in two ways, neither of which can be tolerated. First of all, with a sticky adhesive the probability of it leaking out from around the bond is greatly increased. This "ooze," as it is called, can be disastrous if it is permitted to exist in splicing tape. The adhesive oozing from under the splicing tape will tend to bond one layer of recording tape to the next layer in the roll. The result, when attempting to re-use the recording tape, would be possible removal of the oxide coating or complete blocking at that point in the reel. Secondly, with an increased "thumb appeal," the shear strength of the splice is reduced. This is evidenced by a degree of parting of the once tightly butted ends of the recording tape and referred to as "creep." Not only will creep manifest itself as an absence of program material or a dropout; but now with the parted joint in the recording tape, the exposed portinn nf adhesive causes the additional problems that we cited above when we discussed ooze. This, then, is why a properly designed splicing tape does not feel very sticky.

SIZE CONSIDERATIONS

Having defined some of the terms, we are now ready tn examine the splice itself. There are several variations in splice geometry from which one can select the combination best suited to the cnnditions of use. These include the size of the spliced area and the angle at which the tape ends meet each other.

Initially, it would be well to discuss the length of a splice and the effect it will have on strength. The length of a splice is dictated, basically, by the amount of curvature it will have to sustain in its path from reel to reel. (Figure 3A).

When the recording tape passes around the sharply curved surface of a guide as pictured in Figure 3B, there is a tendency for the leading edge of the splicing tape to continue in its original direction. It is, in effect, attempting to peel itself away from the recording tape. We are now back to one of our three previously discussed adhesive parameters, that of peel back or ASTM adhesion. With a given splicing tape, the amount of peel back is decided in manufacture and, of course, is constant. The length of the splice has no effect on the tendency to peel but is important for another reason.

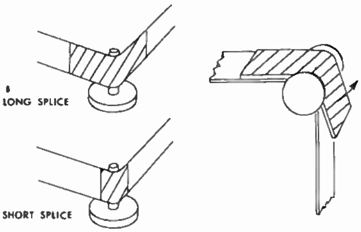

FIGURE 3A & 3B. SPLICE LENGTH AND BEND RADIUS

As shown in Figure 4, a short splice may tend to loosen when subjected to a tight bend because the area of peel may extend far enough into the tape's bond to completely free one end of the recording tape.

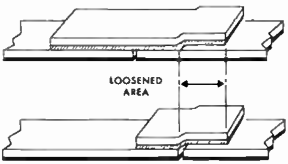

FIGURE 4. ENLARGED VIEW OF LOOSENED SPLICE

A longer splice will exhibit the same amount of peel but the area of peel in this case does not extend all the way to the recording tape junction. The bond at the junction is essentially undisturbed, and the splice passes the guide successfully. Of course, once the spliced area is wound on the take-up reel, the leading edge of the splicing tape that tended to peel is resecured to the recnrding tape by the pressure of the succeeding wraps as they are wound ontn the take-up reel.

While it is impossible to assign a set of definite numerical values, generally speaking, use a long length splice if small radius bends or turns are expected.

As mentioned earlier, the tendency to creep is dependent on the shear strength of the splicing tape adhesive. The force that opposes this shear strength is, of course, the amount of tension the tape encounters on the transport and while wound on the reel during storage. The amount of shear strength is constant for a given splicing tape. If subjected to a constant tension, the important variable affecting creep is then the area of the bond. The larger the bonded area, the better will be the creep resistance.

A splicing tape with poor adhesive shear strength could be used if the area of the splice were greatly increased. Since the width dimension is limited by the recording tape, the area could only be increased by additional length. We could imagine a spliced bond 2 nr 3 feet long, but that, of course, would be almost impossible to execute mechanically. Since the program material may drop in level as much as 4 db in the area of the bond because of the change in flexibility, the shorter the splice, the less disturbance there will be during playback. It is, therefore, important that the splicing tape chosen for use has high adhesive shear strength so the spliced length can be kept short.

SPLICING TAPE WIDTH

Much has been said and written about using splicing tape that is the same width as the recording tape and that which is somewhat narrower. It would be well to examine some of the variables and draw some conclusions.

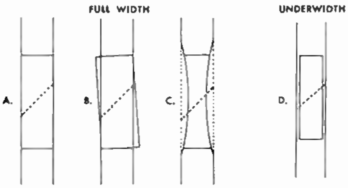

FIGURE 5A-B-C-D. SPLICING WIDTHS

When using the full width splicing method, such as shown in Figure 5A, care must be taken to trim the splicing tape exactly at the edges of the recording tape. If the splicing tape is poorly trimmed (Figure 5B), the overhanging adhesive coated splicing tape is apt to adhere to an adjacent layer on the reel, causing a problem similar to that encountered with ooze. Even though some splicing jigs are designed to cut an arc into each side of the splice, as shown in Figure 5C, to insure against the possibility of overhang, this does not completely eliminate the chances of some adhesive oozing out of the edges.

Figure 5D illustrates a splicing tape somewhat narrower than the tape to be spliced. This technique offers a number of advantages with no apparent disadvantages. Since the splicing tape does not extend to the edges of the recording tape, overlap — as mentioned earlier — is no longer a problem. A simple splicing jig can be used because there is no need to undercut the spliced area in an hour-glass configuration. Notice that the use of a somewhat narrower splicing tape does not appreciably sacrifice the overall bonding area when compared to full width splicing tape that has been undercut.

RECOMMENDED SPLICING METHOD

In conclusion, let's examine the preparation of recording tape prior to the actual application of the splicing tape.

The most desirable method is to cut the recording tape to be spliced at an angle of 45° to 60°, measured with respect to the tape edge. As the angle increases above 60° towards a perpendicular cut, the amount of electrical disturbance is increased because the head sees the discontinuity at the junction as an abrupt change.

The shallower the angle, the less will be the amount of disturbance. But, as the angle is decreased below 45°, the pointed corners of the recording tape become vulnerable to being peeled back or debonded.

Regardless of the type of splice used, the first and possibly the most important consideration is cleanliness. The hands should be free of all dirt, dust, and oils as one fingerprint on the oxide can drop the output several db. Also, contamination of the recording tape backing or the adhesive of the splicing tape will usually reduce the strength of the bond between the two and can result in premature failure. After carefully placing the recording tape in a splicing jig, it should be cut as carefully as possible, using a sharp, demagnetized razor blade. When handling pressure sensitive splicing tape, care should be taken not to handle the adhesive more than is necessary. After carefully laying the splicing tape down so as not to disturb the alignment of the splice, the finger should be rubbed over the tape to promote intimate contact between the two pieces. Then to remove the air pockets, using the flat of the fingernail is recommended. The selection of the proper splicing tape and the use of correct splicing techniques will assure you of a clean, long lasting splice with no audible discontinuities.

If at any time additional information on this topic is desired, it is available by simply writing to:

Product Communications
Magnetic Products Division
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St. Paul, Minnesota 55101

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Magnetic Products Division 3M COMPANY

LITHO IN U.S.A. WITH 3M OFFSET PLATES

VOL. II
No. 3
1969

ALIGNMENT

Since the design of the first tape recorder and the use of magnetic recording tape, the equipment and tape have continuously undergone improvements. Some of the improvements have allowed greater fidelity, better signal-to-noise ratio, excellent reliability and service life. These improvements have also reduced distortion, crosstalk between channels and high frequency (short wavelength) losses.

All of the aforementioned benefits are the result of better machine electronics and also better mechanical drive and guiding systems. Ironically, the advanced electronics, which record and play back the program material, are at the mercy of the mechanical drive and guide systems which move the tape across the head to assure proper head-to-tape contact. Therefore, correct guiding, intimate head-to-tape contact and head alignment are prerequisites for maximum recorder performance, especially when using multi-track or professional type wide-width (½" to 2") tape equipment.

This issue of SOUND TALK will discuss the various elements which are involved in guiding a tape across the deck. Because of the variety of recorders available, guiding and alignment will be discussed in terms of basic principles only; and those adjustments which are deemed necessary should be performed by a qualified technician familiar with the individual machine manufacturer's specifications.

Oftentimes problems with recorder operation or performance degradation are blamed on what appears to be faulty heads or poor tape when, in reality, the problem is actually caused by a misaligned head or improper tape guiding. These problems can occur in any machine, regardless of quality or age. The situation of guiding and alignment is so critical that major recording and duplicating studios make it a practice to systematically check their decks for these parameters to assure proper operation.

To assure that the tape moves across the deck in the proper manner and ultimately crosses the head correctly, it is necessary to establish the correct tape path from the supply reel through the guide system to the heads and back to the take-up reel. Once the correct path is established it is usually quite simple to maintain this condition.

TAPE PATH CENTERLINE

The centerline of each component which is in direct contact with the tape should maintain an unvarying reference plane (Figure 1). The edge clearance limits of these guiding components are within a few thousandths of the prescribed maximum tape widths; therefore, any slight variation from the true centerline reference can cause bending of the tape edges. Additional effects could be tape skewing on a tangent from its path, azimuth misalignment and excessive friction. True centerline tracking is particularly important when using wide tape widths (½" to 2") because wider tapes exhibit greater resistance to longitudinal changes or "steering." Wide recording tape exhibits a tendency to curl or roll along its edges when subjected to the steering action of inaccurate centerline tracking.

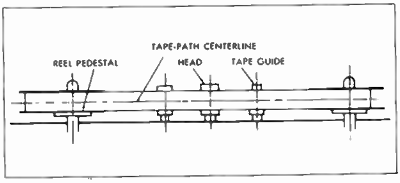


FIGURE 1. SIMPLIFIED TAPE PATH CENTERLINE

PEDESTAL HEIGHT AND ANGLE

Tram error and tape edge damage may occur from improperly positioned tape reel pedestals. A slight variation of the pedestal axis from a true right angle to the tape path creates an exaggerated error at the reel flange circumference (Figure 2).

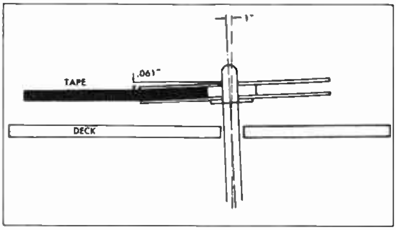


FIGURE 2. IMPROPER PEDESTAL POSITION

A pedestal with an angular error of just one degree will displace the edge of a 7 inch reel flange by 61 mils from the proper horizontal position. If the pedestal height adjustment is incorrect and compounded with an erroneous angle, the total error is cumulative; for example, with an angular displacement of 0.061" and improper height of 0.030" the total error reflected to the tape path is 0.091" – nearly 1/10th of an inch. This type of irregularity will create guiding problems throughout the tape path and can cause the tape to rub along the edge of the reel, creating edge damage (Figure 3).

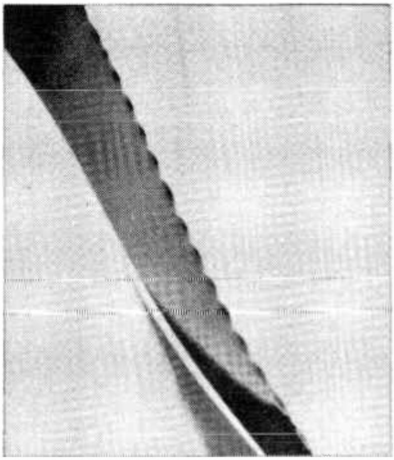


FIGURE 3. SINGLE STRAND OF TAPE SHOWING SEVERE EDGE DAMAGE

The type of edge damage shown in figure 3 may also be caused by a damaged reel. If the reel flanges are bent or warped so that the normal clearance between the tape and the flange is reduced, the tape can scrape against the flanges.

When determining proper pedestal height, differences between the flange thickness of plastic and metal reels must be taken into account. Plastic reels have thicker flanges than metal reels to provide the needed strength. If both metal and plastic reels are used, the centerline reference should be established using a plastic reel. Although this will cause the tape to wind slightly above the center on the metal reel (the thinner flanges will cause it to rest slightly lower on the pedestal), the thinner flanges also provide greater clearance which compensates for the difference in tape wind. When adjusting pedestal height in reference to the tape centerline, it is important to determine the dimensions of the reels normally being used. The Electronic Industries Association (EIA) has suggested basic reel sizes and dimensions in its Standard RS 254A, which specifies a nominal reel width of 0.462" for ¼" reels (nominal tape width plus 0.212"). Other reel widths follow the same standard (½" reel width is 0.500" plus 0.212" = 0.712"). The specified dimensions are standard for the precision reel, which is carefully manufactured to assure concentricity of the hub and flanges, accurate flange run-out and consistent separation distance between flanges. If a precision reel is unavailable or impractical to use (normally available in only 10½" or larger sizes), the dimensions established may be applied to the reel being used to check the transport (see calculations in figure 4).

For ¼-inch Precision Reels:

$$W_F = \frac{W_R}{2} - \left(\frac{W_T}{2} + W_C\right)$$

$$W_F = \frac{.462}{2} - \left(\frac{.246}{2} + .005\right)$$

W_F = .103-inch*

where . . .

W_F = distance between pedestal and nearest flange inner face

W_R = .462-inch average or nominal overall reel width within the lateral mounting area

W_T = average tape width

W_C = desired clearance between each tape edge and adjacent flange

To Then Determine Pedestal Height:

$H_P = X - (W_C + W_F)$

$H_P = X - (.005 + .103)$

$H_P = X - .108$-inch

where . . .

H_P = height of pedestal from reference plane

X = distance from reference plane (deck) to nearest tape edge correctly positioned with respect to guides and heads

FIGURE 4. PEDESTAL HEIGHT AND TAPE PATH DIMENSIONS

*For other (¼, ½, 1, 1¼, 1½, 1¾, and 2") precision reels it can be easily shown that W_F = .102-inch. This is because average tape widths for these reels are .002 rather than .004 inch less than the appropriate multiple of ¼-inch. (e.g., average width for ½-inch tape is .498, for 1-inch tape, .998, etc.).

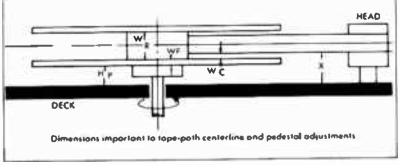


FIGURE 4. PEDESTAL HEIGHT & TAPE PATH DIMENSIONS (CONT.)

Because tape reels are symmetrical, a perfect wind on a correctly adjusted transpnrt would center the tape equally between the flanges. When establishing the centerline, some type of reference must be used. Generally the mounting plate or deck is adequate for measuring the tape position through the entire tape path. By establishing a reference measuring method, such as "X" in figure 4, any deviations in the tape path created by the pedestals, capstan and idlers, or guides can be easily discovered. The intricate calculations shown in figure 4 provide the basic dimensions for establishing pedestal height and tape centerline.

GUIDING

As the tape moves across the deck, its path is determined by a series of guides. The guides may be fixed, roller type, or mounted on tape tensioning devices. Fixed guides, because of the direct mounting, generally will not become misaligned. Fixed guides can create edge damage problems, such as shown in figure 3, if they become worn or damaged.

Movable guides, especially those mounted on tension compensating arms, are vulnerable to misalignment because of bent arms. During tape centerline measurements, be sure to check the perpendicular attitude of the guide with reference to the deck surface throughout its entire operational arc. Precise measurement of the tape path entrance into the guide and the tape exit path will determine proper alignment. Careful visual inspection (with a magnifying glass) of the tape passage through the guide will determine if the tape is being subjected to any excessive edge pressure which may cause curling or bending along the edge of the tape.

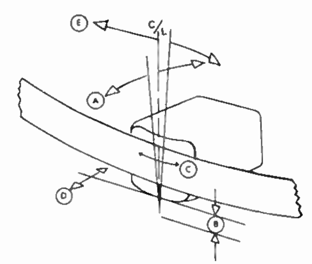
Roller guides, drive capstans and idler wheels can also create guiding problems. Any misalignment or uneven wear on these components may cause the tape to deviate from the ideal centerline. A capstan or idler wheel which is not truly perpendicular to the established centerline or is worn into a tapered shape will cause the tape to travel in an improper path, following the component's angular deviation from perpendicularity.

All of the preceding considerations are intended to assure an even and smooth tape passage throughout the entire path. Proper tape movement across the deck is essential for correct head-to-tape interface. The intimate relationship between the recording tape and the recorder head or heads is the final parameter which must be explored to assure proper operation.

HEAD ALIGNMENT

In magnetic recorders, the high frequency response and inter-channel crosstalk are extremely dependent on head alignment. In most tape transports the heads are adjustable and can be aligned as required to establish the correct head-to-tape interface. The adjustment is very precise and is best accomplished by a factory qualified technician referring to the manufacturer's service manuals. There are five basic adjustments involved in correctly positioning a recorder head, as shown in figure 5. Two of these positioning adjustments (A and B, Figure 5) are concerned with the tape centerline.

HEAD ALIGNMENT — Includes all mechanical adjustments necessary to assure proper coincidence of head gap with tape, or more specifically, a properly recorded tape track.



A) Tilt, in which the face of the head must be simultaneously tangent to the same degree with both edges of the tape and without distortion of either of the latter.

B) Height, in which the gap width dimension is centered on the standard track location.

C) Tangency assures that the tape contacts the portion of the head face containing the head gap.

D) Contact, head position into or away from the tape to assure proper contact pressure between head and tape ("wrap"). Not as critical with machines employing pressure pads at the heads.

E) Azimuth or skew, in which width dimension (corresponds to track width) of gap is exactly 90° with tape edge.

FIGURE 5. HEAD ADJUSTMENT PLANES

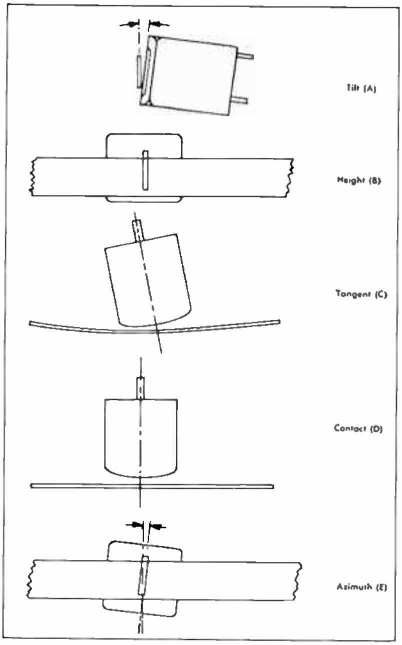


FIGURE 5. HEAD ADJUSTMENT PLANES (CONT.)

TILT

The first basic head adjustment is to establish a true vertical position for the face of the head (Arc A – Fig. 5) with reference to its contact with the tape. The correct attitude is one in which the head neither tilts into nor away from the tape surface. Establishing the correct vertical attitude is important to maintain uniform tension across the entire width of the tape in contact with the head. If the tape is under more tension at one edge than at the other, total intimate contact between the tape and head will be disturbed. The difference in tension can also cause the tape to skew away from the centerline.

HEIGHT

The next basic head adjustment, within the centerline reference, is head height (B – Fig. 5). Improper head height is manifested as mistracking or crosstalk. On multiple track recordings this particular adjustment is very critical in that loss of output, noise and interchannel crosstalk can result if the playback head gap is not perfectly tracking the recorded path on the tape. If recording with a head maladjusted in height, it may be virtually impossible to play the tape back on another machine.

While checking head height, inspect the face of the head for wear. As the head wears, an indentation is formed along the tape path which actually becomes a tape guide (Figure 6). If the head position or tape path is changed, the worn area will no longer coincide with the tape edge. This will cause tape damage. If a severely worn head is discovered, replacement is recommended.

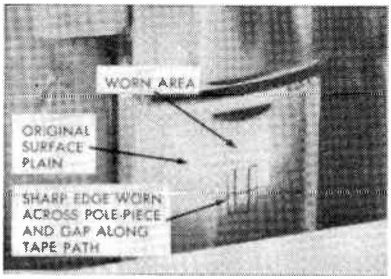


FIGURE 6. EXAMPLE OF IMPROPERLY POSITIONED HEAD EXHIBITING WEAR PATTERN

TANGENCY

Once the tape centerline path across the head is established, the head-to-tape interface must be checked. Tangency (Arc C – Fig. 5) is simply the squaring of the record and playback gaps to the tape's surface. Correct tangency is important to assure proper head-to-tape contact at the head gaps. If the tape contact is not correct, high frequency response will suffer and, more important, the system may become oversensitive to dropouts. Dropouts are usually caused by debris or contamination which separate the tape's oxide surface from the head gaps. Needless to say, any interruption of head-to-tape contact will result in a degraded signal output; and if the separation is severe, a complete signal loss may result.

CONTACT

Contact (D – Fig. 5) is the head position in respect to the tape wrap. Correct head-to-tape contact is assured by the slight bending or "wrap" in the tape path as it passes over the head. Insufficient contact can result in poor high frequency response and oversensitivity to dropouts, as previously mentioned.

Many recorders are equipped with pressure pads which force the tape against the head by applying pressure to the tape's backing adjacent to the head. When inspecting the head position, the pressure pads must be checked for signs of wear or damage. The pad can become worn, developing a channel which corresponds to the tape path. If the pad is deeply worn, head-to-tape contact can be reduced, which will affect high frequency response. Because of the intimate contact between the pad and the tape's backing, surface contamination will tend to stick to the pad. Contamination deposits and build-up on the pressure pad may create hard spots and form an uneven contact surface which can produce "squeal," loss of head-to-tape contact and cause excessive tape wear. If the pressure pad is worn or contaminated, it should be replaced. During pad replacement, care must be taken to assure proper pad size, installation and correct positioning.

A most important head adjustment is that of azimuth (Arc E – Fig. 5). If the reproducing gap (playback head) is not parallel to the recorded poles on the tape, serious loss of high frequency (short wavelength) response will result, as shown in figure 7.

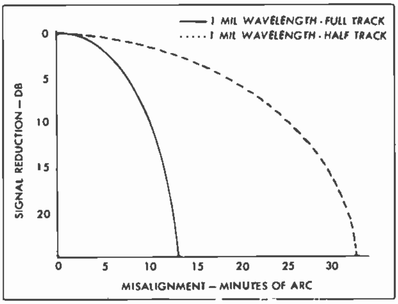


FIGURE 7. HIGH FREQUENCY LOSSES DUE TO HEAD MISALIGNMENT

AZIMUTH

To assure compatibility and interchangeability, it is quite important that record and playback heads are adjusted so the gaps are exactly perpendicular to the tape path centerline. Since it is very difficult to establish true vertical reference with a head because of the extremely small gaps in the pole piece, the azimuth adjustment is best determined by using a special pre-recorded alignment test tape. The alignment test tape has a carefully recorded high frequency signal which, when played back, is used to determine output level. Because of the high frequency dependency on head alignment, any misalignment is readily apparent in the loss of output, as shown in figure 7.

When using an alignment tape to check azimuth, a variety of methods can be employed, the simplest being to deliberately skew the tape across the head while checking output. If the output, as indicated by the signal level meter (or the playback volume), is highest with normal tape alignment across the head, it can be assumed that azimuth is correct. If the output signal level increases while deliberately skewing the tape, it can be assumed that the head azimuth is incorrect and should be re-adjusted. The head should then be realigned to yield maximum or peak output. In the case of separate record-playback heads, the playback head should be peaked per the output signal level determined while using the pre-recorded alignment tape. The record head azimuth should then be peaked while recording on a blank tape and playing back through the correctly positioned playback head. Only a studio prepared pre-recorded tape should be used for an azimuth test.

While checking head azimuth it is also good practice to inspect the pressure pad (if used) for wear. If a pad which has become worn does not properly position itself against the tape, it will have a tendency to skew the tape out of alignment with the head gap, giving the same effect as incorrect head azimuth. If the pressure pad shows signs of a wear-created channel, it should be replaced.

SUMMARY

The improvements in recorder design, electronics and magnetic recording tape have contributed to a media which provides excellent frequency response, low distortion and virtually perfect reproduction of recorded material. The benefits of these improvements are limited to the component condition and the adjustments of each individual machine. By periodically cleaning and inspecting your recorder to verify that the tape path is properly established by the reels and guiding system and that the interface between head and tape is correct, maximum recorder performance can be assured.

If at any time you have specific questions about this topic, simply write to:

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No. 1
1970

THE HANDLING & STORAGE OF MAGNETIC RECORDING TAPE

Much of the world's entertainment and historical events are being preserved on magnetic recording tape. Professional recording studios and tape duplicators, historians and educators, audiophiles and home recordists are all concerned about the permanence and recoverability of the information that is invisibly stored on a thin plastic ribbon.

The preservation of both operating and historical recordings is the primary concern. But, another factor of real importance is the prevention of damage to the recording tape, not just so the information will be safeguarded but so that the maximum use may be obtained from every reel of tape. Both of these factors are economic in nature.

If stored information is unrecoverable because of either lack of safeguards by operating personnel or major catastrophe during storage, the result could be anything from temporary inconvenience to a complete loss of a recording library. If reels of tape are failing before their normal life expectancy, operating expense is increased. Of course, this, too, is undesirable.

This issue of SOUND TALK will discuss in depth the considerations and practices that 3M Company considers of greatest importance to the user of magnetic recording tape. If every one of the many suggestions were followed completely, an ideal situation would exist. Since many recording facilities will function adequately with less than the ideal, you may wish to adopt only a portion of the recommendations. Some of the precautions may be considered too time-consuming or too costly for a given application. In short, it can be said that the overall performance of magnetic recording is directly proportional to the care that is exercised in the two important topic areas: HANDLING & STORAGE.

THE BASIC FACTS

Modern magnetic tape coatings have the ability to retain the intelligence placed on them during the recording process for an infinite amount of time. The recorded information does not tend to fade or weaken with age. It is essentially permanent and will remain unchanged until actually altered by an external Magnetic Field. This erasing of the tape may be done *intentionally*, so that the tape can be used for another recording, or *accidentally*, by operator error or poor storage procedures. Later in this paper the matter of accidental erasure will be more fully discussed.

Even though the magnetic signal will not deteriorate, the physical properties of the recording medium are susceptible to damage. As a general rule, the problems encountered with recording tape performance are predominantly physical in nature. Therefore, it is important to preserve the tape in a form that will make it physically possible to recover the recorded information when needed. Poor handling habits or faulty procedures can render a tape useless because of physical damage. A great deal can be said about the physical preservation of recording tape; and to make the information more meaningful, each of several topics will be treated separately.

THE RECORDING AREA

Ideally the equipment room of a recording studio or professional recording facility should approach, as closely as possible, a "clean room" environment. By definition, this area is characterized by the absence of normally expected airborne dust and lint. The design of the recording equipment area should be such that reasonable control of temperature and relative humidity can be exercised. Variations of temperature should be held within ±5° F. of a pre-selected value and the relative humidity should be kept constant to within ±10%. In broad terms, this would be a temperature in the 70's and a relative humidity of about 40%.

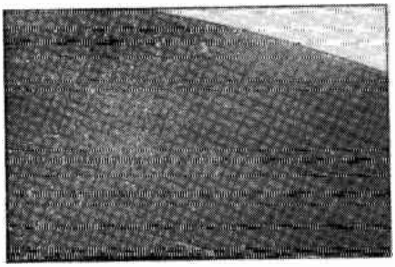

FIGURE 1. ENLARGED VIEW OF DUST CONTAMINATION ON A REEL OF TAPE.

It is doubtful that smoke will contaminate the tape, but ashes can. Therefore, smoking should not be allowed directly over the machines or when handling tape. Food and drink should also be prohibited. Minute food particles can easily be transmitted to the tape and tape decks from the operator's hands. A spilled drink will contaminate not only the tape but also seriously affect a machine's operation.

The integrity of the equipment area should be maintained by periodic cleaning of shelves and floors. When vacuum equipment is used for cleaning, the exhaust from this unit must be located outside the room.

Aside from the direct benefits gained from a well maintained, clean, temperature and humidity controlled environment, the psychological effect upon the employees is of great importance. It is found that operators exercise more care and are more concerned with quality when working in an environment such as just described.

When recording on location or at home, it may be difficult to control the surrounding environmental conditions. Contamination (dust, dirt, debris) can enter the tape transport and cause tape damage. The only positive method of preventing contaminated tape is to eliminate the entry of foreign material into the machine. It is recommended that the recorder (and playback unit) always be covered during storage and as much as possible during operation. Some equipment manufacturers provide, or have available, some type of dust cover which covers the tape drive mechanism and effectively seals out contamination. Many of the protective covers permit the machine to be operated while they are in place and are ideal for use in uncontrolled environment.

TAPE STORAGE

The temperature and humidity of the tape storage area should closely approach that of the work area. The smaller the environmental change experienced by the tape, the better will be its operation and reliability. As a general rule, a temperature between 60° and 80° F. and a relative humidity between 40% and 60% is recommended. If the environmental conditions of the storage area vary widely from the recording area, allow time for the tape to reach temperature and humidity equilibrium before putting it into use.

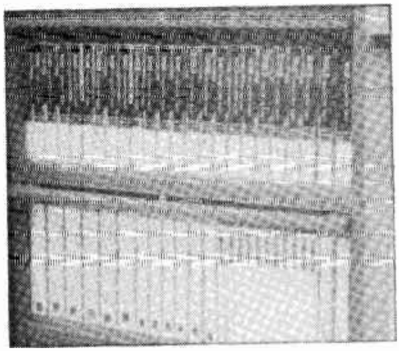

FIGURE 2. EXCELLENT STORAGE METHOD FOR WIDE WIDTH PROFESSIONAL TAPES.

Recording tape, especially cartridges and cassettes, stored or casually laid on the dashboard or in the glove compartment of an automobile can be damaged by the heat generated by strong sunlight. The molded cases used for some cartridges and cassettes can be permanently distorted if subjected to high temperatures. Both cartridges and cassettes use splices within their tape rolls which can be affected by heat. The splices may separate, and the adhesive may soften and "ooze" from the edges of the splice and stick to adjacent tape layers. The exposure of the splice adhesive will also collect any contamination present in the case, causing additional problems.

Protection from accidental erasure while in the storage area is easily accomplished and is, ironically, of little concern. There are two reasons why this is true. First of all, fields strong enough to cause erasure are just not normally found in an "office or home" atmosphere.

Secondly, if the tape is kept as little as 3 inches away from even a strong magnetic source, this spacing should be sufficient to offer adequate protection. During storage, the tape must be enclosed in a container (original box, plastic case, tape canister) for several reasons. One reason is to provide protection from physical damage. Another reason for using a container is obviously protection of the reel from dust.

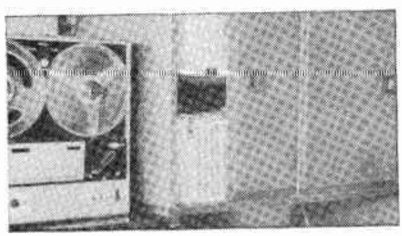

FIGURE 3. TAPE STORAGE AT HOME.

The closed containers should be placed into storage on edge, so that the reel is in an upright position. While they may also be stored individually, lying flat, tape boxes should never be stacked so high that there is a possibility of crushing or distorting the bottom container from the excessive weight of the stack, since this could cause edge damage to the reel of tape in that canister. For long term storage, additional protection from dust and moisture can be gained by sealing the container in a plastic bag. It is generally considered good practice to clean the container before using it so that dust that has accumulated during storage will not contaminate the recorder or tape.

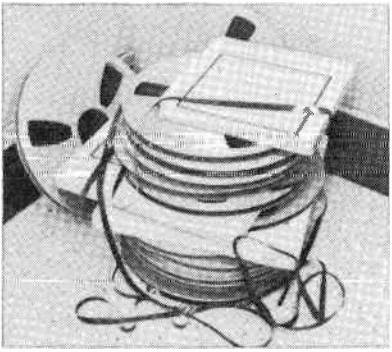

FIGURE 4. OBVIOUSLY THE WRONG WAY TO STORE TAPE.

The care exercised in preparing tapes for storage is every bit as important as the excellence of the storage area. Of primary importance is the way the tape is wound on the reel, since poor winding can result in distortion of the tape's backing.

A wind tension that is relatively low is recommended. Three to four ounces per ¼ inch of tape width is sufficient to render a firm, stable wind on an NAB hub or reel configuration. This tension, while great enough, does not result in high pressures within the roll that could permanently distort the backing. Backing distortion, caused by extreme pressures within the tape pack, may result if a roll of tape wound too tightly is subjected to an increase in temperature while in storage.

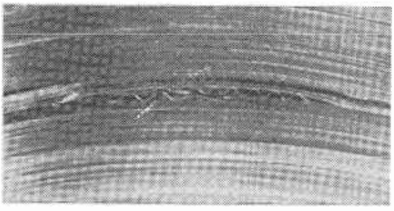

FIGURE 5. CINCHED TAPE. NOTICE DISTORTION OF TAPE LAYERS.

Just as there is the possibility of problems if the tape tension is too great, too low a wind tension can cause difficulty too. If the wind is too loose, slippage can occur between the tape layers on the reel. This "cinching," as it is called, can distort the tape by causing a series of creases or folds in the area that has slipped. When the roll is unwound, the surface will be wrinkled. When an attempt is made to use the tape again, the wrinkles and creases will disrupt the necessary intimate contact between the tape and the head. Because the tape is repeatedly lifted from the head, the result will be a series of signal variations. If the tape is properly rewound immediately after cinching, there is a good possibility that the information may be saved.

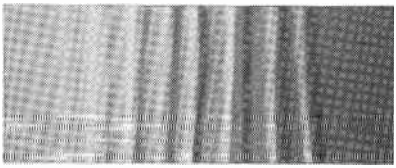

FIGURE 6. TAPE DAMAGE CAUSED BY CINCHING. THIS STRAND OF TAPE CLEARLY SHOWS THE WASHBOARD-LIKE WRINKLES.

Some recorders now in use do not have a method of adjusting wind tension; therefore, care must be taken while operating these machines. Sensible operation of "Fast Forward, Rewind and Start" controls can eliminate the sharp stress loading associated with starting and changing tape directions. Tape distortion and "cinching" can be reduced by allowing a minimum slack loop when threading and starting the machine. It is also good practice to allow the spinning tape reels to completely stop before changing tape direction.

Along with proper tension, another important consideration is wind "quality." The successive layers of tape should be placed on the reel so that they form a smooth wind with no individual tape strands exposed. A smooth wind offers the advantage of built-in edge protection.

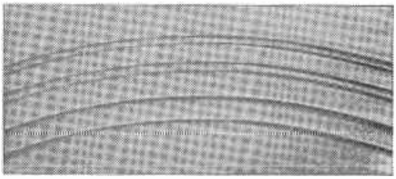

FIGURE 7. SCATTERED WIND. INDIVIDUAL TAPE STRANDS ARE EXPOSED AND VULNERABLE TO DAMAGE.

A scattered wind will allow individual tape edges to protrude above the others. Since there is no support for these exposed strands, they are vulnerable to damage.

It is sometimes suggested that tapes in storage be rewound at specific intervals, such as every 6 to 12 months, to relieve internal pressures. This would be recommended for tapes of marginal quality or for those with other than heavy duty binder systems. For modern day tapes with polyester backings and advanced binders, this periodic rewind might not be necessary.

A good practice, however, is to select a random sample from various areas of the library for visual inspection. The reels chosen can be examined for loose winds and dust accumulations. They should be checked for rippled edges and other signs that indicate the presence of physical distortion. If anything is found that indicates a problem may exist, additional samples should be inspected to ascertain what percentage of the library may be affected.

If the above recommendations concerning the storage environment and the actual preparation for storage are followed, no serious problems should be encountered even in long term storage.

WHEN TAPES ARE SHIPPED

It is sometimes desirable to send recorded tapes from one location to another. There are certain precautions that apply to the shipment of recording tapes that should be followed to insure safety in transit.

Logically, the first consideration would be the physical protection of the tape while being transported. The outer shipping container into which the tapes are placed must afford the necessary strength and rigidity to protect the tape or tapes from damage caused by dropping or crushing. While a container that is 100% water-tight would not be necessary, it must nevertheless provide a reasonable degree of water resistance. It should, for example, be capable of protecting the contents from being damaged if, during shipping, it is left on a loading dock in the rain.

While it is good practice to always secure the free end of a reel of tape, it is particularly important when preparing reels for shipping. A short length of pressure sensitive tape is all that is necessary.

While the purely physical shipping precautions are not unique to magnetic tape but are considered good practice in preparing any item of value for transport, there is another consideration that is of prime importance. Since the tape is a carrier of magnetic information, measures must be taken to protect the reels from accidental erasure.

Laboratory conducted tests have determined what would constitute adequate protection from stray magnetic fields of a magnitude which may possibly be encountered in transit. It was found that field strengths within the tape of 50 oersteds or less caused no discernible erasure.

The average bulk degausser, purposely designed to produce a maximum external field that is used to erase tape while still on the reel, produces a field of 1500 oersteds. Sources of magnetic energy to which tape being shipped might be subjected would be motors, generators, transformers, etc. These devices are designed to contain their magnetic fields to accomplish some type of work. With this in mind, it is safe to assume that field strengths of more than 1500 oersteds would not be encountered in ordinary shipping situations.

Because field intensity decreases rapidly with distance from the source, the 50 oersted point (mentioned earlier as not affecting the tape) is reached at a distance of 2.7 inches from a 1500 oersted source. From this it can be seen that the easiest and least costly method of obtaining erasure protection is by insuring a degree of physical spacing from the magnetic source. It is suggested that tape being prepared for shipment be packed with bulk spacing material such as wood or cardboard between the tape boxes and the outer shipping container.

Based on the information in the paragraphs above, 3 inches of bulk spacing should give adequate protection and virtually eliminate any potential for erasure. This magnetically protective spacing can also be justified because of the excellent protection gained against physical damage to the contents.

Tape in transit may be subjected to temperature extremes. Temperatures as low as −40° F. might be encountered in the cargo hold of high flying aircraft. A temperature of 120° F. could be encountered in a motor vehicle in the summer sun. It must again be emphasized that all incoming tape should be allowed to reach environmental equilibrium before being used.

GOOD OPERATING HABITS

The container in which the tape is stored is probably the cleanest area in the recording studio; and, of course, this is the reason that tapes should remain in the box until actually placed on the tape deck and be returned to the container immediately after use. To maintain the cleanliness of the container, it should be closed when the tape has been removed for use.

The hub is the strongest and most stable part of the reel. *Always handle the reel by the hub and not the flanges.* If this single fact is remembered, you will never be guilty of squeezing the reel flanges together when picking up a roll of tape or when handling it.

It has been said that careless handling and poorly adjusted tape decks are the two predominant reasons why tapes fail prematurely. If strict attention is paid to these two areas, immediate benefits will be noted in increased tape life, and the threat of information loss will be substantially reduced.

When handling tapes, use utmost caution to insure that the tape does not become contaminated by fingerprints. Simply stated, fingerprints are nothing more than deposits of body oils and salts. These oils will not attack the oxide-binder system, but they will form excellent "holding-areas" for dust and lint.

Fingerprints on the backing are just as serious as on the coating because dirt deposits will transfer from the backing of one wrap to the coating of the next wrap on the reel. When a reel that has been contaminated in this manner is put into use, the tape deck itself can be affected and will spread this contamination to other clean reels of tape that are used after the dirty reel.

This is one of the reasons for stressing the importance of visually inspecting the tape deck after each roll of tape is run to determine if cleaning is necessary. If the deck becomes contaminated with dust or wear products from the tape, complete contamination of an entire roll of tape can easily be the result. Contaminants can collect on heads and guides and be dumped along the backing or coating surface of the tape. This contamination will then be wound into the reel under pressure, causing it to adhere firmly to the surface. Each one of these deposits will appear as a dropout or group of dropouts the next time the tape is used.

Tape contamination caused by fingerprints can be reduced by remembering not to touch the tape unnecessarily. Frequent cleaning of the tape deck will reduce the chance of spreading contamination from one reel of tape to others in the library. A cotton swab or lint-free pad moistened with Genesolve-D (an Allied Chemical Trademark) or Freon TF (a DuPont Trademark) or similar cleaner is recommended for cleaning all components along the tape path. If other types of cleaning agents are used, they should be given time to thoroughly dry before loading the tape. This will prevent damage, should the cleaner have any tendency to attack the magnetic tape. Accumulation of tape wear products on the transport can be largely eliminated by using a high reliability tape.

Empty reels should be thoroughly inspected and cleaned before winding tape on them for storage. Reels with hub damage, such as a plastic burr, or with dirty hubs can cause tape distortion exactly as outlined in the preceding paragraphs. Maintaining reel integrity cannot be over emphasized since valuable information can be lost, not because of tape failure but because the tape was distorted by a dirty reel.

One of the most serious and more common forms of tape failure is generally categorized as edge damage. Damaged edges can be caused by the reel, the tape deck or the operator. A broken or badly distorted reel can quickly damage a tape. The effect of a broken or cracked flange is easily noticed since the tape will exhibit a series of nicks or mutilated areas along one edge, and the cause can be easily detected because of the obvious defect in the reel. A bent or distorted reel, however, can also cause damage to one or both edges if the tape is allowed to rub against the flange when being used. A similar type of edge damage will also occur if any of the deck components are misaligned.

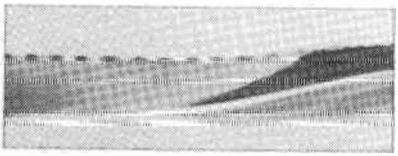

FIGURE 8. SEVERE EDGE DAMAGE.
NOTE THE WAVELIKE WRINKLES ALONG EDGE OF TAPE.

Either of these faults can result in complete failure of a roll of tape. Not only will the edge track be lost, but the debris generated from the edge damage can be redeposited onto the surface of the tape across the entire width. An examination of the edges of a tape that has been damaged in this manner would disclose an accumulation of backing oxide debris.

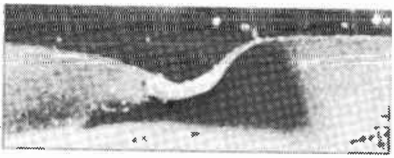

FIGURE 9. MICROSCOPIC VIEW OF A DAMAGED EDGE.
AFFECTED AREA EXTENDS ABOUT 15 MILS INTO TAPE.

While this type of damage is serious, it is sometimes difficult to ascertain its cause or even to notice the effect until the damage is severe. Operators must acquire the habit of physically inspecting the deck in the area of the guides and heads for an excessive build-up of oxide or backing debris. This is generally the first clue that something is wrong. Excessive dropouts on an edge track or loss of high frequencies may also indicate that an alignment or tracking problem exists.

It is also good practice to observe the physical condition of the tape. A sure sign of developing edge damage would be a lip or distortion on the edge being injured. When wound on the reel, the effect of this lip will be cumulative and can stretch the backing. The stretched backing will be rippled and will not conform to the recorder heads the next time the reel is used.

If tape in this condition is properly rewound immediately before being put into storage, it may be possible to salvage the roll. If this is not done, the backing will be permanently stretched and will not recover. This will result in the entire roll having to be discarded.

Operating personnel should use care in handling the reels of tape. It is important that the reel be picked up in a manner that will not cause the flanges to be squeezed together. When the reel is mounted on the recorder, pressure should be applied only to the hub and never to the flange. If the flanges are forced against the tape, this could result in edge damage. This is particularly true if the roll has a scattered wind, since the exposed edges of the misaligned strands can be folded over and creased.

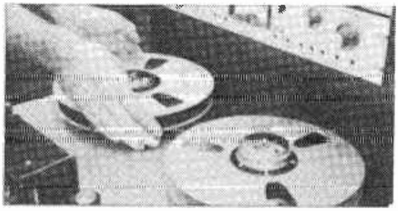

FIGURE 10. PROPER METHOD OF MOUNTING A REEL ON A DECK.
APPLY PRESSURE TO THE HUB AND NOT THE FLANGES.

It is strongly recommended that operators be constantly on the alert for signs of potential trouble. This can best be accomplished by understanding what to look for and by making continuing inspections of both tape and deck a habit.

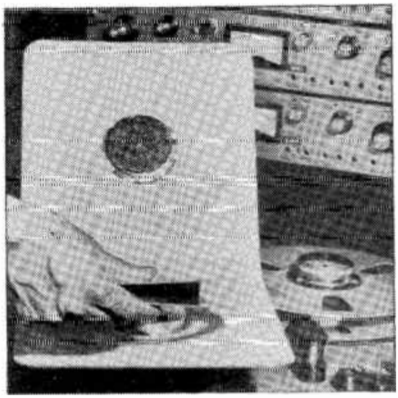

FIGURE 11. IMPROPER HANDLING OF TAPE REEL SQUEEZING THE FLANGES MAY CREATE SERIOUS TAPE DAMAGE.

MAJOR CATASTROPHE

The discussion, to this point, has been devoted to precautions and suggestions involving the day to day routine use of recording tape. Topics have been explored concerning areas in which the tape is used and stored and recommendations for operator education have been made. The final area of concern, while a remote possibility, is nevertheless of utmost importance because it affects not just a single reel of tape or an isolated recorder but the entire recording operation. This section will be devoted to two forms of major catastrophe: *Fire & Nuclear Radiation.*

For a substance to burn, there must be a breakdown of the organic materials contained in it. The organic materials in Magnetic Tape are the plastic backing and the binder. To burn, these must first vaporize – thus increasing their exposure to the oxygen in the atmosphere – and then rapidly oxidize to form light and heat. An ample supply of oxygen is required to sustain burning.

Since Magnetic Tape contains no "built-in" oxidizer, it cannot burn in the absence of air. Simply stated, its behavior can be closely compared to the way in which a tightly wound roll of paper would burn.

While the "self-ignition" temperature of polyester backed tape is in the neighborhood of 1000° F., temperatures below that point can still cause damage. Polyester film will shrink 1½ % at 300° F. and 25% at 325° F. Acetate film, because of its sensitivity to heat, will exhibit greater shrinkage and backing distortion and is more susceptible to heat damage than polyester. If a roll of tape is heated to the approximate temperatures listed below, certain effects would be noted when the roll had cooled.

250° F. – Backing distortion.
320° F. – Softening of both the backing and binder with some "blocking" or adhesion of adjacent layers.
550° F. – Darkening and embrittlement of the backing and binder.
1000° F. – Charring of the backing and binder.

When charring occurs, the tape cannot be unwound from the reel, since it will flake when touched. The temperature limitation of present day tapes is a function of the organic components and not a function of the gamma ferric oxide.

Winding and storing magnetic tape properly will lessen the possibility of damage in the event of fire, since tape is a poor conductor of heat. It is sometimes possible to recover information from a tape receiving slight fire damage by carefully rewinding it at minimum tension. The information it contains should be transferred immediately to another reel of undamaged tape.

We recommend the CO_2- type of fire extinguisher for combating burning magnetic tape. CO_2- is clean and this type of extinguisher contains no chemicals that could harm the tape. If water reaches the tape, it will probably not cause complete failure but there may be some evidence of "cupping" or transverse curvature. The amount of "cupping" will depend on the quality of the wind, backing material and the length of time the roll was exposed. If the wind is loose or uneven, the water can more easily reach the oxide surface and the cupping will be more pronounced. The tape should be removed from the water as soon as possible and certainly within 24 hours.

After removal, the rolls should be allowed to dry on the outside at normal room temperature and then be rewound a minimum of two times. This will aid the internal drying and will also help the rolls to return to equilibrium faster. If moisture is allowed to remain within the roll, severe blocking can be the result.

If a temperature increase is also incurred while the tape is water soaked, steam or at least high humidity will be present. This is more likely to cause damage than water alone. A temperature in excess of 130° F. with a relative humidity above 85% may cause layer to layer adhesion as well as some physical distortion.

Once again, the importance of keeping rolls of tape in their containers must be emphasized. The container, if closed properly, will help keep the water spray of a sprinkler system from reaching the tape.

To prevent fire involving magnetic tape, store tape in a non-combustible area and make sure that no combustible materials are stored in the vicinity. An example of a "non-combustible" area would be a room with metal shelves and sheet metal walls. For maximum fire security, store magnetic tape in a fireproof vault that is capable of maintaining a desirable internal temperature and relative humidity for a reasonable length of time.

As a general statement, it can be said that magnetic tape will be unaffected by Nuclear Radiation until the dosage approaches a level 200,000 times greater than that which would cause death in 50% of the exposed humans. Radiation of this level (100 megarep) would tend to increase the layer-to-layer signal transfer or "print-through" by about 4 db but would not prevent information retrieval.

Nuclear Radiation at the above level will also have some physical effect on the tape coating and backing. The backing will show significant embrittlement, and it is expected that the wear life could be reduced by as much as 60%. It is reasoned that whatever Electro-Magnetic Field might result from a nuclear detonation would not be of sufficient intensity to adversely affect the tape; therefore, the threat of signal erasure is virtually non-existent. The effect of Neutron bombardment would no doubt be limited to activation of the iron-oxide in the coating. This would produce a radioactive isotope that itself might become a source of further radiation, but it is theorized that such activation would not produce a change in the overall magnetic properties of the coating.

Radioactive dust or fallout is not capable of producing the dosage necessary to adversely affect magnetic tape. The recommendations made earlier to protect the tape from normal contamination are applicable, as well.

Recent laboratory tests concerning exposure of recorded tapes to x-ray have determined that the recorded signal is not affected by even severe exposure to this source of radiation. The tests involved a commonly used recording tape with several different frequencies recorded on it. The x-ray machine was operating with 200 MA at 110 KV, and a 6 second exposure time at a 36 inch distance was used. Testing and measuring the signal output before and after exposure indicated no signal loss or degradation.

As can be seen from the above discussion, when speaking of major catastrophe, heat and fire damage are considered much more serious than the effects of radiation.

Under proper storage conditions, magnetic tape has the ability to retain intelligence for an indefinite period of time; of greatest importance is the physical preservation of the medium so that adequate head to tape contact can be maintained when the tape is again put into use.

If at any time you have specific questions about this topic, simply write to:

Product Communications
Magnetic Products Division
3M Company
3M Center
St. Paul, Minnesota 55101

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Recording Institute of America's Course

in

MODERN RECORDING TECHNIQUES

by Robert E. Runstein

Presently available books written about sound and sound studios cover mostly sound reinforcement, film sound, or radio/tv broadcast sound. Pop music recording is mentioned only briefly, if at all. In this book, Robert E. Runstein has fulfilled a need. He has written a book that fills the information gap between recording engineers, record producers, and recording artists.

This book introduces the reader to the equipment, controls, and techniques used in modern studio recording. Both presently employed and aspiring recording engineers will find coverage of all aspects of their job. Producers and recording artists will find that familiarization with sound studio equipment and controls leads to better communication with the recording engineer and, ultimately, to the creation of better records.

ABOUT THE AUTHOR

Robert E. Runstein has been associated with all aspects of recording, working as a performer, sound mixer, electronics technican, A & R man, and record producer. He also served as Chief Engineer and Technical Director of a recording studio. He is an instructor for the Recording Institute of America and has taught several courses in modern recording techniques. Much of the material in this book was originally prepared for these classes. Mr. Runstein is an associate member of the Audio Engineering Society.